ABSTRACT

Title of dissertation: LINK ADAPTATION IN WIRELESS NETWORKS: A CROSS-LAYER APPROACH

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Conventional Link Adaptation Techniques in wireless networks aim to overcome harsh link conditions caused by physical environmental properties, by adaptively regulating modulation, coding and other signal and protocol specific parameters. These techniques are essential for the overall performance of the networks, especially for environments where the ambient noise level is high or the noise level changes rapidly.

Link adaptation techniques answer the questions of *What to change?* and *When to change?* in order to improve the present layer performance. Once these decisions are made, other layers are expected to function perfectly with the new communication channel conditions. In our work, we have shown that this assumption does not always hold; and provide two mechanisms that lessen the negative outcomes caused by these decisions.

Our first solution, MORAL, is a MAC layer link adaptation technique which utilizes the physical transmission information in order to create differentiation between wireless users with different communication capabilities. MORAL passively

collects information from its neighbors and re-aligns the MAC layer parameters according to the observed conditions. MORAL improves the fairness and total throughput of the system through distributing the mutually shared network assets to the wireless users in a fairer manner, according to their capabilities.

Our second solution, Data Rate and Fragmentation Aware Ad-hoc Routing protocol, is a network layer link adaptation technique which utilizes the physical transmission information in order to differentiate the wireless links according to their communication capabilities. The proposed mechanism takes the physical transmission parameters into account during the path creation process and produces energy-efficient network paths.

The research demonstrated in this dissertation contributes to our understanding of link adaptation techniques and broadens the scope of such techniques beyond simple, one-step physical parameter adjustments. We have designed and implemented two cross-layer mechanisms that utilize the physical layer information to better adapt to the varying channel conditions caused by physical link adaptation mechanisms. These mechanisms has shown that even though the *Link Adaptation* concept starts at the physical layer, its effects are by no means restricted to this layer; and the wireless networks can benefit considerably by expanding the scope of this concept throughout the entire network stack.

LINK ADAPTATION IN WIRELESS NETWORKS: A CROSS-LAYER APPROACH

by

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Anneme, babama ve agabeyime

To my mother, father and brother

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Chapter 1

Introduction

IEEE 802.11 [5] standard is the de facto wireless LAN solution widely accepted in the market today. The worldwide adoption rate of Wi-Fi [1] has reached almost to 400 million units margin in 2008 alone (Figure 1.1[2]) with the historical milestone of 1 billion cumulative shipments around midyear. Even more impressive is that, with its extensive popularity in cellular handsets and consumer electronics, the number of Wi-Fi shipments in year 2012 is expected to be well over a billion [3].

Freely available 2.4GHz ISM band is an attractive option for constructing a wireless network; both for high throughput wireless local area networks such as IEEE 802.11 WLANs, as well as low throughput wireless personal area networks such as Bluetooth [10] and low power IEEE 802.15.4 [9] WPANs. As the wireless medium is the common asset shared among all the wireless users regardless of the standards they adhere to, the wireless traffic on the 2.4GHz band has developed into a very heterogeneous system consisting of several different types of packets transmitted with different modulations, lengths, power levels, etc. Coexistence of these different types of wireless standards has been an active research area for the past several years [61, 70, 72, 71, 134, 135, 141].

[1] Wi-Fi Alliance [4] is a non-profit organization offering IEEE 802.11-compliance certification programs. Wi-Fi is commonly used as a synonym for the IEEE 802.11 technology.

[2] Data gathered from In-Stat, Research and Markets, and ABI.

[3] Source: "Wi-Fi IC Market Share Analysis and Forecasts", ABI, 2010.

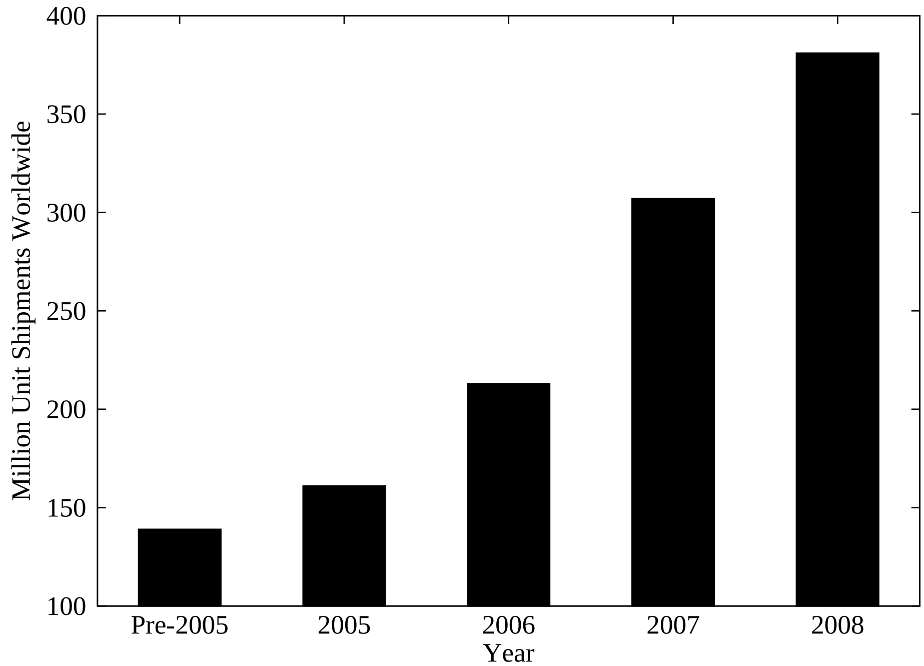


Figure 1.1: Worldwide Wi-Fi Chipset Shipments.

With their broad deployments in widely-ranging settings, high speed wireless standards, such as IEEE 802.11, support interference resistant, low data rate choices in order to provide high throughput levels to their users even in harsh environments. Consequently, these standards offer a plethora of different data rates and create very heterogeneous wireless systems. IEEE 802.11b, for example, can support four different data rates, namely, 1 Mbps, 2 Mbps, 5.5 Mbps and 11 Mbps using different coding techniques. Figure 1.2 shows a small time segment of the wireless network traffic trace recorded during SIGCOMM conference in 2004[4] which illustrates the data rate diversity of the IEEE 802.11 traffic.

The choice of the data rate to use is not specified in the standards and is

[4]Data was gathered from [127], however information extraction and visualization were prepared by the authors.

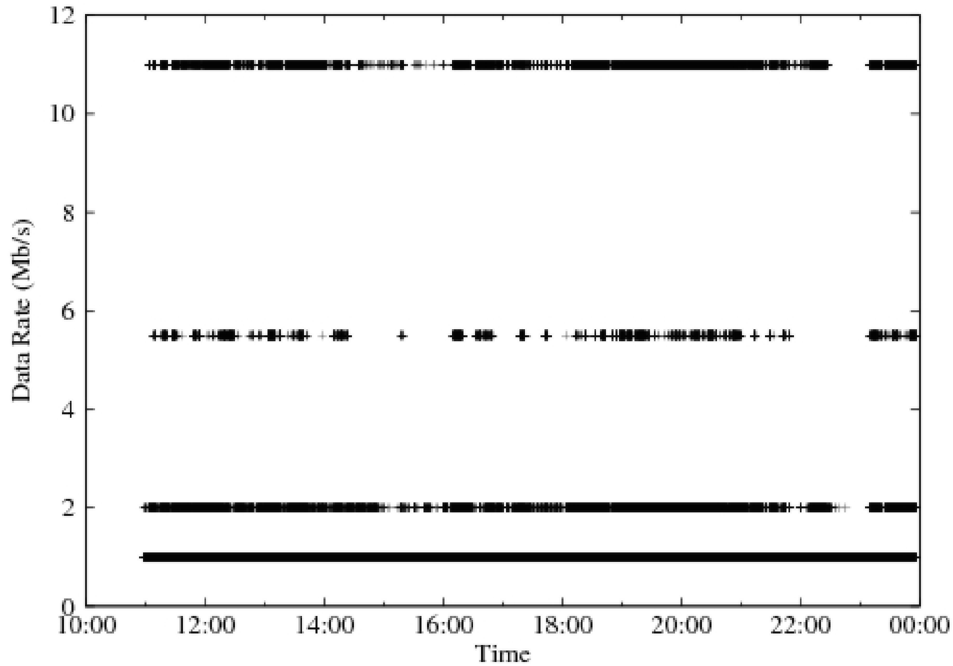


Figure 1.2: Data rate diversity in SIGCOMM 2004 Conference.

left open solely to the decision of the user (i.e. chip vendor). The users are free to choose the data rate as long as they support the basic data rates (1 Mb/s and 2 Mb/s) for specific parts of the communication. The lower data rates are known to be less susceptible to low Signal-to-Noise ratios, as they utilize less complicated modulation techniques with higher amount of redundant data per bit. This freedom let the chip developers design a new set of algorithms for adaptively choosing the best suitable data rates according to the link characteristics, known as *Link Adaptation Techniques* in the literature [123]. In this document, we refer to these types of systems, which are capable of transmitting and receiving packets in different data rates, as *multi-rate networks*.

Conventional networks are designed according to a layered principle, such as

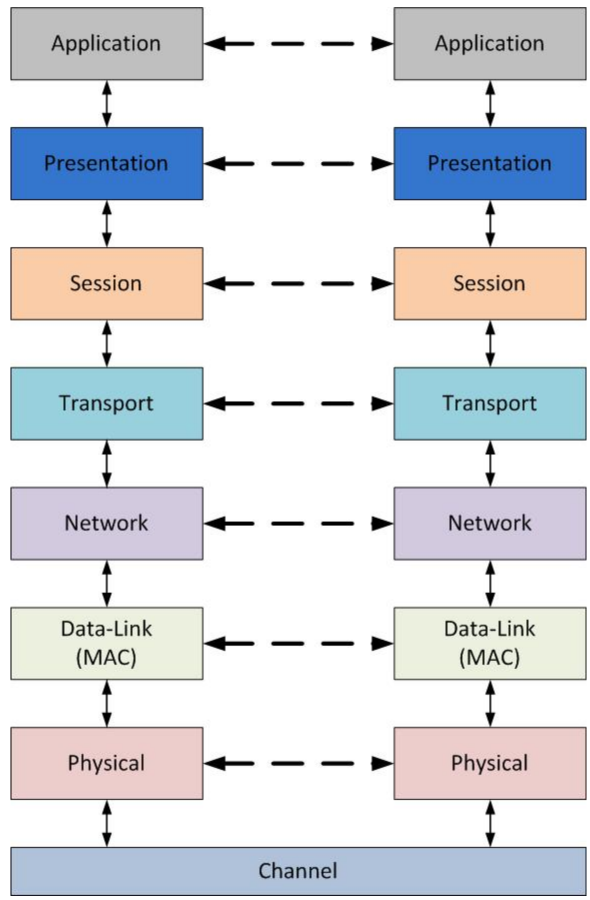


Figure 1.3: Layered OSI Architecture. Dashed lines represent interaction between different communication end points (i.e. devices or implementation instances) and solid lines represent interaction within the same endpoint.

the OSI architecture [156] shown in Figure 1.3. In this approach, the communication is broken into logical entities, called layers, each addressing a specific functionality. Each layer wraps the lower layers isolating them from the higher layers, and each layer interacts with the same corresponding layer at different communication end points (end points might be different devices or different implementation instances on the same device). Layered approach provides modularity, design abstraction and manageability [116], enabling the protocol designers to focus on the layer of interest.

According to the layering principle, the physical layer converts the data bits it received from upper layers into physical representation ready to be transmitted on the channel. The output of the physical layer depends on the data rate and the modulation technique in use which is set by the link adaptation technique in effect, if there is any. The data rate to use for a transmission attempt depends on the channel conditions and it is chosen through estimation of the packet error rates via mechanisms such as counting the number of lost packets [94] and channel probing [5] [69]. In this architecture, the layers above MAC do not utilize any physical layer information and assume that the current data rate in use is optimal for their processes. On the other hand, the interaction between the MAC and physical layers for data rate choices is purely one directional, and can be perceived as the higher layer assigning the modulation technique to use for the next transmission attempt to the lower layer.

Our thesis is that if the information about the choices done by the physical lay-

[5]Channel probing can be done through transmitting short packets just before the actual transmission in order to test the quality of the channel.

ers is conveyed to the higher layers through a cross-layer architecture via stretching the firm layering principle of interaction only between adjacent layers, the performance of the overall system can be improved through proper enhancements.

The use of cross-layer approach in wireless networks comes in different flavors [137]. Some of the works propose joining the functionality of multiple layers together (such as joint PHY-MAC design for better collision resolution [48]), some use extra information from other layers otherwise not transferred to the present layer (such as usage of explicit congestion notification from network layer to transport layer [131]) and others adjust parameters that span across several layers (such as adaptive modulation and ARQ adjustment through delay requirements [98]). Our cross-layer design proposed falls into the second category and does not violate any layer boundaries, merely modifies the encapsulated layers with new algorithms, with the addition of extra information from the physical layer. This dissertation presents two independent new algorithms for MAC and network layers, each using the data rate information obtained from the physical layer. The following sections describe these algorithms and the challenges they attempt to resolve.

1.1 Challenges Related with Multi-Rate Wireless Networks

Regardless of the standards followed or the equipment used, wireless networks pose the following main challenges:

- **Shared Medium:** In wireless networks, the communication is untethered as the medium of data transmission is the radio spectrum (or as commonly termed, *the air*). This feature enables the flexibility of rapid deployment for wireless networks, as there is no need for wiring and a network can be set up in minutes, merely by installing an access point. However, the same feature also limits the performance of individuals as the scarcely available medium has to be shared among several users concurrently.

- **Dynamic Communication Characteristics:** Wired networks enjoy dedicated physical links, shielded strongly from outer disturbances. In contrast, wireless medium is much more dynamic as the radio waves reflect from large objects, diffract around relatively smaller ones and scatter through others. In addition, the received strength of a transmitted signal heavily depends on the distance between the sender and the receiver; and the noise levels in the vicinity of the receiver affect the quality of the transmission. These physical properties complicate the communication, causing effects such as multipath interference, fading and shadowing. Therefore, compared to its wired counterparts, wireless networks are much more dynamic, unpredictable and vulnerable to outside interference.

- **Lack of Physical Boundaries:** Because of the nature of the radio transmission, wireless networks expose a wider, more open network; in which, transmissions can be overheard by any unauthorized users within the range. Creating secure networks through preventing unauthorized and unauthenticated access is a major challenge for wireless networks.

- **Connectivity:** Because of the transient nature of the wireless conditions, keeping the network connected at all times is a difficult task. Furthermore, mobility of the users aggravates the quality variation on the channel. In order to ensure connectivity in changing environments, the nodes might implement a set of protocols which create self-organizing, highly connected networks called mobile ad hoc networks (MANETs) [128]. In a MANET, each device acts as a router and forwards packets from other users to the intended destinations.

The link adaptation techniques target the second challenge mentioned above, and attempt to overcome the harsh channel conditions through adjusting the communication parameters such as the modulation technique to use, resulting in a multi-rate network. Regardless of how efficient these adjustments might be for overcoming the difficulties related with the channel conditions, they can have detrimental effects on the performances of other layers, such as MAC layer and network layer; exacerbating the complexity of the *shared medium* and the *connectivity* challenges aforementioned. The following sections describe such potential problems.

1.1.1 Challenges at the MAC Layer

Wireless systems might suffer from two basic factors: environmental characteristics (such as electromagnetic noise) and traffic behaviors of fellow wireless medium users referred as *contention*. Current wireless standards such as IEEE 802.11 [5] and IEEE 802.15.4 [9] define the MAC layer from contention perspective only, neglecting any potential environmental effects. In these systems, if a packet is not arrived properly, i.e. an acknowledgement is not received, the sender automatically initiates a back-off process for future transmissions. The back-off interval is increased per unsuccessful transmission, ensuring a larger transmission time for on-going communication of other peers, thus a higher probability of success in future transmissions.

Even though contention aware back-off mechanisms yield nice results in the systems where the bottleneck is the number of peers, their performance in noisy environments are not clear. In a system where the noise level (thus the packet loss [6]) is too high, a straightforward back-off mechanism does not have the ability to differentiate between noise levels and performs as if the reason for the packet loss is contention and not the environment. This incorrect perception about the channel causes the backoff windows enlarge more than the required size, wasting valuable bandwidth. Similarly, the same incorrect perception instigates some of the link adaptation mechanisms to lower their data rates prematurely [82, 145, 94], as a result of the presupposition that all the packet losses are caused by bad channel characteristics. Some of the previous works propose to use a priori channel probing before each transmission in order to isolate the packet losses caused by interference [89], or estimate the channel conditions at the receiver side [69].

On the other hand, using packet-loss as the primary metric for choosing the data rate might yield sub-optimal solutions since lowering the data rate will make the user occupy the medium more and worsen the contention status. It has been shown before that regardless of the transmission speed used, the throughput observed by

[6]High noise levels cause low SNR values, which increase the probability of a packet being received erroneously.

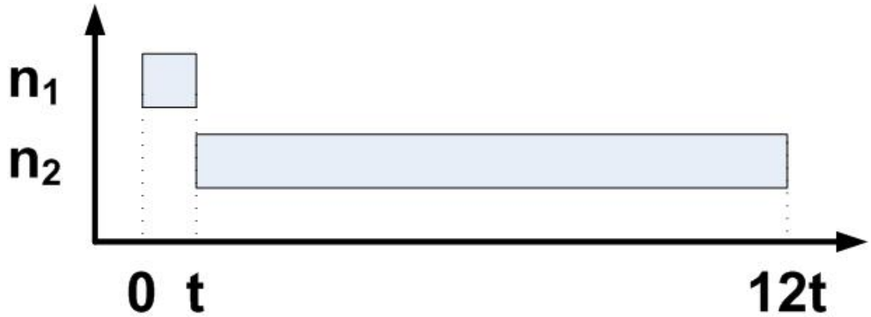


Figure 1.4: Multi-rate anomaly example. Slow user utilizes the medium for a longer amount of time in order to transmit the same amount of information as the fast user, creating a network with imbalanced medium usage times.

any wireless user in a system is limited by the lowest data rate user; as a result of the abnormality identified as *multi-rate anomaly* [67]. In an oversimplified manner, we can illustrate this anomaly with the following example as seen in Figure 1.4: if there are two users, n_1 and n_2, in a wireless system with the data rate choices 11 Mb/s and 1 Mb/s respectively, n_1 will observe throughput values as if there are 12 high data rate users in the system, whereas n_2 will observe throughput values as if there are 1.1 low data rate users on the medium. These asymmetric throughput gains create an unfair environment, in which the lower data rate users are favored over their faster counterparts.

It has been argued that the multi-rate anomaly can be resolved through distributing equal share of medium usage times (amount of time the users accesses the medium) to the users, i.e. temporal fairness [140]; rather than distributing equal share of medium access times (number of transmission attempts) which is sustained through the long-term fairness guarantee of CSMA/CA networks, such as IEEE

802.11. Some of the previous works utilize the temporal fairness concept in order to regulate the amount of time the users access the medium. However, these works either propose to use majorly modified standards [129, 68], or inject mechanisms into the wireless protocols beyond their means, such as packet scheduling [140].

1.1.2 Solution: The MORAL Algorithm

In this thesis, we describe the design and implementation of MORAL (Multi-User Data Rate Adjustment Algorithm), a novel algorithm which has the objective of increasing the throughput fairness observed in multi-rate networks through proper parameter adjustment. MORAL does not require any modifications in the widely accepted IEEE 802.11 standard and merely adjusts available parameters for changing channel conditions. MORAL targets the networks at saturation in which each user has a packet to transmit at all times and uses the *baseline fairness* principle as its guideline. This principle states that in a baseline-fair system, each user's throughput share is the same as the user would achieve in a single-rate network with all the competing nodes were running at its own rate (that we refer as *baseline throughput*) [140].

Returning to the previous example in Figure 1.4, the baseline throughput of n_1 is 1/2 packets/t, whereas the baseline throughput of n_2 is 1/22 packets/t. Comparing the actual throughput to the baseline throughput, we find that n_1 obtains only 16.7% of its baseline throughput whereas n_2 obtains 183.3%. These relative throughput values indicate that the multi-rate systems can be heavily favored towards the lower data rate users. MORAL, on the other hand, attempts to keep the relative baseline throughput percentages of the users close to each other, lessening the unjustness caused by multi-rate anomaly.

In saturated networks where traffic generation rate is high, users will observe high collision rates because of heightened contention on the channel. Consequently, users will require higher number of transmission attempts to successfully send out their packets in order to overcome the high contention situation, frequently reaching their retry limits. Our hypothesis is that, under saturated conditions, it is possible to control the contention window size through regulating the retry limits of the users. This premise constitutes the core of our proposed MORAL algorithm.

MORAL keeps its fairness promise through maintaining diverse contention window sizes for each user. Limiting the contention window sizes can be done using several mechanisms. For instance, the initial contention window size can be set arbitrarily [39] or the optimal contention window size to use might be calculated independently [68]. However, these mechanisms conflict with the MAC mechanisms defined in the IEEE 802.11 standard. In our work, on the other hand, we propose to use one of the parameters, the retry limit, in order to control the growth of the contention window without conflicting with the IEEE 802.11 standard.

The retry limit is the total number of times a packet will be attempted to be transmitted on the medium. For instance, if the retry limit value 3 is chosen for a particular packet and if the initial transmission attempt fails, the node goes into a back-off stage and after a random amount of time chosen between $[0, 2 \times CW_{min} - 1]$, the next transmission attempt takes place (this is the second attempt). If the second attempt fails, the node waits again for a random amount of time, at this point chosen from the interval $[0, 4 \times CW_{min} - 1]$. If this attempt also fails, the node reaches the maximum retry limit and announces failure informing the higher layers [7]. Clearly, if smaller retry limits is chosen, the growth of the contention window will be kept under lower bounds. However, if the retry limits chosen are too small, then the transmission attempts will not have enough randomness to avoid collisions and the main objective of the collision avoidance mechanisms would fail. Hence, picking arbitrary retry limits for users is a thorny task that has to be done carefully.

MORAL constantly monitors the other users' traffic on the medium and estimates the collision possibilities through heuristic principles. According to the data rate in use and the access characteristics of the fellow wireless medium users, MORAL calculates the best retry limit to use in order to achieve a better baseline-fair environment. As a result of the differentiation provided by MORAL for different data rate users; high speed nodes achieve more throughput as they gain additional access to the medium. Incidentally, overall throughput also improves since the medium is used more efficiently.

MORAL is a link adaptation mechanism which monitors the medium for different data rate usage and adaptively adjusts the retry limit parameter. MORAL

[7] IEEE 802.11 defines two retransmission related parameters for RTS/CTS disabled communication: SSRC (STA Short Retry Count) and dot11ShortRetyLimit. The station resets SSRC to 0 after each successful transmission, and increases it by 1 after every unsuccessful one. Once SSRC reaches the value set by dot11ShortRetryLimit, the retry attempts cease, and the packet is discarded. The default value of dot11ShortRetryLimit is 7.

can be applied independently than any other automatic data rate adaptation mechanisms, only assumes that these mechanisms are capable of differentiating the packet loss through bad channel characteristics and the packet loss through contention. This is a necessary assumption as regulating the retry limits affects the collision rate and might cause the data rate adaptation mechanisms to reconsider their data rate choices if they cannot identify the reason of packet loss. Several past studies propose mechanisms in order to detect the packet loss rate [89, 105, 69, 111]. These mechanisms are out of the scope of our work.

MORAL promises baseline-fairer, higher throughput wireless systems without any modifications in the standards. Furthermore, MORAL estimates the network conditions locally and does not require any large-scale information about the network characteristics. These features make MORAL highly distributed, very efficient and easily deployable. Through extensive simulations for a large variety of scenarios, we report that MORAL outperforms the default static retry limit case through its better usage and better allocation of a very precious commodity, wireless medium.

1.1.3 Challenges at the Network Layer

In classical wired networks, as the connectivity is secured through physical wires, the communication is not disturbed from outside interference and the transmission delivery is guaranteed as long as the wires are not damaged physically. This feature makes each physical communication reliable and identical from the performance perspective. Subsequently, the traditional wired systems create networks in which the distance between network elements is measured through the number of identical links that has to be traversed, named as the *number of hops* metric. This metric gives the number of intermediate devices that the transmission has to hop over in order to reach the intended destination.

In the wireless domain, however, as the users are mobile and the communication heavily depends on the physical distance between the users and the environmental characteristics; the definition of a wireless link becomes vaguer. In wireless networks, a link might be erroneous (if the users are far away from each other), asymmetric (one user might hear the other, but not vice versa), and transient (as the mobile users can move physically). In a previous experimental study, it has been shown that the transmission delivery rates on wireless links change diversely, both temporally on a single link and quantitatively between links [47]. Therefore, no two wireless links perform the same and using the hop metric as the main route selection criteria can be deceiving for the wireless networks.

The broadly diverse nature seen in the quality of wireless links compelled the researchers to evaluate different types of metrics for the routing process. These innovative metrics aim to enhance the routing performance via distinguishing the links according to their qualities. Among those, there are the signal strength based routing [50], high throughput route selection [18] and expected transmission count (including retransmissions) metric [46]. These metrics improve the wireless routing performance as their path creation process considers the link characteristics and transmission capabilities. However, they either do not address the link adaptation techniques used [46, 50], or their analysis are based on rather simpler approximations (e.g. no retransmission concerns [18]); otherwise they do not consider different physical loss rates for different data rates [133].

1.1.4 Solution: Data Rate and Fragmentation Aware Ad-hoc Routing

IEEE 802.11 supports a fragmentation mechanism which divides the packets into smaller pieces of units (i.e. fragments). Each fragment is transmitted and acknowledged separately. Similar to the data rate choice of the system, the fragmentation packet length has also an effect on the packet loss rate. Smaller packets have lower probability of arriving erroneously as the amount of information carried is lesser. Hence, fragmenting packets *might* yield higher throughputs in noisy environments [43].

In this dissertation, we introduce a new routing algorithm which leverages both the data rate and fragmentation size preferences in order to estimate the expected energy cost per packet. Expected energy cost is calculated through expected number of retransmissions, which is affected by both data rate and fragmentation size choices. In essence, energy cost is closely coupled with the total number of bits transferred in order to complete the full transmission and it is a good estimation for the amount of bandwidth wasted through retransmissions.

In addition, we also present a unified data rate and fragmentation aware link adaptation mechanism which adjusts these two parameters according to the existing channel conditions. Using our table-driven mechanism, the users can easily discover the best (data rate, fragment size) pair to use for their one-hop communications. Consequently, this link adaptation mechanism is used to evaluate the energy efficiency of each potential path during the route discovery phase of our routing algorithm.

We modified the well known AODV [115] protocol for the expected energy cost metric and unified data rate and fragmentation aware link adaptation mechanism. Through simulations, we have observed that our proposed routing schema, with its more efficient link adaptation and route characterization premises, outperforms the original AODV protocol immensely.

1.2 Contributions of this Dissertation

Our research reported in this dissertation demonstrates that the physical layer specific information can be valuably used within the scope of other layers in order to improve their performance. Physical information conveyed to other layers in a cross-layer manner can be used to estimate the expected performance on the highly transient and vastly variable wireless links in a more reliable manner. As a result, each layer can optimize their decision mechanisms in a better fashion through better adaptation to changing channel conditions.

Link adaptation techniques are by definition one-step mechanisms, i.e. they only adjust certain parameters according to their optimization criteria for varying conditions. They answer the questions of *What to change?* and *When to change?* in order to improve the present layer performance. Once these decisions are made, other layers are expected to function perfectly with the new communication channel conditions. In our thesis, we show that this assumption does not always hold; and provide two mechanisms that lessen the negative outcomes caused by these decisions.

Link adaptation starts at the physical layer through conventional adjustment mechanisms such as data rate regulations. These adjustments inevitably cause changes in the perceived dynamics of the other layers such as MAC and network layers. In this dissertation, we extend the link adaptation concept beyond the scope of a single layer and consider the aftereffects of this concept throughout the entire communication stack.

Our mechanisms proposed are augmented link adaptation mechanisms operating on higher layers, extending the notion of the *link* in *link adaptation*. At the physical layer, a link represents the physical interaction between a transmitter and a receiver. At the MAC layer, a link represents not only the interaction between two users, but also the neighboring users in the vicinity of the two users. Finally, at the network layer, a link represents all of the physical links traversed along an end-to-end transmission, i.e. a path. Our dissertation consists of two link adaptation mechanisms designed for MAC and network layers which adapt to the new link conditions caused by the physical layer adjustments.

Our mechanisms do not heavily modify the layering structure: we do not create new layers, or collapse the existing ones. Also, we do not violate any layer responsibilities or boundaries; each layer is still responsible for its original task. Our mechanisms merely suggest adding new interfaces to the physical layer in order to make its current parameter setting available. Other layers can use this interface in order to get an assessment of the current channel conditions which subsequently can be used in their internal optimization algorithms.

As the wireless technology becomes increasingly pervasive around the globe today, the applicability and deployment of newly proposed techniques are restricted by the current standards in effect. Thus, methods requiring major changes in the existing systems are not practical for backward compatibility reasons. Moreover, (aside from some purposely engineered networks) wireless networks are very distributed in nature and any mechanisms requiring special infrastructures and centralized entities are not applicable to most wireless use cases.

Our mechanisms proposed have the following advantages:

- **Standard Coherence:** Our methods do not propose any changes in the current standards and can be easily used with the current devices without any hardware changes. Instead of altering the underlying algorithms in the already widely accepted standards, we use the flexibility given through parameter adjustment.

- **Distributedness:** In our mechanisms, we neither require any centralized control entity nor entail holistic view of the system. Our mechanisms are entirely distributed and can be applied in an ad hoc fashion.

- **Adaptability:** Our mechanisms can adapt to highly changing environments resourcefully.

Our first mechanism focuses at the unfairness issue caused by the link adaptation techniques. We show that positive differentiation through assigning more resources to more efficient high data rate users lessens the unfairness caused by careless data rate adjustment. Out method has the following benefits:

- **Improved Fairness:** Each user attempts to take the *fair* share among the resources available and not more. The fairness criterion is defined relative to the *baseline throughput*, i.e. the system is fair when all the users achieve the same ratio of the throughput value that they would observe in a hypothetical world where all the users are identical to the current user.

- **Improved Throughput:** Differentiating the highly capable users over the lesser capable has the incidental effect of the increased throughput. As the high data rate users observe more of their capabilities in a fairer environment, their individual contribution to the overall performance increases.

- **Wide Applicability:** The method we propose is orthogonal to the link adaptation techniques used and can be applied independently than any underlying mechanisms.

Our second mechanism focuses on the link quality variability issue caused by the inherent physical characteristics of the wireless links. We show that differentiating the links according to the physical parameters used for the communication improves the performance of the routing process. Out method has the following benefits:

- **Improved Energy Efficiency:** The routing paths constructed are chosen in the most energy-efficient way. The path construction process is improved through consideration of the physical parameters set by the link adaptation techniques.

- **Improved Transmission Performance:** Our method utilizes two important transmission related parameters jointly in order to better adapt to the changing channel conditions.

- **Wide Applicability:** The method we propose is independent than the actual routing algorithm in use and can be applied to most of these algorithms with minimal change.

1.3 Dissertation Structure

This dissertation is structured in the following way: In Chapter 2, we briefly describe the rationale behind the link adaptation techniques in wireless networks and present the details of the IEEE 802.11b standard relevant to our research. Chapter 2 also surveys the literature for the state of the art link adaptation techniques. Chapter 3 begins with the details of the multi-rate anomaly and presents our analytical model for the multi-rate networks with arbitrary retry limit assignments. Next, our solution for the multi-rate anomaly, MORAL algorithm, is presented with extensive simulations, demonstrating the effectiveness of the proposed mechanism. Chapter 4 presents our next solution, Data Rate and Fragmentation Aware Ad-hoc Routing protocol, designed for creating energy-efficient paths in link adaptable networks. This chapter also describes a novel link adaptation technique, Combined Data Rate and Fragmentation Size Link Adaptation, which adjusts two transmission parameters together in order to better adapt to changing channel conditions.

This dissertation ends with conclusions and future directions in Chapter 5.

Chapter 2

Background and Related Work

2.1 Background

2.1.1 Link Adaptation in Wireless Networks

Wireless systems, as opposed to their wired counterparts, are very much prone to ambient link characteristics present in the environment. Harsh noise environments, such as factory buildings, might damage the performance of a wireless system significantly. In order to lessen the effects of these problems inherent to the environment, modulation and protocol related parameters can be adjusted accordingly. Following real-life analogy presents the rationale for this adjustment mechanism.

Assume that Bob and Alice are talking on cell phone: Bob is telling Alice where she can find the best sushi in DC. Supposing that there is no technology related problem, i.e. no low interception, etc.; both Bob and Alice will keep a reasonable level of voice and both parties will understand each other without any problem. Now, assume that just before Bob is going to tell the street name, "Tier St." in our example, there is a convoy of trucks passing next to Alice. In this case, Alice will have difficulty hearing what Bob is saying, and most probably want him to repeat it again. Bob, getting fed up with these kind of request from female part, gets angry and says "Look, it is T-I-E-R, Tier". If Alice is still unsuccessful in hearing the street name, Bob deploys another strategy and emphasizes each letter with better detail: “It is T of Texas, I of Iowa ...”.

This example illustrates the effects of a link condition on communication performance from noise perspective. Our single link between Bob and Alice suffered from high noise on the receiver side. The sender, then, adaptively changed his communication parameters in order to restore the minimum standards needed for a healthy communication. Bob, first, changed his communication body length, i.e. fragmentation size: he used to talk by emphasizing per word, but seeing that it is not enough, he started emphasizing per letter. Seeing that this change is not enough, he changed his modulation technique: he encoded his items to transmit (i.e. letters) not only as single letters, but as initials of states. This redundant information strategy which actually transmits a state name rather than a single letter, in the end improved the performance since it became easier for Alice to understand what Bob was trying to tell.

Previous example illustrated a real-life situation where link adaptation is needed in order to overcome noise caused by a real-life element: a truck convoy. In wireless networks, on the other hand, noise comes from three main areas: Firstly, ambient noise coming from other electronic devices causes a major increase in the overall noise level. For example, IEEE 802.11 shares 2.4GHz band with other appliances such as microwave ovens. Secondly, fading issues related with wireless communications might affect the devices in the same way as ambient noise. And lastly, since wireless channels are not exclusive and a communication that is going through two devices might affect several other third-party devices, coverage area

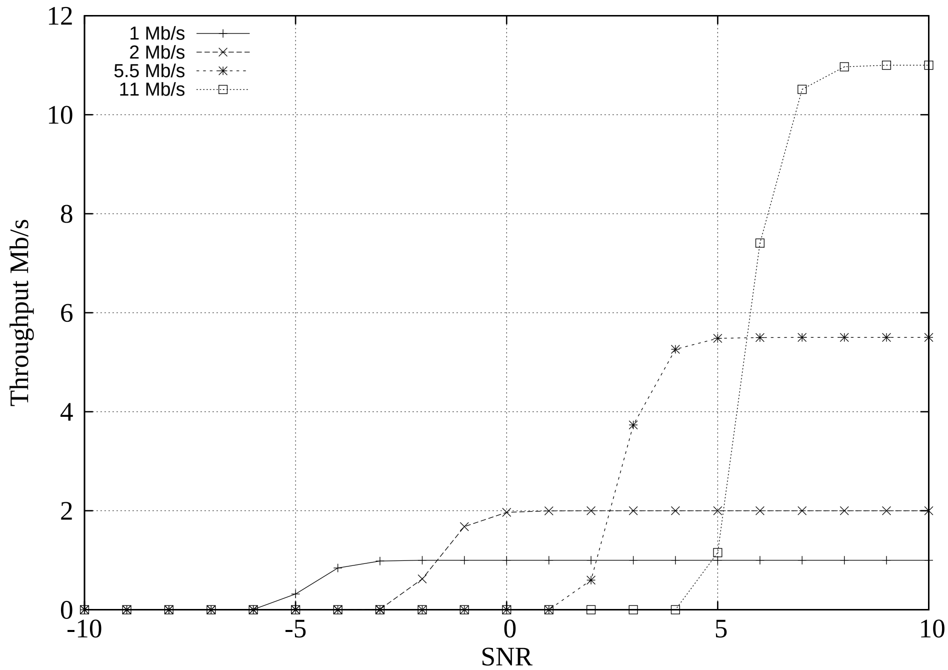


Figure 2.1: Effective physical throughput capacity for different data rates of IEEE 802.11b. Higher data rate choices are more vulnerable to low values of Signal-to-Noise ratio.

of a wireless device puts additional complexity on the noise analysis of a wireless system.

Quality of the received signal is often characterized using the ratio of the signal power to the noise power corrupting the signal, termed as *Signal-to-Noise Ratio* (SNR); and usually expressed in logarithmic decibel scale. The higher the SNR value, the better the quality of signal is.

Figure 2.1 illustrates the physical throughput sensitivity of the four different data rates that IEEE 802.11b supports, for changing channel conditions. Visibly, higher data rate transmissions require very good channel characteristics (thus very high SNR). Lower data rate transmissions, on the other hand, are less susceptible to bad channel conditions and can perform well in larger SNR intervals. Link adaptation techniques exploit this feature and attempt to adjust the transmission parameters, such as data rate to use, through estimating the channel conditions.

Given the popularity and wide market acceptance of WiFi products [109], we have chosen the widely used IEEE 802.11 standard as our experimental infrastructure. However, the mechanisms proposed in our dissertation are not restricted to this standard only, and can be integrated with any other CSMA/CA based standard as well. As one of the earlier versions of this standard, IEEE 802.11b, is still in use today and as this early version is supported by the newer versions for backwards compatibility reasons, we have chosen the IEEE 802.11b standard for our studies.

In the following sections, we briefly describe the IEEE 802.11b [5] protocol we use in our experiments [1].

2.1.2 IEEE 802.11b Protocol Description

IEEE 802.11b standard defines the physical and medium access mechanisms of a WLAN and leaves other choices related with higher layers to the vendor. An IEEE 802.11b network can be formed in two ways: *infrastructure* or *ad-hoc*. Infrastructure mode is designed to fill the connectivity gap between wireless networks and existing wired networks. In this mode, a logical portal which belongs both to the wired and the wireless networks is used to integrate these two infrastructures. All of the other network devices (nodes) are directly connected to this logical portal and use

[1] For an in-depth explanation of IEEE 802.11 related standards, please see [57].

the portal for accessing the wired network, thus the portal takes the name *access point* (AP). Note that every communication goes through the access point, even though the destination of a packet is directly reachable from the sender. AP acts as the control body of the system and carries out the services related to association, authentication, privacy and message delivery.

When an AP and at least one more device come together, they form a *Basic Service Set* (BSS). Similarly, several BSSs may form an *Extended Service Set* (ESS), adding a new level to the hierarchy. Devices are expected to move transparently between BSSs of the same ESS. A BSS can be formed without the centralized support of AP, as well. In this case, it is called an *Independent Basic Service Set* (IBSS) or an ad-hoc network.

In ad-hoc mode, devices come together and form a network without any infrastructural support. In this mode, as devices are expected to be highly mobile and there is no central body for organizing the infrastructure, the devices are required to form their own communication architectures by connecting to their neighbors within their communication range.

Even though the associated higher layer requirements might differ vastly, both of these network modes use the same physical and medium access layers defined by IEEE 802.11b. Following is the description of these two layers.

Data Rate	**Code Length**	**Modulation**	**Symbol Rate**	**Bits/Symbol**
1 Mbps	11(BarkerSequence)	BPSK	1 MSps	1
2 Mbps	11(BarkerSequence)	QPSK	1 MSps	2
5.5 Mbps	8 CCK	QPSK	1.375 MSps	4
11 Mbps	8 CCK	QPSK	1.375 MSps	8

Table 2.1: IEEE 802.11b Physical Layer Specifications.

2.1.2.1 PHY Layer

The physical layer deals with radio signaling related issues such as modulation, coding, error correction, carrier sense and clear channel assessment. This layer hides the radio chip details from the higher layers by supplying simple transmit, receive and channel assessment interfaces.

IEEE 802.11b physical layer uses 2.4 GHz band and specifies 14 frequency channels. However, FCC allows only channels 1 through 11 within the US; whereas in Europe channels 1 through 13 are available. In Japan, all channels are usable [2]. Notice that, in US, only three of these eleven channels are mainly used because of overlapping issues.

Four different coding schemas are supported by IEEE 802.11b yielding to 1Mbps, 2Mbps, 5.5 Mbps and 11 Mbps data rates. All of these schemas use *Direct Sequence Spread Spectrum* (DSSS) which spreads the signal across a larger band-

[2]Original IEEE 802.11 standard [5] and the amendment for IEEE 802.11b extension [6] define only channel 14 as usable in Japan, however in October 2001, a second amendment [7] was announced which extended the operating range in Japan to include channels 1-13 as well.

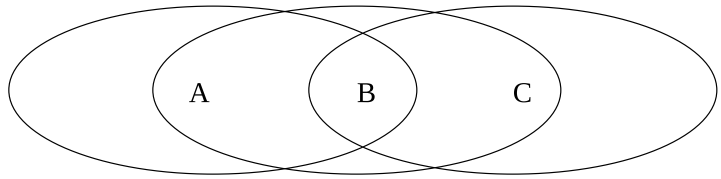


Figure 2.2: Hidden-node problem: Node C cannot hear the transmissions from node A to node B and regards the medium as free, causing collisions at node B.

width by sending redundant bits in order to overcome channel noise. Even tough higher data rates offer higher bandwidth utilization; they are more sensitive to noise levels because of more complex coding techniques. IEEE 802.11b standard does not specify any mechanisms for finding suitable data rates for a given system, and leaves this decision to the devices themselves. Table 2.1 summarizes the physical choices supported by IEEE 802.11b.

2.1.2.2 MAC Layer

IEEE 802.11b standard offers two schemas for MAC layer: (Distributed Coordination Function) DCF and (Point Coordination Function) PCF. DCF is mandatory and proposes contention based access. PCF, on the other hand, is optional and ensures QoS requirements of its users by assigning guaranteed time slots to them via a coordinator, usually the access point in the infrastructure mode. Since PCF is not mandatory and not widely used, we turn our attention to DCF.

DCF organizes the access mechanism in a distributed manner through two different methods: physical carrier sensing together with binary exponential back-off (BEB) based collision avoidance, and collision avoidance with four way control handshaking. The latter collision avoidance method is used in order to minimize the effects of hidden-node problem [3]. As illustrated in Figure 2.2, when A is transmitting to B, C is not aware of the transmission since it is out of the range of A. So, if C initiates a transmission during an ongoing communication from A to B, a collision will occur at B. IEEE 802.11b solution for this problem is a derivative of MACA [83] protocol and involves a four way handshake: RTS-CTS-DATA-ACK.

Actual algorithm can be summarized as follows: a user that has a packet to send monitors the medium for a DIFS (Distributed Inter-Frame Spacing) time and waits for a random number of slot times. This number is decremented when the channel is idle and it is frozen when the medium is occupied. When the number expires, the user tries to obtain the medium access by sending an RTS (Ready to Send) packet. When the intended destination receives this packet, it replies with a CTS (Clear to Send) packet. These two packets have embedded information about the length of the medium occupation needed for this data transfer. All the other neighbor nodes which hear these two packets back off for the period of time indicated in the packets. This schema is called Network Allocation Vector (NAV) and eliminates hidden-node problem a great deal because it reduces the data collision cases to RTS collision cases which are less appalling because of the short frame length of these control packets. This message exchange procedure is illustrated in Figure 2.3. Note that, in IEEE 802.11b, all the data packets transmitted have to be acknowledged separately. If a higher data rate choice is utilized in a transmission; the RTS,

[3] A hidden node is one that is within the range of the intended destination but out of range of the sender [60].

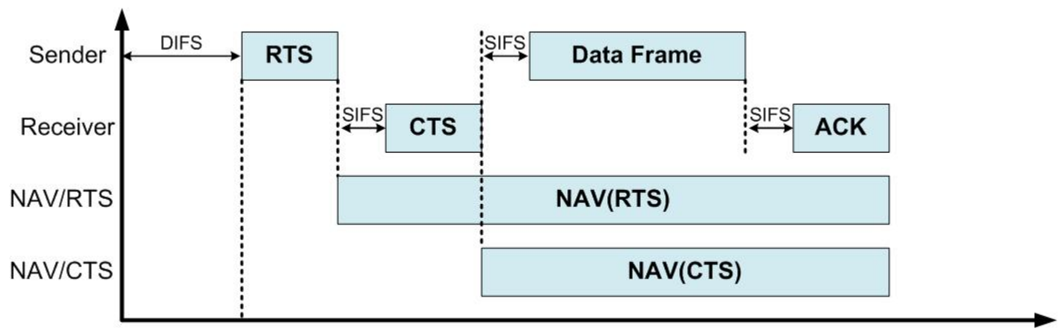


Figure 2.3: IEEE 802.11b DCF message exchange.

CTS, ACK packets and certain physical and MAC headers of the data packet are transmitted using the base data rate (typically 1 Mb/s) for backward compatibility reasons. Finally, RTS/CTS packet exchange is optional and the transmission can start directly with a data packet as well.

In case of failed transmissions, the transmitter waits for a period before trying to retransmit. This wait period is determined in terms of predefined constant slot times ($20\mu s$ for Direct Sequence Spread Spectrum (DSSS) [4]). Number of time slots to wait is selected as a random number with a uniform distribution in the interval $[0-w)$, where w is CW_{min} (32 for IEEE 802.11b) in the beginning and is doubled per unsuccessful transmission until it reaches CW_{max} (1024 for IEEE 802.11b). After

[4]IEEE 802.11 standard supports two modulation techniques: Frequency Hopping Spread Spectrum (FHSS) and DSSS. FHSS deploys the idea of switching the radio frequencies during a transmission which consequently minimizes the effects of interference and decreases the vulnerability of the transmission to interception. DSSS, on the other hand, divides the information to be transmitted into small pieces using a chipping code and assigns these pieces to a frequency channel across the spectrum. The redundant chipping code helps the signal resist interference and increases redundancy.

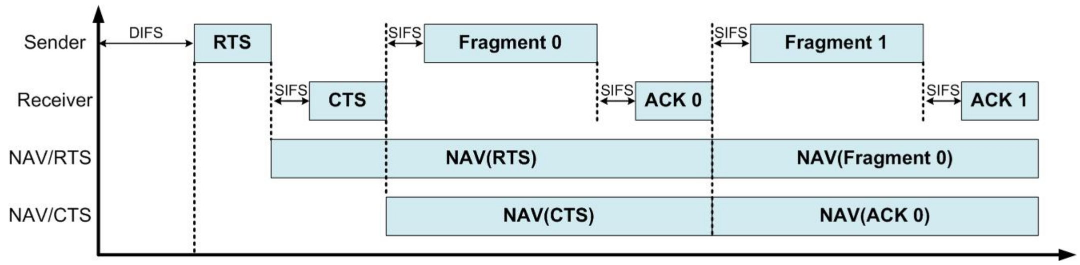


Figure 2.4: IEEE 802.11b DCF message exchange with fragmentation.

every successful transmission, the window size is reset to CW_{min}.

The packets exchanged can be prioritized through different inter-frame spacing selections. After the transmission of the data frame, in order to avoid other users accessing the channel and a potential collision of a new transmission with the ACK packet of ongoing transmission, a shorter inter-frame spacing (SIFS) is used.

IEEE 802.11b supports packet fragmentation for increased reliability and better resistance to higher levels of interference. Larger packets have higher probability of being received erroneously, since all of the bits in a packet have to be received correctly by the destination in order to complete the transmission successfully. Fragmenting each packet into smaller entities (i.e. fragments) and acknowledging each fragment separately improve the probability of successful transmission. A typical message exchange with fragmentation mechanism is shown in Figure 2.4.

2.2 Related Work

Link adaptation strategies are discussed extensively in wireless community for diverse number of systems. For example, in [108] Nanda discusses different link adaptation techniques in cellular networks, in [15] Alonso proposes a new cross-layer rate adaptation schema for CDMA-based packet switching networks and in [79] Gass describes a code-rate adaptation mechanism for FHSS systems. Similarly, in [102] Merz suggests a mechanism for UWB band ad-hoc networks and in [87] Kim proposes a receiver-based, time fair data rate adjustment mechanism for TDMA based 802.15.3 networks.

Even though these mechanisms are effective in their own domains, they are usually not usable in CSMA based WLAN systems. IEEE 802.11 based networks neither enjoy highly adjustable (but very complicated!!) signal processing techniques as in CDMA-based networks, nor have the luxury of assignment-based TDMA access mechanisms. Hence, we concentrate on the mechanisms which are related to 802.11 based networks.

2.2.1 Data Rate Adjustment Mechanisms

Data rate adjustment algorithms can be categorized into two classes according to the station where the decision is made. In sender-based mechanisms, the transmitter decides on the rate choice and usually exploits some sort of statistical analysis based on previous transmissions and transmission attempts. In receiver-based mechanisms, on the other hand, the receiver selects the data transmission rate and a loop mechanism for notifying the sender about receiver‘s decision is deployed. The latter mechanisms are more suitable for asymmetric channels since in such environments, information based on the transmitter will not reflect the status of the receiver promptly.

One of the earliest works for sender-based data rate adjustment mechanisms, Automatic Rate Fallback (ARF) [82] algorithm is based on the number of missed ACKs. If two consecutive ACKs are missed during a transmission, data rate is lowered and a timer is started. If either the timer expires or the number of successfully received ACKs reaches 10, the device attempts to send at a higher rate. If the very first attempt to transmit at a higher rate (probe packet) is failed, the system rolls back to the lower rate and resumes normal operation. ARF is easy to implement and requires no modifications in the standard, but it has several drawbacks. First of all, it is not fast enough for dynamic environments and it has to wait for the timer to expire or the number 10 to be reached for a data rate increase attempt to occur. Secondly, if the conditions are stable, the algorithm still tries to send at higher rates periodically resulting in wastage of resources. And thirdly, ARF estimates the direction of changes only and is slow to adapt under diverse data rate selection systems. The reason for the last item is the binary nature of the algorithm. At a given time, ARF might give the status of the channel as "good" or "bad". If the channel conditions vary greatly and large scale data rate adjustments are needed, ARF will provide enough feedback for the direction of the change but will not inform about the amount of change.

In order to make ARF more adaptable, Lacage proposed Adaptive ARF (AARF) in [94]. AARF uses a Binary Exponential Back-off (BEB) mechanism for adjusting the number of successful ACKs threshold present in ARF. In this schema, if the transmission of a probe packet fails; the transmission rate rolls back to the lower rate and the threshold is multiplied by 2 (max 50). If the reason for data rate decrease is two consecutive missing ACKs, the threshold is set back to 10. This method increases the period between successive failed attempts to use a higher data rate.

Pang modifies ARF for collision differentiation and proposes Loss-differentiating ARF (LD-ARF) in [111]. The main motivation in this work is that the missing ACKs do not differentiate if the transmission failure is because of the erroneous channel conditions or because of collisions caused by contention. If the data rate is decreased because of a missing ACK arising from a collision, data rate decrement will make the system worse since lower data rates capture a longer time on the medium. Moreover, if two packets with two different data rates collide, both of the parties will see the same collision period determined by the lower rate. Hence, the reason of the transmission failure should be known so that a healthy data rate adjustment decision can be made. This phenomena is discussed and verified through experimentation in [42] as well. LD-ARF uses RTS/CTS mechanism of IEEE 802.11 networks. Since these packets are small and they are always sent at basic rates, the probability that these packets will be received erroneously is very small. Thus, if a CTS packet is not received by the sender, it is decided that a collision is occurred. LD-ARF also proposes to use a NAK mechanism in order to increase the feedback quality. In this schema, the MAC header of the data frame is protected through a checksum such that the receiver can recognize if the header is erroneous. Since the probability that a short MAC header received erroneously is small, a receiver can capture the MAC header correctly most of the time and identify if it is erroneous using the checksum.

If the receiver sees a packet destined for itself and the packet is erroneous, it reports back to the sender using a NAK. A NAK packet is same as an ACK packet except a one-bit frame type field difference.

A similar effort for differentiating between collisions and erroneous frames is presented in [89] by Kim. In this work, Collision Aware Rate Adaptation (CARA), authors use two mechanisms: RTS/CTS probing and Clear Channel Assessment (CCA) detection. First mechanism relies on RTS/CTS exchange mechanism‘s ability to detect erroneous links as in [111]. However, since this mechanism wastes precious bandwidth, CARA utilizes this option by enabling it when transmissions start to fail. Their mechanism deploys a threshold for the number of failed transmissions; if this threshold is exceeded, RTS/CTS probing is enabled. As a second mechanism, they propose to use CCA for heterogeneous users. In heterogonous user environments, medium occupancy durations of different data rate transmissions differ. In case of a collision, the higher data rate user finishes transmitting its packet earlier. Hence, if the higher data rate user listens the channel immediately after its transmission (before ACK, during SIFS), it may observe the channel busy because of slower user‘s ongoing transmission. The authors claim that the effects of these systems get stronger as the number of users in the system increases because of heightened probability of collision. Essentially, CARA describes “when to decrease?” the data rate and leaves “when to increase?” question to other works.

Chen et al. further extends the RTS/CTS probing schema and proposes a probabilistic RTS/CTS enabling mechanism through estimating the collision probability of the network using the historical transmission information. Following the probabilistic nature of their schema, the authors name their mechanism as Probabilistic-Based Rate Adaptation (PBRA) [38].

In a more recent work [80], Judd et al. reports on the Channel-aware Rate Adaptation Algorithm (CHARM), which eliminates the usage of RTS/CTS packages altogether and proposes to use the packets overheard instead. Using the overheard packet properties, CHARM estimates the path loss assuming reciprocity and combines this information with the success rate of its transmissions in order to select the best data rate to use.

In a similar effort titled Robust Rate Adaptation Algorithm (RRAA) [150], Wong et al. mitigates the RTS/CTS inefficiency issue through an adaptive RTS window, which keeps track of the recent RTS and frame losses in order to enable or disable the RTS/CTS mechanism. RRAA utilizes the frame loss information gathered during a short-term time window as opposed to the probing frames which either succeed or fail, and do not provide dependable information.

BEWARE (Background Traffic-Aware Rate Adaptation) [147] shows that the background traffic affects the throughput performance considerably, and proposes to use a mathematical model to calculate the expected packet transmission time of each data rate, estimating the other wireless devices' transmission probabilities.

In Loss Differentiated Rate Adaptation (LDRA) [27], Biaz et al. propose to use the 802.11 beacons that are already periodically being transmitted in order to estimate the initial data rate in a transmission attempt. LDRA utilizes the basic data rate for the retransmissions in order to differentiate between the packet losses caused by congestion and packet losses caused by erroneous channel conditions.

In [32], Braswell defines a Frame Error Rate (FER) based schema. In this schema, if FER exceeds some threshold and the current rate is not the minimal rate, then the transmitter switches to the next lower rate. If the FER falls below another threshold, a few probe packets (usually only 1) are sent at the adjacent higher rate. If all these frames are acknowledged, switching to the higher rate occurs. In order to prevent the algorithm from oscillating between two adjacent rates, upscale action may be prohibited for some time after a downscale decision. It is difficult to find reasonable values for the parameters defined in this mechanism. The optimal settings depend on the current link condition but are generally fixed at design time. Moreover, there is a clear tradeoff between the quickness of the mechanism that might be adjusted by changing the window size and the reliability of the FER measurements. Hence, in this schema, many frames are transmitted at a non-optimal rate.

In [40], Chevillat proposes a dual threshold mechanism. In this schema, the transmitter maintains two counters, one for successful transmissions and one for failed transmissions. If a frame is successfully transmitted, success counter is incremented and the failure counter is set to zero; similarly, if a frame transmission fails, the failure counter is incremented and the success counter is set to zero. If the failure counter reaches to a certain threshold, the data rate is decremented and the counter is reset to zero; similarly if the success counter reaches to a certain threshold, the data rate is incremented and the counter is reset to zero. As a failure threshold, Chevillat proposes to use 1 (one). Success threshold, on the other hand, is susceptible to errors. Fast-changing channels require a small success threshold in order to adapt quickly; slow changing channels require a large success threshold in order to avoid ineffective switching to higher rates when the channel has not improved. In order to deal with this problem, a sub-threshold mechanism is proposed. In this mechanism, two possible thresholds are determined, one for fast changing channels and one for slow changing channels. After the current success threshold is reached, transmitter switches to higher data rate and sends one probe packet. If this transmission is successful, then it is assumed that the link condition is improving rapidly and the next success threshold is set to the small value. If, however, the probe transmission fails, it is assumed that the link quality is either changing slowly or not at all and the success threshold is set to the larger value. This mechanism is extended in [77] for power adaptation. In this work, it is proposed that if power utilization is critical, initially data rate is kept fixed and transmission power level is adjusted. If minimum power level is reached, further data rate is adjusted.

Receiver-based data rate adjustment methods rely on the famous work of Holland [69]. In this work, Holland describes the Receiver-based AutoRate (RBAR) mechanism which enables receivers transmitting the desired data rate of a transmission through RTS/CTS frame exchange. RBAR‘s main motivation is that the best information is at the receiver. RBAR uses SNR readings at the receiver and finds the best data rate that coincides with the current transmission. In order to handle this, RBAR uses modified RTS/CTS frames such that they include (rate, frame length) fields rather than the duration field. Receiving an RTS frame with the proposed data rate, the receiver estimates the data rate suitable for this transmission and sends back the CTS frame with the new data rate. The third-party nodes hearing RTS/CTS packets update their NAVs accordingly. Notice that in case the receiver does not accept the data rate proposed in the RTS frame, the new rate has to be announced to the receiver-side hidden nodes. This is handled by adding a new header RSH (Reservation Sub Header) to the PLCP header of the data frame. Third-party nodes extract RSH from the data frame and update their NAV. Even though RBAR is expected to be more accurate, continually using RTS/CTS packets even without the hidden-node situation wastes considerable amount of bandwidth. Furthermore, RBAR is not compatible with current IEEE 802.11 standard.

Notice that SNR-based systems (such as RBAR) are expected to respond very fast to changing link conditions, however due to the uncertain and fluctuating relation between SNR information and BER (Bit Error Rate), they lack stability and reliability. The statistics based approaches (such as ARF), on the other hand, give robust performances but are very slow to react the changes. In order to find a solution to this trade-off, Haratcherev proposed a hybrid mechanism [62, 63]. In this schema, for each data rate, three SNR thresholds are calculated: two for stable conditions (high and low), and one for low threshold for volatile conditions; volatile low threshold being larger than the stable threshold. The detector observes the differences between three consecutive SNR readings of ACKs. If both differences have the same sign and their sum exceeds a certain threshold, then the volatile low threshold is used since conditions are changing fast and consistently. Otherwise, the stable threshold is used. This schema is supported by a statistics based method as follows: For each data rate, there are two counters, one for successful packets, and one for failed packets. If the number of successful packets exceeds a threshold, this means that the SNR threshold value for that transmission is too high. Similarly, if the number of failed packets exceeds a threshold, SNR threshold for current communication is too low. The latter mechanism ensures that the SNR threshold values correspond to the FER values estimated.

In [112], Pavon proposed to use Received Signal Strength (RSS) values for SNR estimation and assumes a linear relationship between two. The data rate adjustment is made when the RSS values pass some thresholds. For each data rate, their method keeps corresponding thresholds which are changed dynamically. For example, if a transmission attempt at a particular rate is unsuccessful, the threshold for that rate should be subsequently raised. For the rate selection, a station considers the average RSS value which is taken in a time depending manner. The algorithm will automatically decrease the rate when the number of retransmission attempts exceeds the retransmission limit. After every retransmission, the thresholds are adjusted. Notice that RSS values are captured from the packets received.

Qiao in [120] proposes a Signal-to-Noise-Ratio (SNR) based data rate and fragmentation size selection for goodput enhancement in IEEE 802.11a based networks. Their mechanism assumes that the transmission link is symmetric and the SNR estimation at the receiver coincides with the situation at the receiver. Their results show that fragment size adaptation does not have an effect for goodput enhancement. They extend their work in [123] for frame retry count. In this work, they propose to construct an offline table for finding the appropriate data rate for each (payload length, frame retry count, wireless channel condition) triplet. Their analysis enables per packet adaptation by allowing the adjustments taking place per retransmission. This is different than the previous works in which it was assumed that each retransmission was taking place at the original data rate. Our work in this document is similar to this study since we also create a table for matching the channel conditions to (data rate, fragmentation size) pair. However, in our study, we concentrate on the energy efficiency rather than goodput maximization. Indeed, our experimental analysis shows that fragmentation size has a big effect on the energy efficiency of a link, contrary to Qiao‘s analysis.

In [121], Qiao further defined a method for organizing the rate increase attempts. Their aim is to control the rate increase attempts such that the responsiveness is guaranteed with minimum number of attempts. Quicker increase attempts are quick to react to changing conditions, but their failure rates are high because of premature decisions. Their mechanism relies on a delay factor that is determined by the application. As long as the delay factor bound is not violated, the mechanism allows transmitting at the current data rate. They also set the maximum consecutive successful frame transmissions at the same rate to 50 in order to avoid potentially long response delays when the wireless channel condition changes slowly.

Qiao also proposed a hybrid data rate/power level adaptation schema, MiSer (Minimum Energy Transmission Strategy) in [122]. Their mechanism relies on an offline table as in their previous work [120], which maps (payload length, path loss, RTS retry count, data retry count) to the appropriate (data rate, transmission power) combination. Their table calculation is based on the theoretical analysis of the link condition and requires information about the number of stations in the system and transmitter power levels. For the latter parameter, they propose to use IEEE 802.11h's TPC mechanism. They also propose to send CTS frames using the highest transmission power in order to eliminate aggravated interference caused from hidden nodes.

In [153] Wu et al. propose BARA, a sender based rate adaptation mechanism which uses the periodic beacon messages for the initial data rate selection and a weighted averaging technique for the ongoing transmissions. Their work also proposes to transmit the retransmissions using the basic data rate in order to better assess the varying channel conditions.

Madwifi Project [2], which implements a multiband driver for Atheros chips [1], uses three bit-rate selection algorithms. The first algorithm, Onoe, uses a simple approach of increasing the bit-rate when less than 10% of packets require a retry and decreasing the bit-rate when packets need at least one retry. AMRR [94], on the other hand, uses binary exponential back-off mechanism similar to AARF in order to handle short-term variations and uses a different threshold for each data rate in order to handle long-term variations. Lastly, SampleRate [28], periodically sends packets at bit-rates other than the current one to estimate the instance another bit-rate will provide better performance. In this schema, small fraction of the data is sent at the two adjacent rates. Performance values of all three cases are compared (usually throughput) and the rate with the highest performance is selected. Experimental analysis and comparison of Onoe, AMRR and SampleRate protocols can be found at [110].

TARA (Throughput-Aware Rate Adaptation) [16] is an extension of the Madwifi Project and it utilizes the estimated congestion status of the channel for rate adaptation. TARA combines the per-packet transmission times and the number of retransmissions required for the packet to be transmitted successfully, in order to estimate the channel status.

In [152], Wu describes a potential enhanced hidden node problem if High Rate (HR) PLCP headers are used and proposes a modified Network Allocation Vector, NAVp, in order to solve this problem. NAVp reserves the channel conservatively for the slowest data rate if information can not be gathered from HR headers.

2.2.2 Fragmentation Size Adjustment Mechanisms

Fragmentation size/packet size choice might affect the performance of a wireless system as well. Smaller packets yield to lower packet error rates (PER) since the information carried in those packets are less. However, using smaller packets increases the overhead since each packet has a constant header associated with it. Larger packets, on the other hand, are more vulnerable to link errors and might cause several retransmissions. Clearly, there is a tradeoff between using small, high overhead packets and large, high PER packets.

In one of the earliest efforts [103], Modiano attempts to dynamically optimize the packet size based on the estimates of the channel bit error rate. The author describes a mechanism to estimate the bit error rate (BER) from the number of failed transmissions in a time window. For high bit error rates, his estimation is very accurate, however, for low bit error rates, larger time windows are needed. Even though the estimation presented can give reasonable results for slowly changing channels, their BER estimation is not precise for highly variable channels. Performance of this algorithm is best when channel conditions vary relatively slowly to the observation period.

In [44], Ci proposes a Kalman Filter approach for predicting the optimal frame size. Their method improves the prediction at the cost of processing time. They also propose an optimal fragment size for known BER values in [43].

In [144], Tourrilhes proposed to use exponential halving of fragment size. In their scheme, each packet is transmitted initially without fragmentation and for each failure the fragment threshold is divided by two until a predefined minimum fragment size.

Kwon et al. defined a metric (Error Rate Estimator) in [93] and used this metric in order to find optimal fragment sizes. Their metric relies on the actual time elapsed for transmitting a packet.

In [86] Kim et al. further extended the data rate adaptation techniques by employing fragment size as well. Their mechanism optimizes the fragment size after a rate adjustment is made by the data rate protocol. Moreover, they use per-fragment, not per-packet adjustments where consecutive fragments might have been encoded in different data rates and might be at different sizes.

2.2.3 MAC Extensions for Multiple Data Rate Networks

In wireless networks where multiple heterogeneous users exist, diversity among these users might be used to increase the performance of the system. Packet scheduling by giving higher precedence to better users increases the efficiency of the system by reducing the average delay times of the packets in queues.

In such a work [146], Wang proposes to use multicast RTSs with candidate lists piggybacked inside. Receivers calculate their response transmission times according to the position of their addresses in the RTS candidate list. Transmitter, consequently, chooses the best receiver available according to the CTS frames it receives. Clearly, list size affects the intensity of the system: bigger lists mean more diversity, but transmitter has to wait longer. Their work also supports packet bursting by sending higher number of packets to better receivers. This mechanism alleviates the effects of the Head-of-the-line (HOL) problem defined in [24] where a packet destined to a receiver which is located in a fade zone prevents other packets in the queue being transmitted. Thus, if the channel to a specific receiver is in the burst error state, all receivers suffer throughput degradation.

In a similar work [78], Ji proposes a query/reply mechanism where query includes the list of the potential receivers. Their work also recommends using packet concatenation which sends several back-to-back packets to the same receiver without waiting for ACKs. This method eliminates the extra time passed for SIFS idle periods and ACK transmissions. They support their mechanism by introducing two heuristics for improving the fairness among users.

In one of the latest works [97], Li describes Full Auto Rate (FAR) MAC mechanism. Their target is to send not only data frames but also control frames (i.e. RTS/CTS/ACK) at higher rates. As this modification affects the standard NAV mechanism, authors also propose a Modified Virtual Carrier Sensing (MVCS) mechanism in which reservation of the medium is done only for the immediate next frame instead of all the remaining frames in the sequence. For example, a third party node which intercepts an RTS packet defers until CTS passes through. Then during the data frame transmission, physical carrier sense mechanism will make this node defer. As $DIFS > SIFS$, ACK transmission is guaranteed to be contention free. Since FAR proposes to transmit RTS at higher rates, it is possible that a third party node intercepts erroneous RTS packets. In that case, the node defers for EIFS ($EIFS = SIFS + Tx(ACK) + DIFS$) amount of time which allows the transmission of control frames successfully. However, if a CTS frame is received erroneously, deferring for EIFS amount of time does not guarantee successful data frame transmission since data frame transmission time might be larger than EIFS. Thus, if data frame transmission time is larger than EIFS, CTS frames are sent using basic rates; otherwise they are sent at higher data rates. FAR also proposes to use a hybrid sender/receiver based data rate adjustment such that the sender caches the best data rate for RTS frames and this value is updated per new adjustment request that is sent by the receiver.

In their seminal work [67], Heusse et al. analyze the performance of the IEEE 802.11b networks through users with heterogeneous data transmission rates. Their results show that slow hosts may considerably limit the throughput of other hosts roughly to the level of the lower rate. Even though their work is inspiring, it does not represent a complete CSMA/CA analysis as in [26]. In [68], Heusse et al. also proposed a MAC algorithm which adjusts the contention window size in order to solve the heterogeneous user problem through ensuring time fairness among users.

In a similar work, Dunn proposed to use fragmentation size in order to ensure such fairness [52]. Also, Sadeghi proposes Opportunistic Auto Rate (OAR) method to regulate the back-off procedure after successful transmissions such that high data rate users transmit more number of back-to-back packets [129]. Wu[154] further extends OAR through consideration of the congestion level at the receiver side.

In a notable work [85], Khan et al. propose to use the application layer requirements in a cross-layer manner during the data rate selection process. Their work attempts to keep the buffer occupancy of the MAC layer at acceptable levels.

Babu et al. discuss the multi-rate fairness issue in [20] and compare the traditional throughput-based fairness with the time-based fairness. In [19], the same authors propose to calculate the optimal minimum contention window size and the optimal frame size to use in order to maximize the defined fairness metric. In similar efforts, Tinnirello utilizes the transmission opportunity (TxOP) concept in 802.11e WLANs in [143] for temporal fairness; and Qiao et al. [124] define a new priority-based fair MAC schema (P-MAC) which replaces the slotted binary exponential backoff mechanism by a new scheme which determines the optimal backoff values for the wireless stations through keeping track of the number of idle and busy slots.

In two distantly related works, fairness of the wireless networks is discussed from different perspectives: short-term unfairness of CSMA/CA is observed in [92] and the fairness issues caused by the physical layer capture is discussed in [55].

2.2.4 Cross-layer Routing Mechanisms

Traditional routing algorithms use minimum number of hop metric as their routing strategy. Even though this metric gives reasonable results in conventional wired technologies, its operation on wireless systems is questionable since wireless links are not expected to be identical. In highly dynamic environments where link conditions rapidly change, variations between link qualities can be very high and a single lossy link might affect the performance of the whole network poorly. In order to manage such irregularities, cross-layer routing mechanisms have evolved. These mechanisms use the lower layer parameters such SNR and RSSI readings in their decision mechanisms in order to create more vigorous paths.

In [50] Dube et al. describes on-demand Signal Stability-Based Adaptive Routing (SSA) which uses signal strength criterion to differentiate between strong and weak channels. SSA also uses location stability to bias the protocol choosing a channel which existed for a longer period of time. The first criterion is imposed during the route discovery phase such that nodes that receive route discovery packets ignore them if the signal strength criterion is not met. The second criterion is executed by each node identifying their neighbors as Strongly Connected (SC) or Weakly Connected (WC) according to the signal strengths of received beacons. Similarly, if a route discovery packet is received from a WC node, it is ignored.

In [47] De Couto et al. shows that using the minimum-hop metric does not work well in high loss rate environments and proposes a new metric, Expected Transmission Count (ETX)[46] which minimizes the expected total number of packet transmissions required to successfully deliver a packet. The authors argue that end-to-end delay metric is not appropriate because of varying queue lengths in intermediate nodes. Nevertheless, their metric is not appropriate for link adaptable nodes as in such nodes, high number of transmissions might be tolerated for higher throughputs switching to higher data rates.

In [133], Multi-Rate and Multi-Range Routing Protocol (MR2RP) is proposed. MR2RP estimates the expected time to complete a transmission as their metric. This metric includes the MAC delay at a link (including back-off and NAV originated delays) and buffer queuing delay estimations. Their analysis assumes that each link data rate is already chosen according to the distance between two ends.

In [130], Seok et al. describes the Multi-Rate Aware Routing Protocol which prefers high number of high data rate links to low number of low data rate links. Their metric assumes that the cost associated with a link can be characterized by taking the inverse of the data rate used on that path and ignores any retransmission attempts.

In [18], Awerbuch presents Medium Time Metric (MTM) that selects higher throughput paths and avoids long unreliable paths. MTM minimizes the total medium time consumed for sending a packet from source to the destination. They also propose to use interference graphs in order to include the affects of inter-path interferences in their computations.

In [30], Biswas et al. illustrate the opportunistic routing mechanism in which network nodes choose their forwarding nodes according to the success rates of their previous transmissions. In this strategy, minimum-hop metric is abandoned for the probability of successful transmission to the very first node on the path.

In [59], Goff et al. propose a preemptive routing protocol for ad-hoc networks. Their aim is to avoid disconnection that might happen because of path breaks. They propose to monitor RSSI values of packets received in order to identify potential path breaks and encourage to start route discovery process in advance.

In [106], Nadeem et al. proposes a fragmentation aware energy-efficient routing mechanism which finds minimum energy paths in a network where each link can adapt a different fragmentation size for its transmissions.

Link adaptation techniques are not limited to data rate, fragmentation size and power level regulations. There are further techniques for overcoming highly erroneous transmission channels such as usage of convolution codes as [99], and Forward Error Correction (FEC) based mechanisms such as [53, 142]. However, these techniques are out of the scope of our work.

2.2.5 Performance Analysis of Wireless Systems

One of the most famous works for performance analysis of wireless systems is Bianchi‘s analysis of IEEE 802.11 DCF using Markov chains in [26]. His work has been extended for distance metric in [22] and for noisy environments in [105, 36]. [34] further develops an analytical model of rate adaptive LANs, but this analysis considers rate adaptation from delay perspective only and lacks a global view of the wireless system.

Chapter 3

MAC Layer Enhancements for Multi-Rate Networks

In a wireless network armed with link adaptation mechanisms, a resulting system with high data rate heterogeneity is inevitable. For instance, in Figure 3.1, the effective data rate[1] of the WLAN users in the Portland State University cafeteria is shown [2] over a four hour time interval. Figure 3.2[2] illustrates the data rates chosen for the regular Ethernet packet transmissions extracted from the same dataset[3]. The twelve data rates supported by the IEEE 802.11g[4] [8] are easily identifiable in this figure. Notice that the data rate usage on the medium is very diverse; and at (almost) any given time, the observed traffic contains transmissions from several different data rate users.

By definition, in a fair system, each user takes the adequate amount of the shared assets available and expects the other users to do the same. In a system where all the users are identical in every aspect, fairness translates to equal division

[1] Effective data rate is the total number of bits transferred over total amount of time elapsed for transmission.

[2] Data was gathered from [117], however information extraction and visualization were prepared by the authors.

[3] Effective data rate calculation includes the header overheads, which are transmitted using the base data rates of 1 Mb/s or 2 Mb/s. Thus, the effective data rate values shown in Figure 3.2 are slightly lower than the actual physical rate.

[4] IEEE 802.11g is fully compatible with IEEE 802.11b and provides additional data rates of 6, 9, 12, 18, 24, 36, 48, and 54 Mb/s.

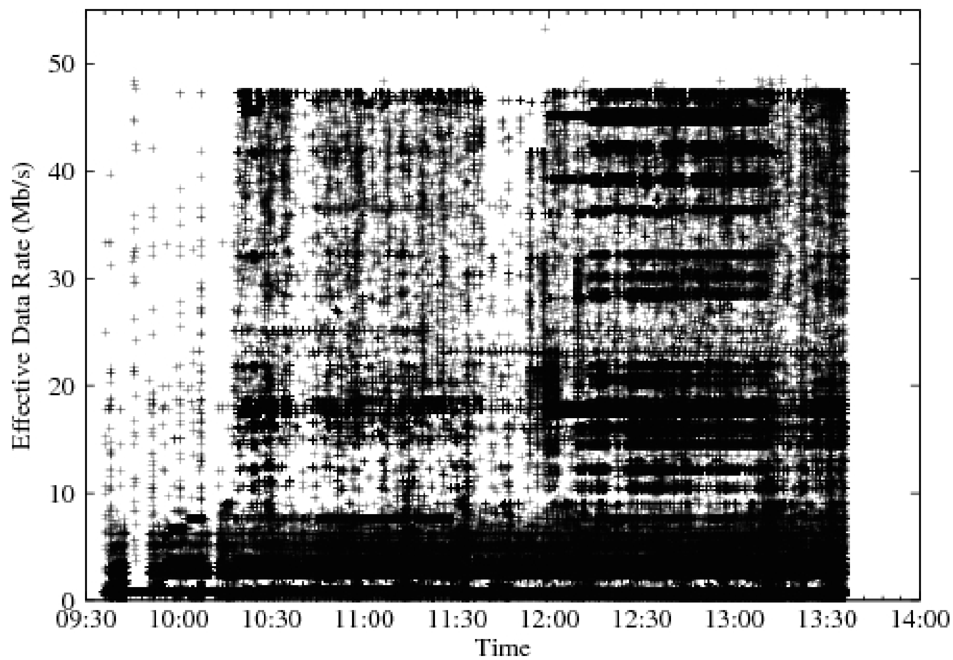


Figure 3.1: Effective throughput of the hot spot in Portland State University cafeteria, Portland, Oregon.

of all the assets. However, in systems where the capabilities of the users are different, the fairness criterion has to be redefined.

In CSMA/CA based systems, such as described in Section 2.1.2, fairness is defined assuming user equality. CSMA/CA algorithm gives each user the equal opportunity of using the wireless medium regardless of its capabilities. Hence, on the average, the number of times the users access the medium is the same. However, as the amount of transmission times is different among various data rate users; once the users access the medium, the length of their medium occupancy times differ. Larger occupancy times by the slow users leave the faster users with smaller medium shares. Consequently, performance of the higher data rate users is bounded by the

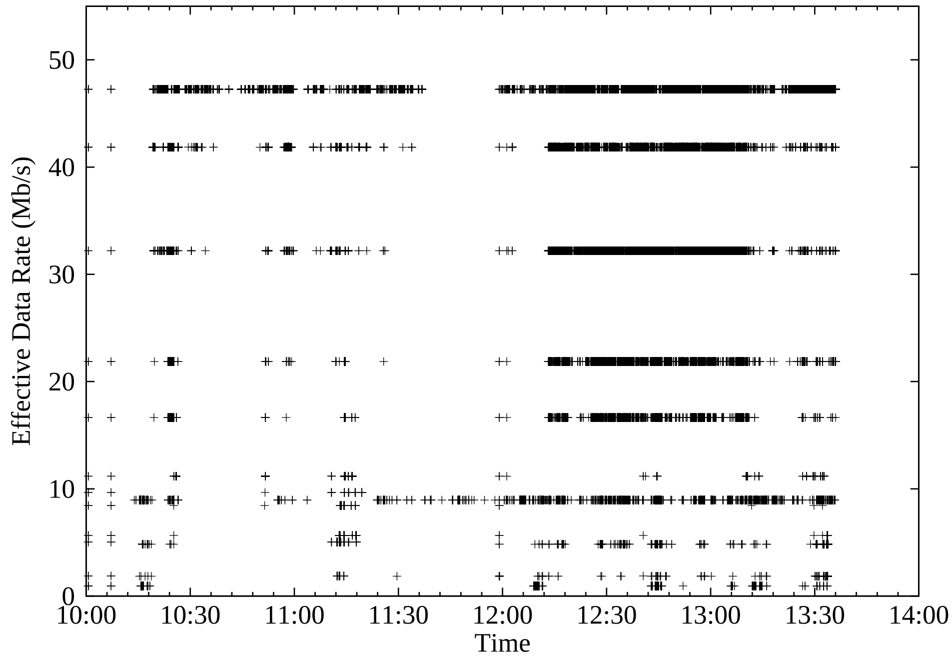


Figure 3.2: Effective throughput for the same size packets (1536 octets) observed at the hot spot in Portland State University cafeteria, Portland, Oregon.

existence of the slower ones.

Assigning the fair share to each user according to their capabilities is a challenging task. In this chapter, we analyze the multi-rate networks from fairness perspective and propose a highly distributed and adaptable method for reducing the unjustness plunged on the highly capable users by the lesser capable ones. Our method leverages one of the openly adjustable parameters, the retry limit, in order to restore the fairness in multi-rate networks under saturated conditions.

Our chapter starts with the multi-rate anomaly description and analysis. Next, we provide a metric for analytically approximating the expected throughput values of heterogeneous data rate users in order to understand how the multi-rate anomaly

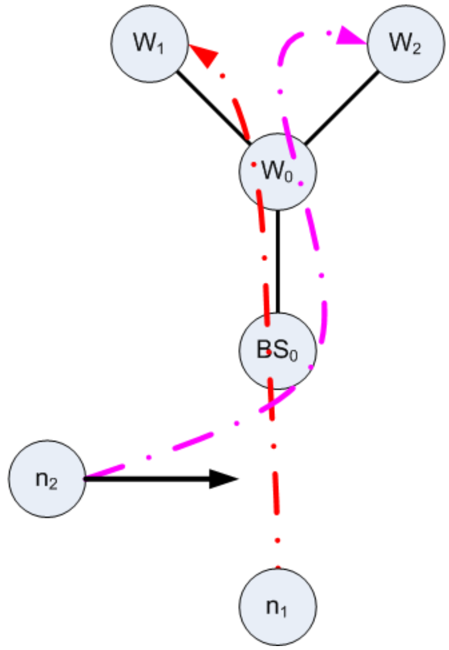


Figure 3.3: Simulation scenario created for multi-rate anomaly.

takes place. Through this metric, we provide a new mechanism, MORAL, which is capable of adjusting the available parameters for changing conditions in order to recover the unfairness caused by the unjust wireless medium share. Our chapter ends with the experimental results and discussions.

3.1 Multi-Rate Anomaly

Consider the simulation scenario illustrated in Figure 3.3. In this network, there are two wireless stations n_1 and n_2 communicating to W_1 and W_2 respectively through the basestation BS_0. Furthermore, suppose that there is only uplink transmission from the wireless stations to the wired end points. In the beginning of the simulation, n_2 is out of the range of the basestation and moves in the direction shown

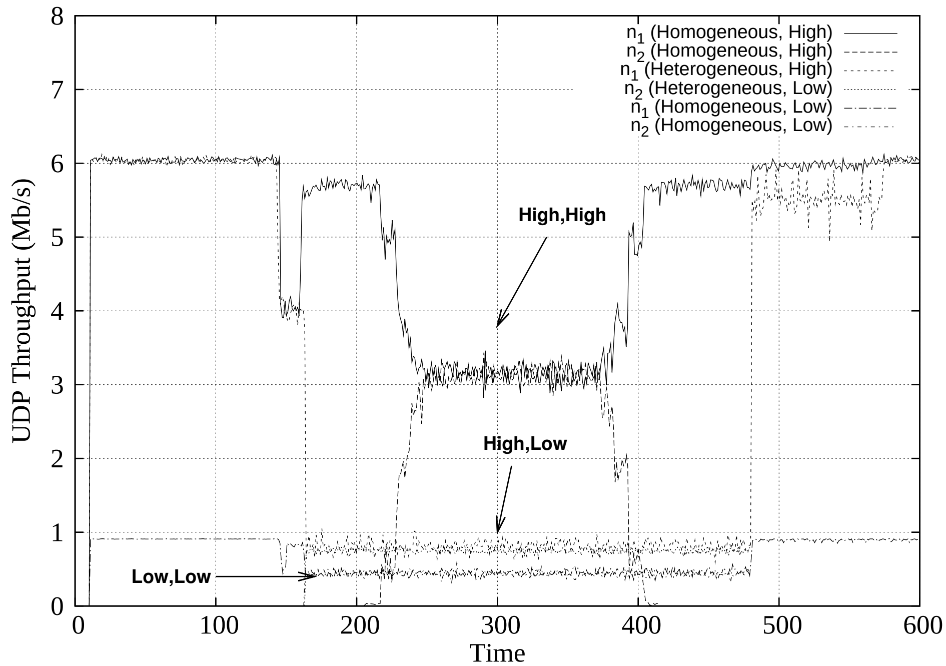


Figure 3.4: Throughput values seen in multi-rate networks anomaly simulation.

in the figure. Using this setup, we simulated three situations in order to understand the effects of different data rate users in a wireless system: 1) a homogeneous system in which both of the wireless users choose the same data rate of 11 Mb/s for their transmissions, 2) a homogeneous system in which both of the users choose 1 Mb/s, 3) a heterogeneous system in which n_1 chooses 11 Mb/s and n_2 chooses 1 Mb/s.

The throughput values seen for these three situations are shown in Figure 3.4. For the 11 Mb/s homogeneous case, it is shown that around $t = 200$, user n_2 reaches the communication range of the basestation and starts to observe successful transmissions. Consequently, the throughput of n_1 starts to decrease as there are more users in the system now, until around $t = 250$; when both of the users are very close to the basestation. During this period, the medium is shared by two transmissions both with very good characteristics with high SNR values. As n_2 starts to get farther away from the basestation, the throughput observed by this node[5] decreases and equivalently, throughput observed by n_1 increases. This scenario demonstrates a fair system in which both of the users get half of the maximum medium capacity. The second scenario is very similar to the first one and the resulting system in this case is also fair as the available bandwidth is equally shared among all of the users.

In the final heterogeneous scenario, on the other hand, we see rather peculiar throughput values. High data rate user's throughput decreases vastly to a much smaller amount than it would get in a fair environment. The low data rate user, on the other hand, achieves a throughput level more than its fair amount! In this heterogeneous system, not only the higher data rate user is punished hardly, but also the low data rate user is rewarded beyond its deserved amount.

This phenomenon can be explained through the packet transmission example illustrated in Figure 3.5. For the sake of simplicity, it is assumed that there are no collisions or packet losses and all the transmitted packets are received correctly by the destination. Indeed, these are reasonable assumptions for our example as there are only two users competing for the channel (thus, probability of collision is very low) and both of the users are extremely close to the basestation at around $t = 300$(thus, probability of error is very low). Moreover, it is assumed that the packet sizes for both of the nodes are identical with the transmission times of t and

[5]We use the terms *wireless node*, *wireless user* and *wireless device* interchangeably throughout this dissertation.

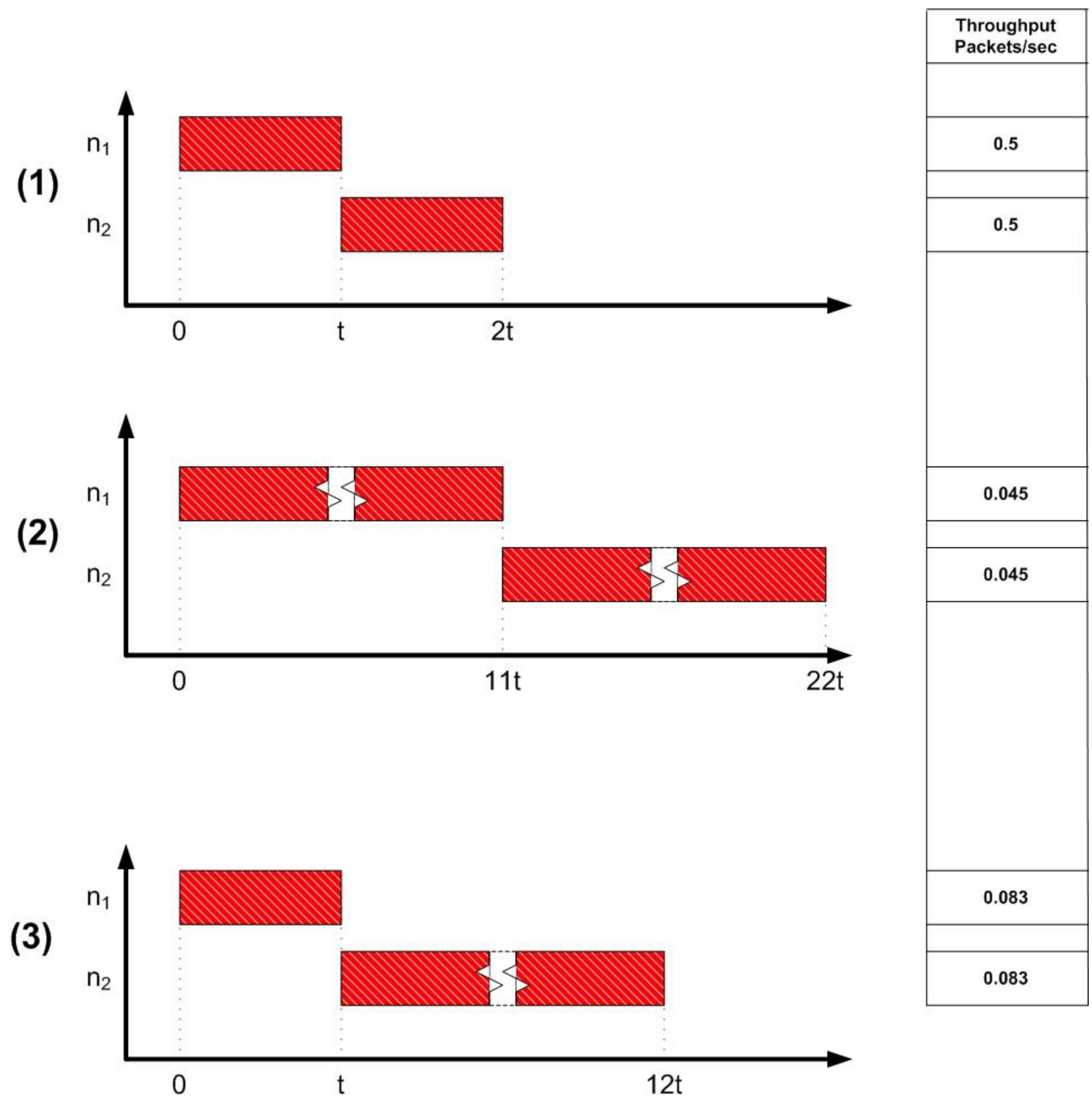


Figure 3.5: Packet access scenarios for the multi-rate anomaly example. In the multi-rate scenario, the fast user's throughput is severely degraded whilst the slow user observes throughput levels even more than its expectations.

$11t$ for the fast and the slow user respectively.

The long-term access schemas of the users can be approximated as one-by-one turns as shown in Figure 3.5[6]. In this two-user system, each user takes turns to access the medium and transmit their packets. In the top example, both users utilize high data rates and the transmissions end in $2t$ amount of time yielding identical throughput values of 0.5 $packets/t$. Similarly, in the middle scenario also, both of the users utilize the same low data rate, this time yielding identical 0.045 $packets/t$ throughput values alike. In the bottom scenario, however, the system is more heterogeneous and each user utilizes a different data rate. In this picture, low data rate user's occupation of the medium is 11 times more than the faster user's. The throughput values suddenly decrease greatly to 0.083 $packets/t$. This value is much smaller than what the fast user expects in a two-user system (top scenario), whereas it is much larger than what the slow user expects in a two-user system (middle scenario). Notice that n_2 might actually have very reasonable causes for utilizing the lower data rate choice (such as high environmental noise), but as it is trying to maximize its own throughput, it damages the other user's throughput immensely.

In the next section, we propose a metric in order to characterize the throughput performance in a multi-rate network.

---

[6]CSMA/CA algorithm is known to be short-term unfair [92]. However, as this unfairness will be observed with equal probability for all the users in the system in the long-run, it can be assumed that the system is long-term fair. This behavior is similar to the Ethernet capture effect [125].

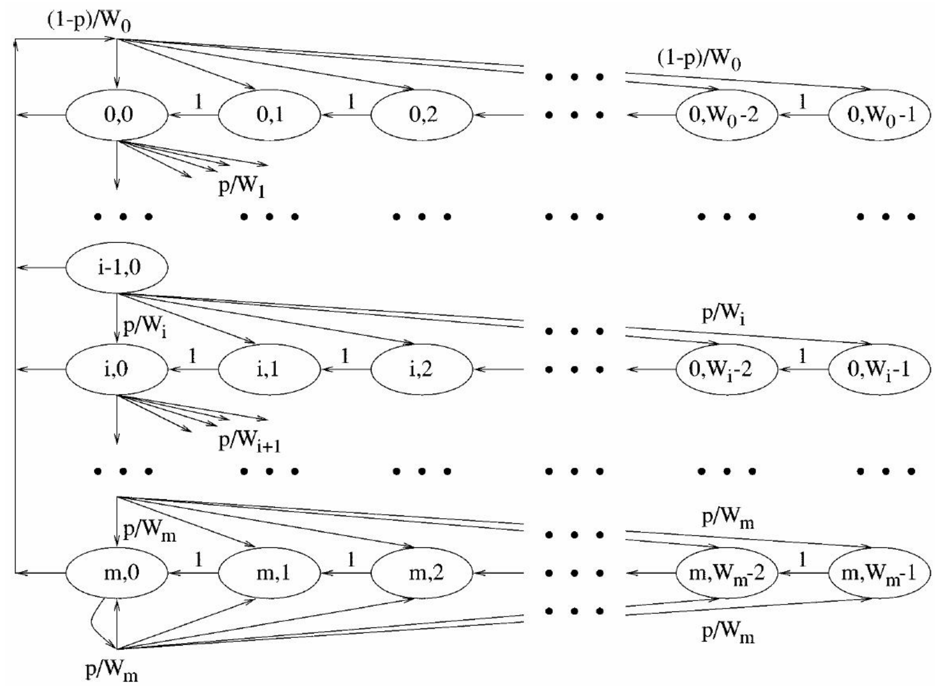


Figure 3.6: Markov chain model for the backoff window size in the Bianchi[26] analysis.

3.2 A Performance Metric for Multi-Rate Wireless Networks

3.2.1 Background

In [26], Bianchi provides an analytical method to compute the IEEE 802.11 DCF throughput under saturated conditions using Markov processes. In his work, the author assumes that the probability of a packet transmitted by each station to collide (denoted as p) is constant and independent. Moreover, the stationary probability that the station transmits a packet in a generic slot time is denoted as τ. Using these two parameters, the author formulates the CSMA/CA backoff process as a Markovian process as shown in Figure 3.6.

Analyzing this chain yields the following two important formulas.

$$p = 1 - (1 - \tau)^{n-1} \tag{3.1}$$

$$\tau = \frac{2(1 - 2p)}{(1 - 2p)(W + 1) + pW(1 - (2p)^m)} \tag{3.2}$$

where m is the maximum backoff stage and is equal to 5 for IEEE 802.11b, and n is the number of nodes in the system. Likewise, the saturated system throughput, G[7] , is denoted as the following:

$$G = \frac{P_s P_{tr} E[P]}{(1 - P_{tr})\sigma + P_{tr} P_s T_s + P_{tr}(1 - P_s) T_c} \tag{3.3}$$

where the parameters are defined in Table 3.1. Bianchi proposes to solve the two equations, Eq. 3.1 and Eq. 3.2, numerically and use the obtained parameter values for the throughput calculation using the Eq. 3.3.

In a similar work, Nadeem et al. [107] extends Bianchi's model further for accommodating the packet loss caused by imperfect channel conditions. In his work, Nadeem uses two different metrics in order to calculate the probability of unsuccessful transmissions: probability of collision and probability of error caused by ambient noise. He modifies the equations Eq. 3.1 and Eq. 3.2 as the following:

$$p_d = 1 - (1 - p_e)(1 - \tau)^{n-1} \tag{3.4}$$

$$\tau = \frac{2(1 - 2p_d)}{(1 - 2p_d)(W + 1) + p_d W(1 - (2p_d)^m)} \tag{3.5}$$

[7] *Saturation Throughput* is a performance metric defined as the limit reached by the system throughput while the offered load increases, and represents the maximum load that the system can carry in stable conditions [26]. We use the terms *saturation throughput* and *throughput* interchangeably throughout this dissertation.

Symbol	Description
P_s	Probability that a transmission is successful
P_{tr}	Probability that there is at least one transmission
$E[P]$	Average packet payload size
σ	Duration of an empty slot time
T_s	Time the channel is busy because of successful transmission
T_c	Time the channel is busy during a collision
W	Initial contention window size

Table 3.1: Parameter descriptions for the Bianchi model.

where p_e is the packet error rate and p_d is the cumulative error rate which considers both the collision probability and the packet error rate, i.e.

$$p_d = p_c + p_e - p_c p_e \tag{3.6}$$

where p_c is the probability of collision.

Also, the throughput is calculated as follows:

$$G = \frac{(1 - p_e)P_{tr}E[S]}{P_{id}\sigma + (1 - p_e)P_{tr}T_s + p_e P_{tr}T_f + P_{cl}T_c} \tag{3.7}$$

where P_{id}, P_{tr} and P_{cl} are the probabilities of a time slot 1) being idle where no node is transmitting 2) has transmission of only one node with probability of p_e of the packet getting corrupted 3) has a collision (cl) because two or more nodes are transmitting at the same time; respectively. [107] calculates these parameters in the following way:

$$
\begin{aligned}
P_{id} &= (1-\tau)^n \\
P_{tr} &= n\tau(1-\tau)^{n-1} \\
P_{cl} &= 1 - P_{id} - P_{tr}.
\end{aligned}
$$

3.2.2 Modification of the Bianchi Model for Different Data Rates

3.2.2.1 Homogenous Case

If homogenous users are assumed (i.e. all of the users in the system use the same data rate regardless of any environmental characteristics), modification of the Bianchi model is straightforward and can be extended directly from Nadeem's work [107].

If the network nodes (collaboratively) can choose different data rates, the throughput is affected by the decision of the group of individual nodes. Since the Markovian chain defined in Bianchi model does not change in such a case, the equations Eq. 3.4 and Eq. 3.5 still hold. However, when the data rate choice changes, transmission sensitivity to the channel characteristics (such as noise) changes as well. Hence, the error probability used in Eq. 3.7 changes with the data rate choice. Then, if all the nodes are using the data rate d, then the previous equations transform into

$$p = 1 - (1 - p_{e,d})(1-\tau)^{n-1} \tag{3.8}$$

$$G = \frac{(1-p_{e,d})P_{tr}E[S]}{P_{id}\sigma + (1-p_{e,d})P_{tr}T_{s,d} + p_{e,d}P_{tr}T_{f,d} + P_{cl}T_{c.d}} \tag{3.9}$$

where $p_{e,d}$ is the packet error rate seen using the data rate choice d.

Figure 3.7 illustrates the relationship between the packet error rate and the cumulative packet loss rate: when the error rate increases, the packet loss rate also

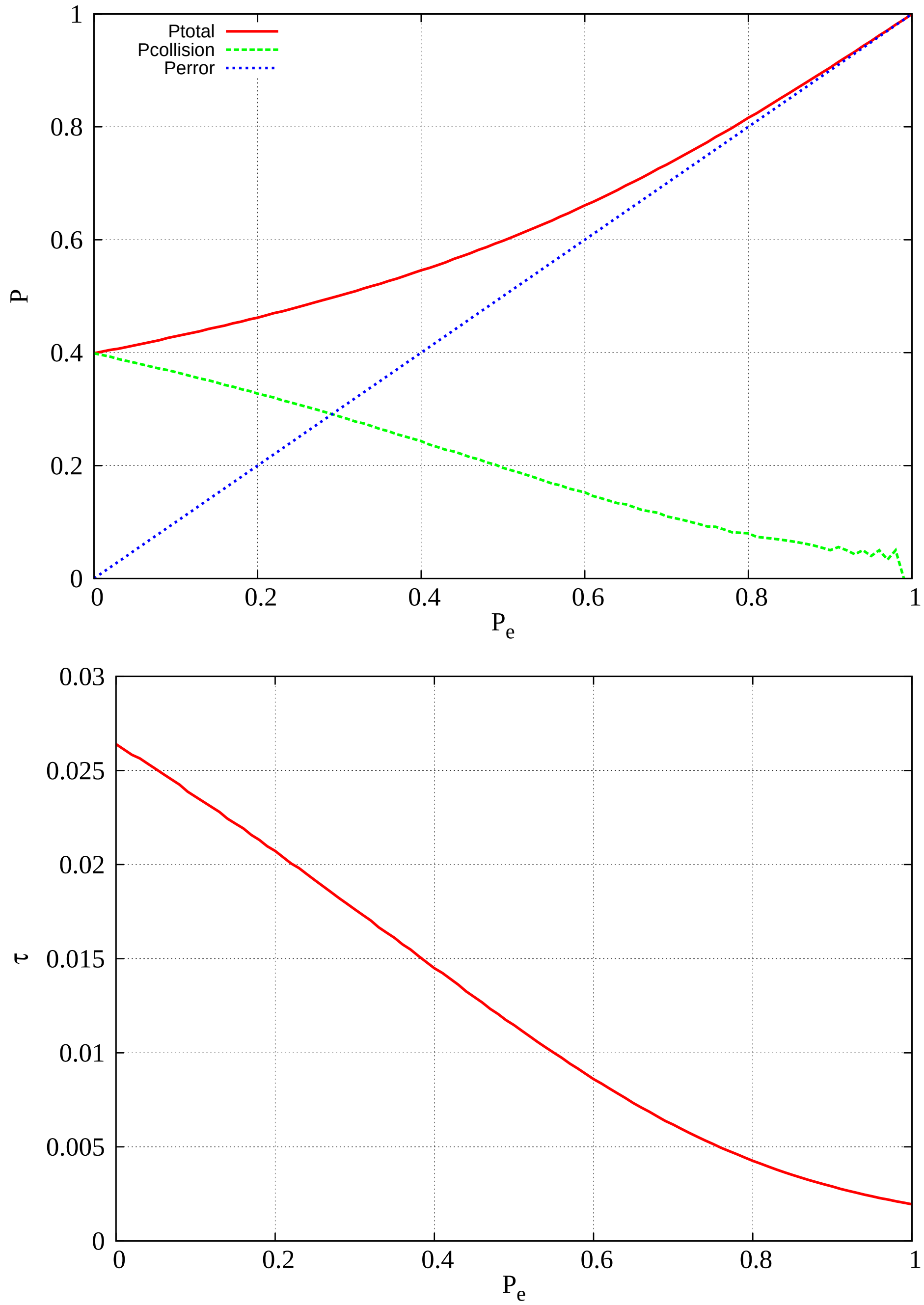


Figure 3.7: Packet loss sensitivity to packet error rate in a network with 20 users.

increases as expected; however, the collision probability decreases as the nodes start to wait for more amount of time in their backoff states. Hence, as the probability of error increases, the number of users staying in their *wait* states increases; consequently, this decreases the number of users contending for the channel. In the bottom part of this figure, we see that the probability of transmission decreases as the probability of error per packet increases, validating this observation.

3.2.2.2 Heterogenous Case

In a diverse wireless network, the data rate choices that the users prefer might differ according to the individual link characteristics, hardware capabilities and the link adaptation mechanisms they use. Formally speaking, in a heterogeneous environment, it is possible that some users choose data rate d_1 whereas some others choose d_2 for their ongoing communications. This results in an environment where there are several different sets of users, each with different probabilities of error and thus probabilities of loss.

Let's say that $D = d_1, d_2, \ldots, d_k$ represent the available data rates. Moreover, $N = n_1, n_2, \ldots, n_k$ represent the number of nodes that are using the particular data rates. For example, $d_1 = 1$Mb/s and $n_1 = 4$ mean that there are 4 users in the system at the moment which uses 1Mb/s data rate. Of course, $|N| = \sum_{i=1}^{k} n_i$. Then, for each set of data rate, the previous equations are extended as the following:

$$\forall i \in \mathbb{D} \quad \tau_i = \frac{2(1-2p_{t_i})}{(1-2p_{t_i})(W+1) + p_{t_i}W(1-(2p_{t_i})^m)} \tag{3.10}$$

$$\forall i \in \mathbb{D} \quad p_{t_i} = 1 - \frac{(1-p_{e_i})}{(1-\tau_i)} \times \prod_{d \in \mathbb{D}} (1-\tau_d)^{n_d} \tag{3.11}$$

Notice that the first equation has not changed much as it solely depends on the Markovian chain which does not require a modification. The latter equation, however, depends on the other users' transmission patterns and it has to be modified since the probabilities of transmission for different set of users are different in a heterogeneous environment. For a user with the data rate choice of d_i, the probability of successful transmission is observed when nobody in other data rate sets as well as nobody *else* in its own data rate set attempt to transmit.

As different data rates create different classes of users, each class' throughput value has to be calculated separately. The following equations represent the probability of the channel being idle, the probability that a user belonging to data rate d transmits; and the probability that a collision occurs on the medium respectively

$$P_{id} = \prod_{d \in \mathbb{D}} (1 - \tau_d)^{n_d} \tag{3.12}$$

$$\forall d \in \mathbb{D} P_{tr_d} = \frac{n_d \tau_d}{(1 - \tau_d)} \prod_{k \in \mathbb{D}} (1 - \tau_k)^{n_k} \tag{3.13}$$

$$P_{cl} = 1 - P_{id} - \sum_{d \in \mathbb{D}}^{d} P_{tr_d} \tag{3.14}$$

Using these equations, we can easily reconstruct the throughput calculation formula per each class of data users d as follows:

$$G_d = \frac{(1 - p_{e,d}) P_{tr,d} E[S]}{P_{id}\sigma + \sum_{d \in \mathbb{D}}^{d} (1 - p_{e,d}) P_{tr,d} T_{s,d} + \sum_{d \in \mathbb{D}}^{d} p_{e,d} P_{tr,d} T_{f,d} + E[T_c]}. \tag{3.15}$$

where $T_{s,i}$ and $T_{f,i}$ represent the total time spent in a successful and failed transmission attempt for a packet tranmsitted using data rate d_i, repsectively. Following the DCF message exchage shown in Section 2.1.2.2, these values can be calculated

Parameter	Description	Simulation Value
PHY	Size of physical header	24 bytes
MAC	Size of MAC header	28 bytes
ACK	Size of acknowledgement packet	38 bytes
S	Packet payload size	1480 bytes
d_{base}	Base data rate	1 Mb/s
$SIFS$	Short inter-frame space	10 μs
$DIFS$	DCF inter-frame space	50 μs

Table 3.2: Parameter descriptions and the simulation values used for the DCF message exchange.

as

$$T_{s,i} = (PHY + MAC)/d_{base} + S/d_i + SIFS + ACK/d_{base} + DIFS \tag{3.16}$$

$$T_{f,i} = (PHY + MAC)/d_{base} + S/d_i + DIFS \tag{3.17}$$

where the parameters are defined in Table 3.2.

$E[T_c]$ represents the expected amount of time spent in collisions and requires a more complex calculation than the homogeneous case. In the homogeneous case, the collision cost is equal to the duration of the transmission; whereas in the heterogeneous case, the collision cost varies according to the data rates of the packets involved in the collision. The following analysis investigates the expected cost of a collision.

Let $d_1 < d_2 < \cdots < d_{i-1} < d_i < d_{i+1} < \ldots < d_k$ represent the entire data rate choices available in the system. Then, the time costlier scenario is the one with at least a single user, which utilizes the data rate d_1, is enrolled in the collision. The next costlier case is the one in which at least one user with the data rate d_2 choice is involved in the collision, but none with d_1; and so on, so forth. The probability that the first case occurs can be calculated with the following calculation

$$P_c(1) = (1 - (1 - \tau_1)^{n_1} - n_1\tau_1(1 - \tau_1)^{n_1 - 1}\prod_{j=2}^{k}(1 - \tau_j)^{n_j}) \tag{3.18}$$

which excludes the cases of 1) no d_1 node transmits and 2) there is one and only one tranmission and it belongs to a d_1 user [8]. Hence, the resulting probability includes at least one unsuccessful (collided) d_1 transmission. The expected cost of collisions including at least one d_1 user is

$$T_c(1) = P_c(1) \times T_{f,1}. \tag{3.19}$$

The probability that the d_2 users can collide can be calculated in a similar way. However, notice that if a d_1 user is included in such a collision, since collisions of d_1 users are costlier, the medium will be occupied for the duration of the d_1 user. Hence, these cases should be excluded from this calculation. This can be achieved via creating a pseudo-world containing only the users d_2 through d_k. In this world, d_2 users would be the costliest users if involved in collisions and the cost associated with these collisions cost can be calculated as follows

$$T_c(2) = (1 - \tau_1)^{n_1} \times [1 - (1 - \tau_2)^{n_2} - n_2\tau_2(1 - \tau_2)^{n_2 - 1}\prod_{j=3}^{k}(1 - \tau_j)^{n_j}] \times T_{f,2}. \tag{3.20}$$

[8]Notice that this equation reduces to the homogenous case discussed before in Eq. 3.14 if there is only one data rate used by all the users in the system.

Generalizing, the expected amount of time elapsed in a collision can be represented as

$$
\begin{aligned}
E[T_c] &= \sum_{i=1}^{k} T_c(i) \\
&= \sum_{i=1}^{k} \Big[(\prod_{j=1}^{i-1} (1-\tau_j)^{n_j}) \times \\
&\quad [1-(1-\tau_i)^{n_i} - n_i \tau_i (1-\tau_i)^{n_i-1} \prod_{j=i+1}^{k} (1-\tau_j)^{n_j}] \\
&\quad \times T_{f,i} \Big]. \qquad (3.21)
\end{aligned}
$$

3.2.3 Modification of the Bianchi Model for Arbitrary Retry Limits

In this section, we analyze the Markovian chain presented in the previous chapters with finite retransmission limits. Figure 3.8 demonstrates the modified chain with the retry limit k. If $b_{i,k}$ represents the steady state probability of the Markovian state (i, j), then the entire process can be examined through the following analysis

$$b_{i-1,0}.p = b_{i,0} \rightarrow b_{i,0} = p^i.b_{0,0}$$

$$b_{0,0} = (1-p) \sum_{0}^{k-2} b_{i,0} + b_{k-1,0}$$

$$b_{i,h} = \frac{W_i - h}{W_i} . \begin{cases} (1-p) \sum_0^{k-2} b_{i,0} + b_{k-1,0} & \text{if } i = 0 \\ p.b_{i-1,0} & \text{if } 1 \leq i \leq k-1 \end{cases}$$

$$W_i = \begin{cases} 2^i \times W & \text{if } i \leq m \\ 2^m \times W & \text{if } m \leq i \leq k-1 \end{cases}$$

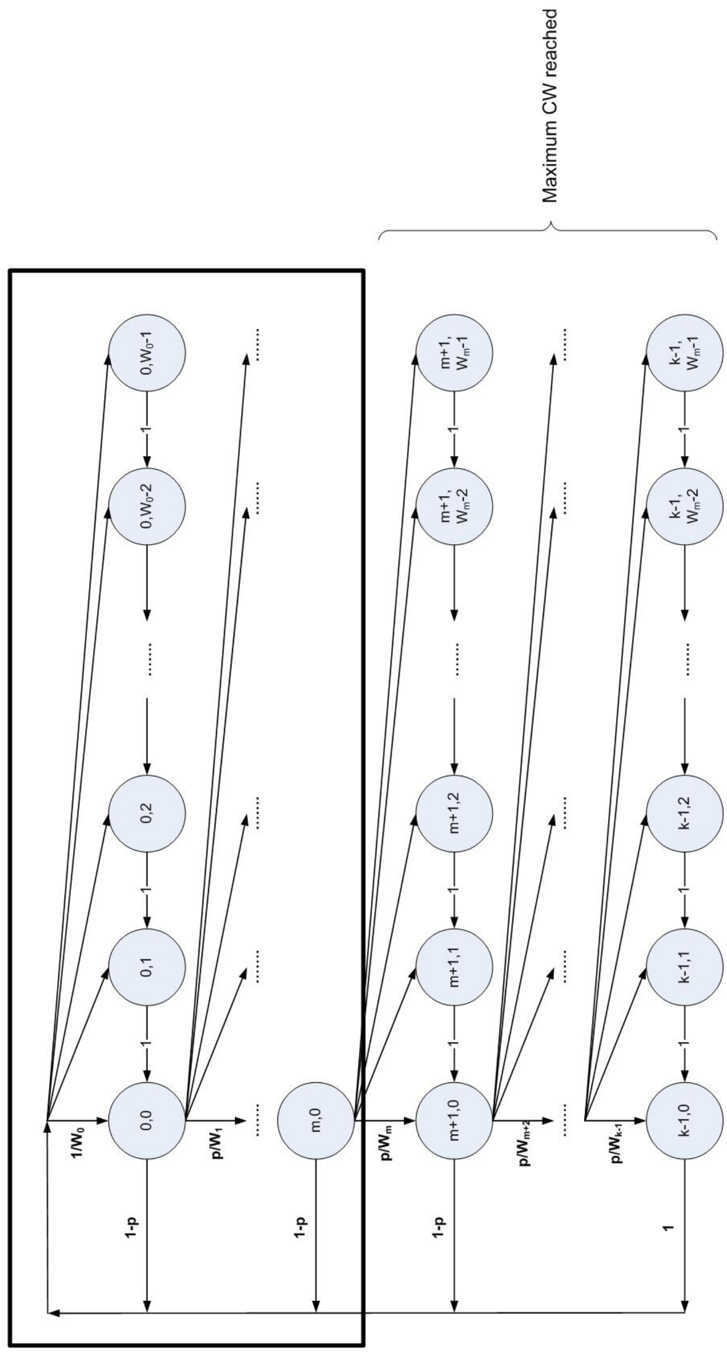


Figure 3.8: Modified Markov chain model for the backoff window size with finite number of retransmissions.

Of course,

$$b_{i,h} = \frac{W_i - h}{W_i}.b_{i,0}, i \in (0, k-1), h \in (0, W_i - 1).$$

Following the general Markovian process properties,

$$1 = \sum_{i=0}^{k-1} \sum_{h=0}^{W_i-1} b_{i,h} = \sum_{i=0}^{k-1} b_{i,0} \sum_{h=0}^{W_i-1} \frac{W_i - h}{W_i} = \sum_{i=0}^{k-1} b_{i,0} \frac{W_i + 1}{2} = \sum_{i=0}^{k-1} p^i \times b_{0,0} \times \frac{W_i + 1}{2}.$$

Then,

$$1 = \frac{b_{0,0}}{2} \Big[\sum_{i=0}^{k-1} (p^i \times W_i) + \sum_{i=0}^{k-1} p^i \Big]$$

$$\begin{aligned}
\textstyle\sum_{i=0}^{k-1}(p^i \times W_i) &= \textstyle\sum_{i=0}^{m}(p^i \times W_i) + \sum_{i=m+1}^{k-1}(p^i \times W_i) \\
&= \textstyle\sum_{i=0}^{m}((2p)^i \times W) + 2^m \times W \times \sum_{i=m+1}^{k-1} p^i \\
&= W \times \tfrac{1-(2p)^{m+1}}{1-2p} + 2^m \times W \times p^{m+1} \times \tfrac{1-p^{k-m-1}}{1-p}
\end{aligned}$$

$$1 = \frac{b_{0,0}}{2} \times \Big(W \times \underbrace{\Big[\frac{1-(2p)^{m+1}}{1-2p} + 2^m \times p^{m+1} \times \frac{1-p^{k-m-1}}{1-p} \Big]}_{\kappa(p)} + \underbrace{\frac{1-p^k}{1-p}}_{\zeta(p)} \Big)$$

$$\frac{2}{W \times \kappa(p) + \zeta(p)} = b_{0,0}. \tag{3.22}$$

In order to calculate $\tau = \sum_{i=0}^{k-1} b_{i,0}$, we use the fact that $b_{0,0} = (1-p)\sum_{i=0}^{k-2} b_{i,0} + b_{k-1,0}$, and deduce that

$$\tau = b_{0,0} \times \frac{1-p^k}{1-p} = b_{0,0} \times \zeta(p) \tag{3.23}$$

$$\tau = \frac{2\zeta(p)}{W \times \kappa(p) + \zeta(p)} \tag{3.24}$$

Notice that Eq. 3.24 assumes that $k-1 > m$. Otherwise, $\kappa(p)$ reduces to the following:

$$\kappa(p) = W \times \frac{1-(2p)^k}{1-2p}. \tag{3.25}$$

Generalizing for multiple data rate users,

$$\tau_i = \frac{2\zeta_i(p_i)}{W \times \kappa_i(p_i) + \zeta_i(p_i)} \tag{3.26}$$

where ζ_i and κ_i represent the functions calculated through utilizing the data rate d_i values.

Finally, Eq. 3.26 and Eq. 3.11 can be numerically solved and the results found can be used in the G calculation in Eq. 3.15 in order to estimate the expected throughput of a given multi-rate system. Notice that Eq. 3.26 enables different retry limit assingments per data rate users.

3.2.4 Model Validation

In this section, we investigate how well our model represents the real world wireless networks through comparison with simulative results. Original Bianchi model is known to estimate the system throughput well for changing number of wireless stations [26]. Thus, our simulation scenarios focus on two new aspects we have added to the original Bianchi model: 1) capability of handling multi-rate wireless networks and 2) capability of handling arbitrary retry limits assignments to different data rate groups.

Intuitively, in multi-rate networks, degree of heterogeneity can have considerable effect on the performance outcome. For instance, a network with 20 high data rate users and 1 low data rate user is expected to behave differently than a network with 20 low data rate users and 1 high data rate user. Hence, we have investigated the validity of our model in two different types of networks: *balanced* and *unbalanced*. In balanced networks, even though there are several different data rate users, the number of users in each data rate group is equal (for instance, 20 high and 20 low data rate users); whereas in unbalanced networks each group might have diverse number of users.

We have conducted all of the simulations we have provided in this chapter using the ns-2 tool [3] (version 2.29) with the extension provided by Fiore [54] for multi-rate transmission capability. All of the simulations were performed using UDP agents assigned per user with $11Mb/s$ traffic generation and 1480 bytes packet sizes. Our simulations assumed two-ray ground propagation model and we used the theoretical BER/SNR curves calculated by Pavon et.al. [112].

Each simulation scenario is conducted five times and the moving window average along the time-axis is calculated for each time point. The 95% confidence intervals are provided as well, as suggested by the previous works [76].

3.2.4.1 Balanced Multi-Rate Networks

Our first simulation creates a wireless network with two types of data rate choices, 11 Mb/s and 1 Mb/s, with 20 users utilizing each of these choices. In this 40 user network, all the users operate on the default retry limit of 7. Figure 3.9 illustrates the aggregated throughput values observed per different data rate groups. Notice that after the network reaches equilibrium around $t = 100$, both of the groups tend to observe almost identical throughput values close to our analytical calculation of 0.495 Mb/s.

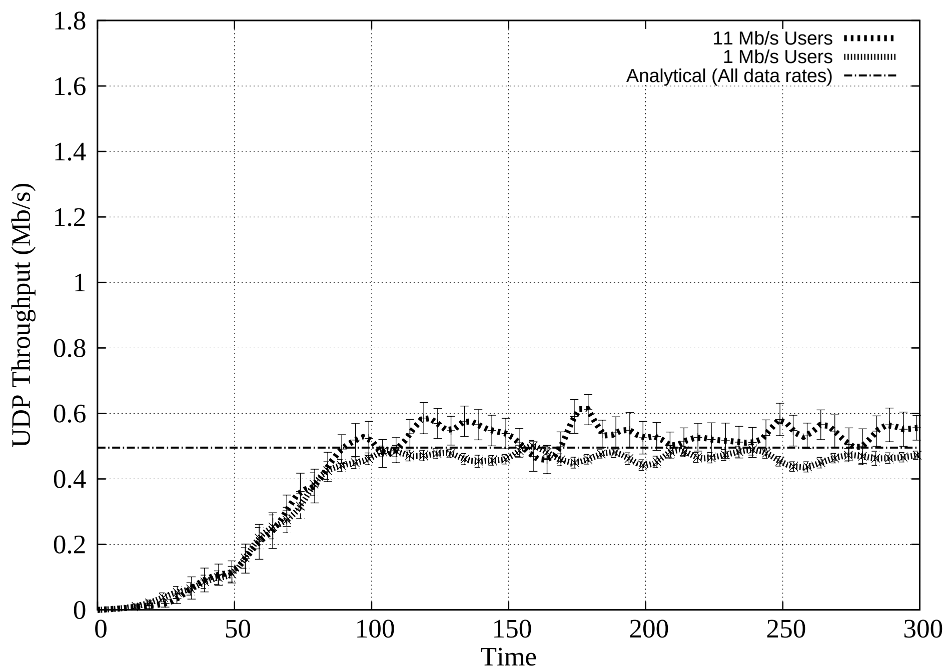


Figure 3.9: Aggregated throughput values observed by 11 Mb/s and 1 Mb/s users in a wireless network with 20 users utilizing each of these choices. Analytical results closely resemble the observed throughput values.

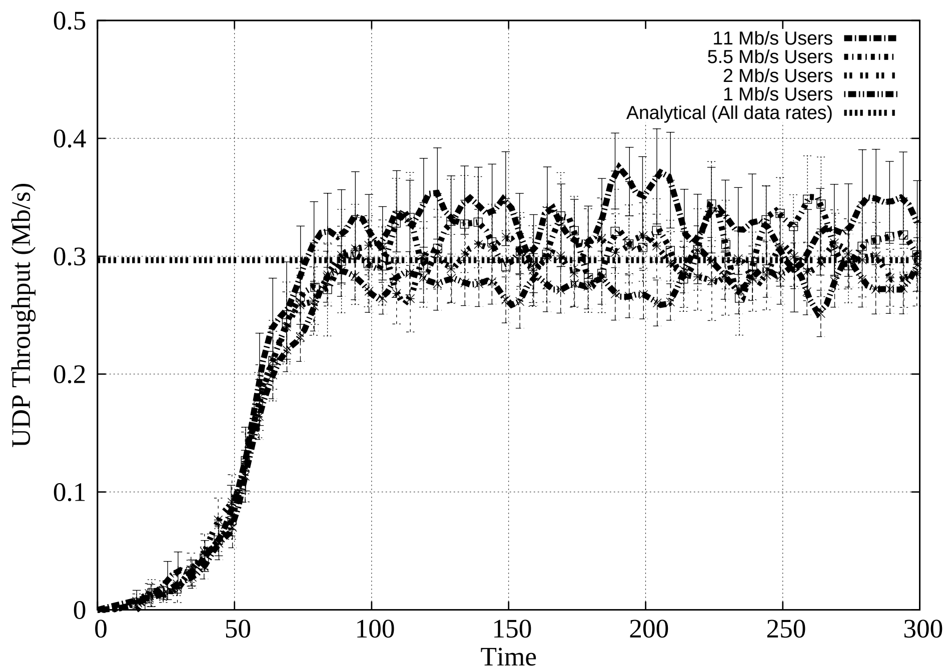


Figure 3.10: Aggregated throughput values observed by 11 Mb/s, 5.5 Mb/s, 2 Mb/s and 1 Mb/s users in a wireless network with 10 users utilizing each of these choices. Analytical results closely resemble the observed throughput values.

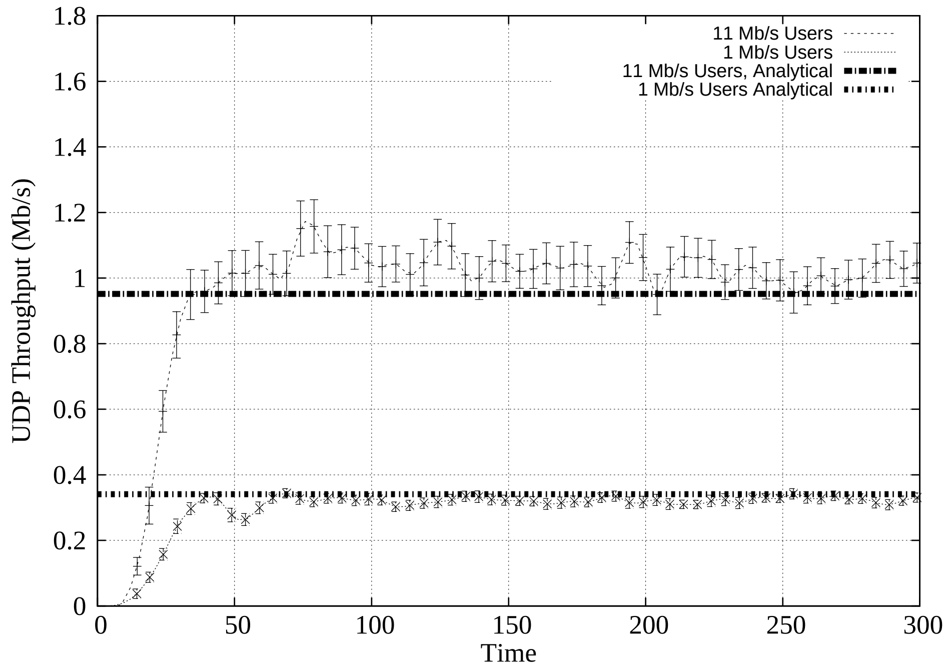


Figure 3.11: Aggregated throughput values observed by 11 Mb/s and 1 Mb/s users in a wireless network with 20 users utilizing each of these choices and with retry limits of 3 and 9 respectively. Analytical results closely resemble the observed throughput values.

Our second simulation considers a similar network of 40 users, this time with four data rate choices of 1 Mb/s, 2 Mb/s, 5.5 Mb/s and 11 Mb/s. Each user group contains 10 users with the default retry limit of 7. Figure 3.10 provides the aggregated throughput values for different data rates. Our analytical calculation of 0.2967 Mb/s approximates the simulative results closely.

Our third simulation focuses on arbitrary retry limit assignments to different data rate users and replicates the first simulation of two data rate choices of 11 Mb/s and 1 Mb/s with 20 users in each data rate group. As opposed to the first simulation,

Simulation Setup				Analytical Throughput (Mb/s) (Per Data Rate Group)	
11 Mb/s Users		1 Mb/s Users			
Size	Retry Limit	Size	Retry Limit	11 Mb/s	1 Mb/s
1 User	7	20 Users	7	0.0353	0.7085
1 User	3	20 Users	9	0.0564	0.7024
20 Users	7	1 User	7	3.5065	0.1745
20 Users	3	1 User	9	3.7484	0.0866

Table 3.3: Model Validation: Simulative results for the four unbalanced network scenarios.

we assigned the maximum retry limit of 9 to the 1 Mb/s users and 3 to the 11 Mb/s users. Figure 3.11 illustrates the aggregated throughput values observed for the two different data rate groups. Notice that, compared to the default retry limit of 7 in the first simulation, in Figure 3.9, two different data rate users observe very dissimilar values of aggregated throughput. High data rate users achieve almost 3 times the throughput values of the low data rate users. Our analytical model estimates the throughput values of 0.3409 and 0.9518 Mb/s for 1 Mb/s and 11 Mb/s users respectively. As can be seen from Figure 3.11, analytical results closely resemble the simulative results.

3.2.4.2 Unbalanced Multi-Rate Networks

In this section, we report on the validation of our model for networks with diverse sizes of different data rate groups. As in the previous sections, we have constructed networks with two different data rates: 11 Mb/s (high data rate) and 1 Mb/s (low data rate). Our four scenarios include two network setups: 1) 1 high data rate user and 20 low data rate users; 2) 20 high data rate users and 1 low data rate user. For both of these setups, we looked at different retry limit assignments: 1) the default case of assigning the retry limit 7 to all the users and 2) assigning 3 to the high data rate users and 9 to the low data rate users. These four scenarios are illustrated in Table 3.3.

The first two scenarios are illustrated in Figure 3.12 for the default (top) and arbitrary (bottom) retry limit assignment scenarios respectively. Similarly, third and fourth scenarios are shown in Figure 3.13. Both of these figures confirm our model's validity as the observed throughput values are visibly close to our analytical results.

3.3 Fairness

Fairness in multi-rate networks is defined through the occupancy times of the users on the medium [68] and referred as *time fairness* [140](or *temporal fairness*). According to the time-fairness principle, each user accesses the channel for equal amount of time, thus letting faster users transmit more than the slower.

In this section, we define a metric which estimates the fairness values of a

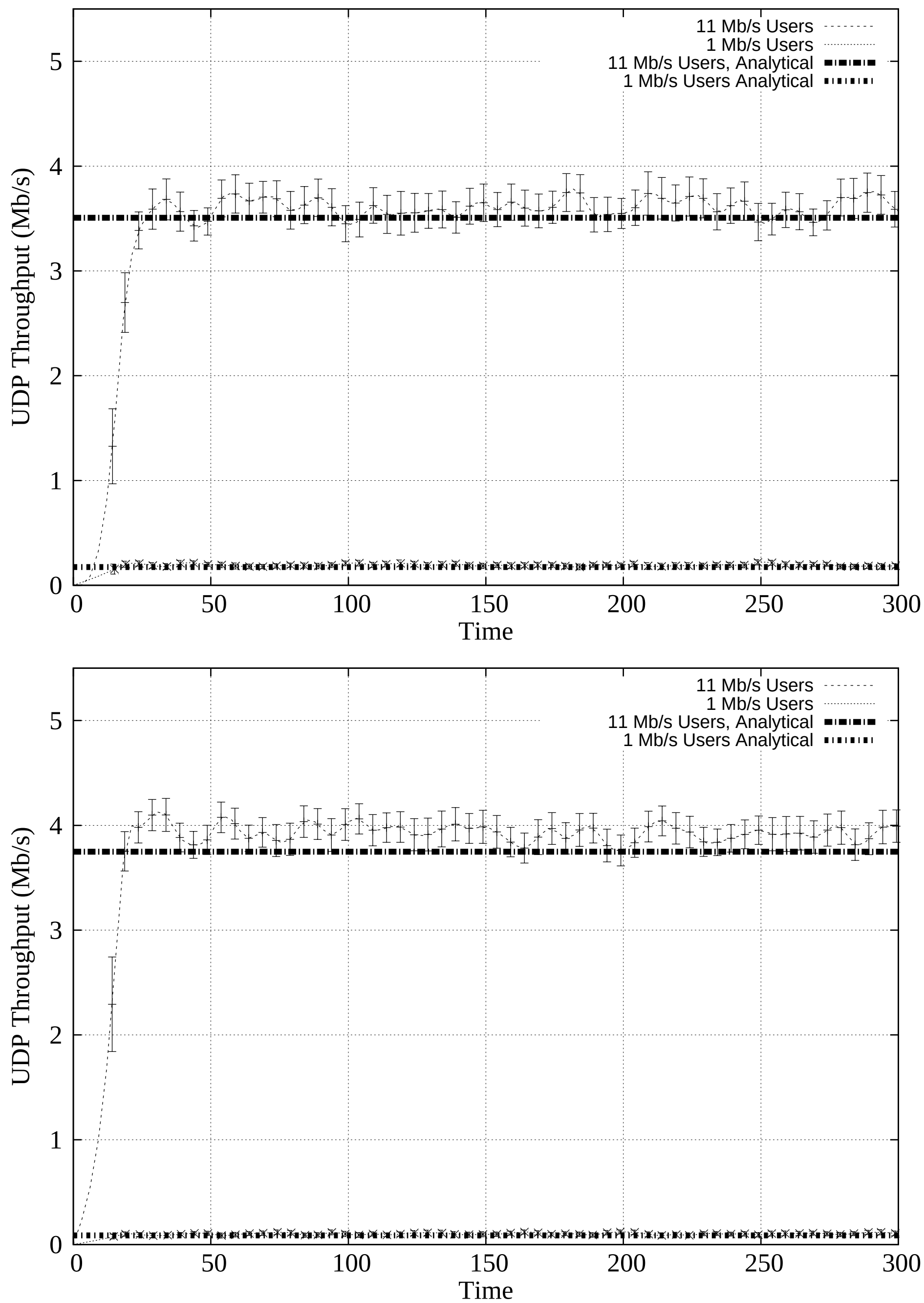


Figure 3.12: Aggregated throughput values observed by 11 Mb/s and 1 Mb/s users in a wireless network with 20 high and 1 low data rate users. In the top figure, all the users utilize the default retry limit of 7. In the bottom figure, high data rate users and low data rates utilize 3 and 9 as their retry limits respectively.

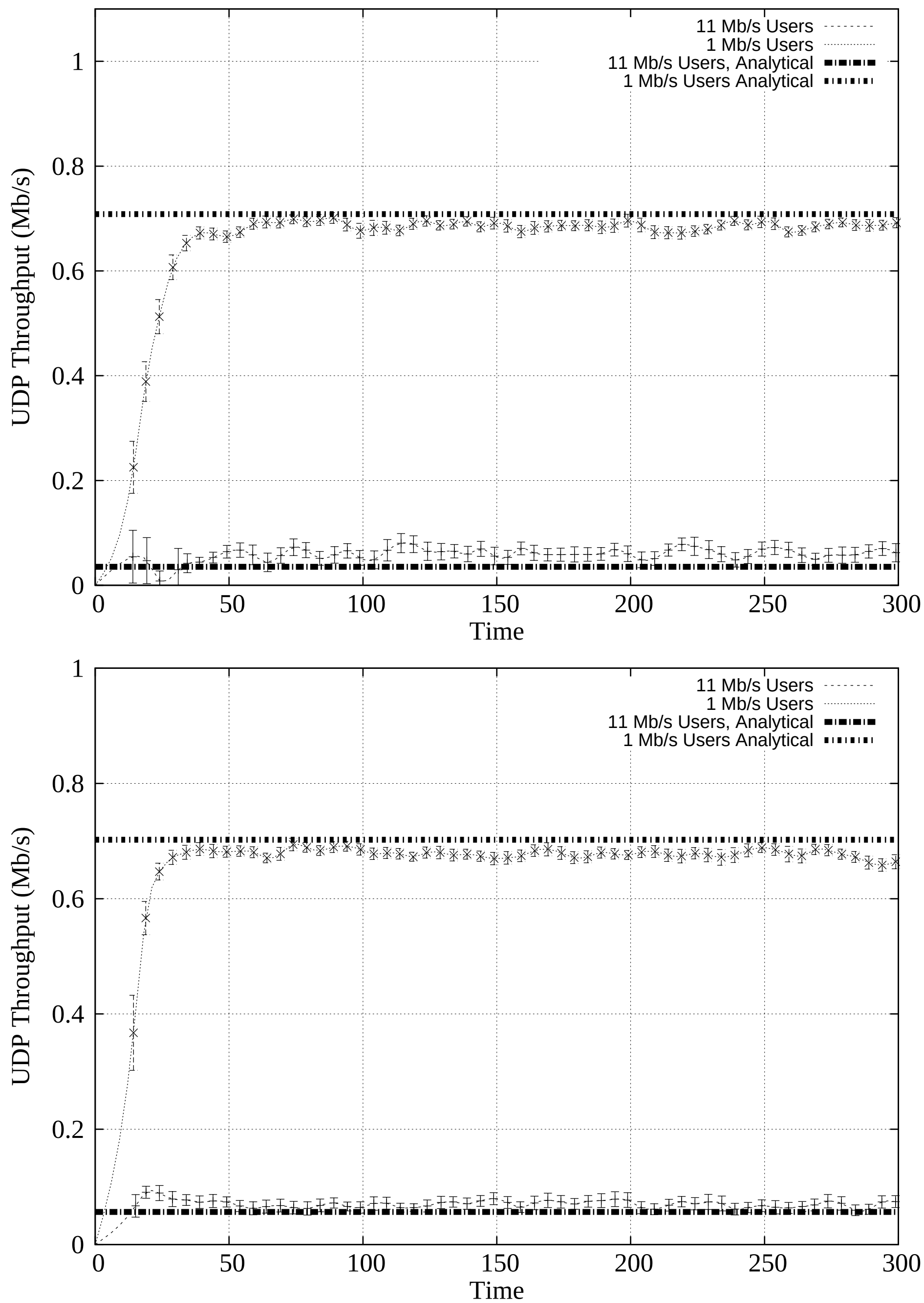


Figure 3.13: Aggregated throughput values observed by 11 Mb/s and 1 Mb/s users in a wireless network with 1 high and 20 low data rate users. In the top figure, all the users utilize the default retry limit of 7. In the bottom figure, high data rate users and low data rates utilize 3 and 9 as their retry limits respectively.

given wireless network. Our analysis follows the baseline property described in [140] as follows:

> **Baseline Property:** The long-term throughput of a node competing against any number of nodes running at different speeds is equal to the throughput that the node would achieve in an existing single-rate 802.11 WLAN in which all competing nodes were running at its rate.

Without loss of generality, let's assume that we have a system which consists of two kinds of data rate users: fast and slow. Moreover, let's assume that the number of users that belong to these classes are denoted as n_f and n_s respectively. According to the baseline property defined above, the high data rate users are going to expect throughput values as if all the other users in the system use the high data rate; and similarly, all the low data rate users are going to have expectations as if all the other users utilize low data rates.

In order to formalize the throughput values observed in different wireless systems, we define a new annotation, $G_d^{n_s,n_f}$ which describes the throughput seen in a system in which there exists n_s number of slow data rate users, n_f number of fast data rate users and the given value belongs to the d data rate users. For instance, $G_f^{5,10}$ denotes the throughput seen by the high data rate (i.e. fast) users in a wireless system with 5 low data rate users and 10 high data rate users.

For a system with n_s low data rate users and n_f high data rate users, the throughput amount seen by the former and latter can be shown as $G_s^{n_s,n_f}$ and $G_f^{n_s,n_f}$ respectively. Similarly, if the users all utilize the same data rates, the throughput values seen would be $G_s^{n_s+n_f,0}$ for the low data rate user and G_f^{0,n_s+n_f} for the high data rate user. Then, in a fair environment where each user gets the same percentage of the associated expected throughput level, the following ratio should hold

$$\frac{G_f^{n_s,n_f}}{G_s^{n_s,n_f}} \times \frac{\frac{1}{n_f}}{\frac{1}{n_s}} = \frac{G_f^{0,n_s+n_f}}{G_s^{n_s+n_f,0}} \times \frac{\frac{1}{n_s+n_f}}{\frac{1}{n_s+n_f}}. \tag{3.27}$$

Notice that the wireless systems using homogenous data rates (shown on the right side of the equation) both define a wireless network with $n_s + n_f$ users. Hence, if ideal situations are assumed, i.e. $p_e = 0$ for both cases, both of the systems will converge to the same τ and p values. Thus,

$$\begin{aligned}
\frac{G_f^{0,n_s+n_f}}{G_s^{n_s+n_f,0}} &= \frac{\frac{P_{tr}\times L}{P_{id}\times\sigma+P_{tr}\times T_{succ,f}+P_{cl}\times T_{fail,f}}}{\frac{P_{tr}\times L}{P_{id}\times\sigma+P_{tr}\times T_{succ,s}+P_{cl}\times T_{fail,s}}} \\
&= \frac{(1-\tau)^n\sigma + n\tau(1-\tau)^n T_{succ,s} + (1-(1-\tau)^n - n\tau(1-\tau)^{n-1})T_{fail,s}}{(1-\tau)^n\sigma + n\tau(1-\tau)^n T_{succ,f} + (1-(1-\tau)^n - n\tau(1-\tau)^{n-1})T_{fail,f}}
\end{aligned}$$

where $T_{succ,i}$ and $T_{fail,i}$ represent the total time spent in a successful and failed transmission attempts using data rate i.

As stated before, τ is the probability that a user transmits in a given slot. Given the nature of the CSMA/CA mechanism [9], $1 > \tau$. Thus, $1 > \tau > 0$ and

$$\lim_{n\to\infty}(1-\tau)^n = 0.$$

Then, for large number of users, previous equation reduces to

$$\frac{G_f^{0,n_s+n_f}}{G_s^{n_s+n_f,0}} = \frac{T_{fail,s}}{T_{fail,f}}. \tag{3.28}$$

[9]CSMA/CA is a probabilistic collision avoidance mechanism in which there is no state with a user transmitting with absolute certainty. Hence, it is not possible to have $\tau = 1$.

Formally, if all the users expect the associated baseline throughput values, the following property holds for each i, j user pair in the network:

$$F_i = F_j \tag{3.29}$$

where $F_i = G_i \times T_{fail,i}$. We denote this property as the *baseline fairness* property.

In order to compare the fairness acquired in different situations, we use the Jain's index described in [75] and define our *fairness index* as the following:

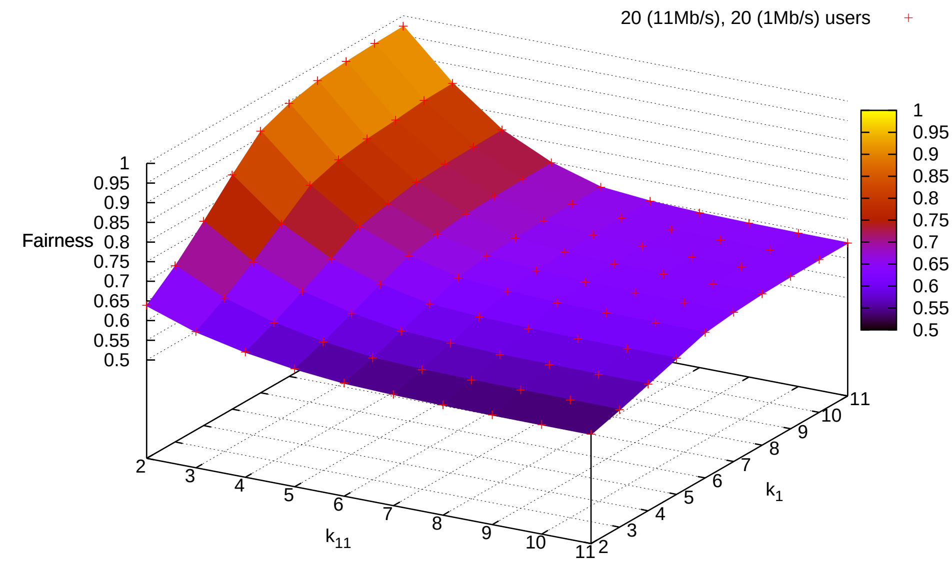
$$\frac{\left(\sum F_i\right)^2}{|F| \times \left(\sum F_i^2\right)}. \tag{3.30}$$

Fairness index above approaches to 1 for absolutely fair environements whereas it gets closer to 0 for unfair ones.

3.4 Analytical Results

Our hypothesis is that the fairness caused by the multi-rate anomaly can be restored through assigning different maximum retry limits to different data rate users in a multi-rate network. In order to validate this claim, we analytically examined our model developed in the previous sections. Intuitively, decreasing the retry limit of a user would shrink the number of backoff states available for this user to wait and reduce the expected amount of time the user *waits*; and consequently, increase the channel transmission probability for a given generic slot.

In our first analysis, we have investigated a wireless network which consists of 20 high data rate users (11 Mb/s) and 20 low data rate users (1 Mb/s). We have assigned arbitrary retry limits to these high and low data rate users, denoted as



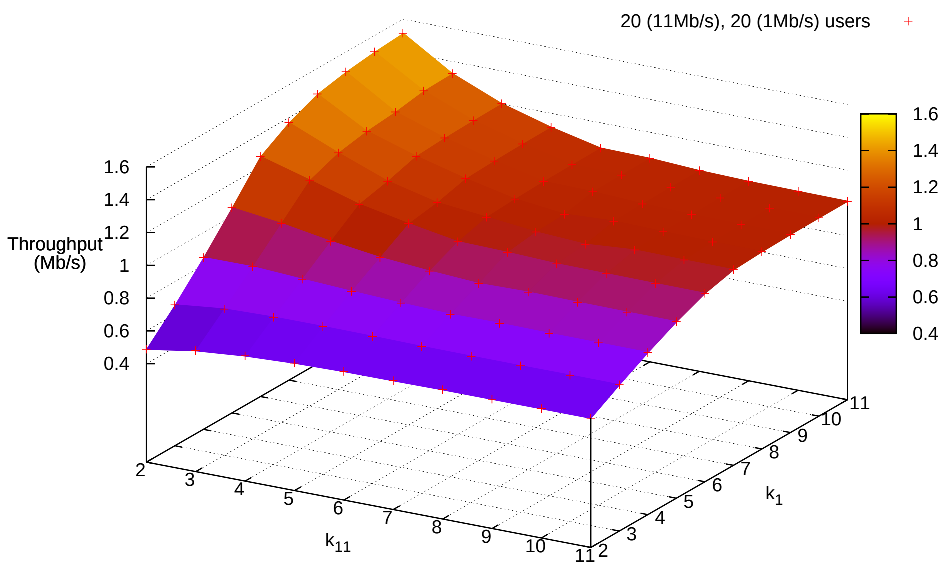


Figure 3.14: Fairness and total throughput values calculated analytically for a network with 20 11 Mb/s users and 20 1 Mb/s users, for changing the retry limit values of k_{11} and low k_1 respectively.

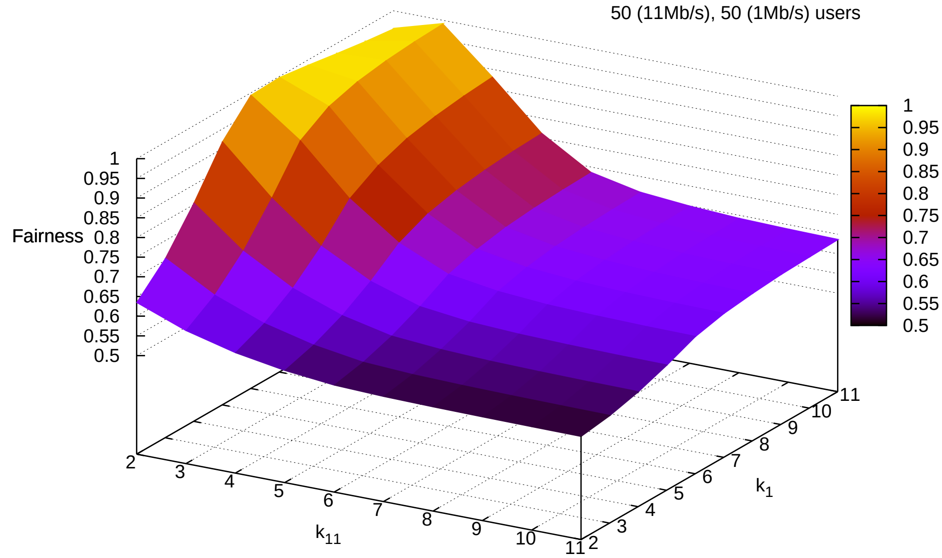
k_{11} and k_1 respectively. All of the other parameters are assigned as described in Table 3.2.

In Figure 3.14, top diagram illustrates the fairness observed in this network setup for different retry limit combinations. If the default retry limit of 7 is assigned to all of the users, the fairness value of the system is around 0.64. Notice that decreasing k_{11} improves the performance vastly. For instance, for $k_{11} = 2$ and $k_1 = 7$, the fairness jumps to 0.943. Further, increasing k_1 has also a similar effect, however in a more gentle way. For instance, fairness gain is merely around 0.01 when k_1 is increased to 11 and k_{11} is kept at the default value of 7. Similarly, increasing k_{11} or decreasing k_1 has a negative effect on the fairness.

Increasing fairness does not always guarantee increased throughput. For instance, a network in which all the users achieve 0.01% of their baseline throughput is theoretically fair since there is no discrimination among the users. However, the same network is less fair when half of the users achieve 100% of their baseline throughput whereas the other half achieve 75%. Clearly, the latter network would reach much better overall throughput values. Ultimately, a network in which all the transmissions are unsuccessful (thus achieved throughput is zero for the entire user set) is fair as well; however, such a network would be completely undesirable. Hence, in our analysis, we consider the achieved total throughput levels as well (as in Figure 3.14) in addition to the fairness values.

The only difference between the two diagrams in Figure 3.14 is at the lower left corner of the two images. Notice that for low k_1, decreasing k_{11} improves fairness, however worsens total throughput. Assigning lower retry limits has a potential drawback of decreased total throughput since the retry limits serve an important purpose: resolving potential future collisions. In networks with small number of users, collisions might not occur very frequently and assigning lower retry limits might not deteriorate the total throughput. However, in larger networks, collisions are unavoidable and the contention window used in the CSMA/CA mechanism has to be large enough in order to prevent collisions in future transmission trials as much as possible. Restricting the expansion of the contention window through low retry limit assignments might decrease the total throughput for larger networks as seen in Figure 3.14. Since there will be a lot of collisions in such a network, all the users are affected negatively regardless of their data rate choice; hence, theoretically, fairness improves in an undesired way.

In a similar effort, we looked at a larger network of 100 users with 50 users for each data rate choice. Figure 3.15 presents the fairness and total throughput values produced as in the previous example. Notice that Figure 3.15 resembles Figure 3.14 in many ways except one: decreasing the retry limits for higher data rate users (i.e. k_{11}) does not ensure continuous fairness and total throughput improvement. Both the top and the bottom diagrams show that both of the performance indicators reveal very high values for $k_{11} = 3$ and $k_1 \geq 8$ and further decreasing k_{11} forces the network performance to decrease. The reason behind this behavior, as in the previous example, is the inability of small contention windows to avoid collisions. If very low k_{11} values are assigned, the contention windows of high data rate users never grow large enough to resolve future collisions. Hence, high data rate users start to observe many collisions between each other and transmission failure rate



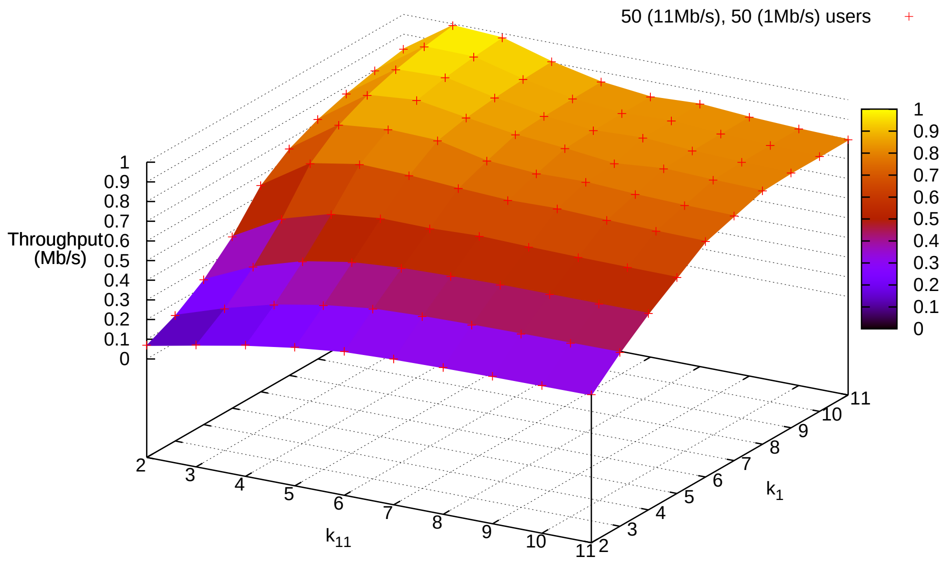
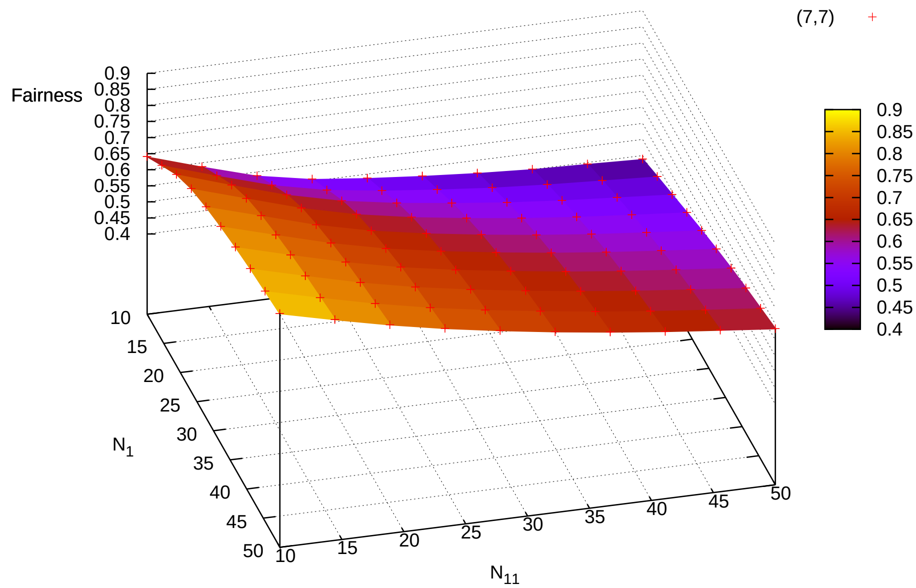


Figure 3.15: Fairness and total throughput values calculated analytically for a network with 50 11 Mb/s users and 50 1 Mb/s users, for changing the retry limit values of k_{11} and low k_1 respectively.

increases. Thus, lowering retry limit in an uncontrolled way might be detrimental to the overall performance.

In our previous two analyses, we have considered balanced networks in which the numbers of high and low data rate users are equal. In our next analysis, we have examined the effects of having data rate groups with different number of users. Figure 3.16 illustrates these effects in two different networks: one using the default retry limit of 7 for all of its users (top diagram), and one assigning 3 and 9 to its high and low data rate users respectively (bottom diagram). In these diagrams, N_1 and N_{11} denote the number of 1 Mb/s users and the number of 11 Mb/s users respectively. Notice that as a network comprising of only single data rate users would not contain any imbalance created by multi-rate anomaly, homogeneous networks are intrinsically fair. Hence, it can be conceived that the system is absolutely fair (i.e. $fairness = 1$) where $N_1 = 0$ or $N_{11} = 0$.

The top diagram in Figure 3.16 shows that fairness is affected in a much worse way when the number of high data rate users is larger than the number of low data rate users. Existence of low data rate users lowers the performance of high data rate users immensely. In such networks, majority of the users (i.e. high data rate users) observe unfair throughputs, thus low fairness values. However, as the number of low data rate users exceed the number of high data rate users, majority of the users become low data rate users which are observing relatively fair throughput values. Hence, fairness in these cases is higher than the former. Summarizing, we make the following observation: In a network with the majority of the users utilize low data rate choices, fairness is expected to be higher than a network with majority of the



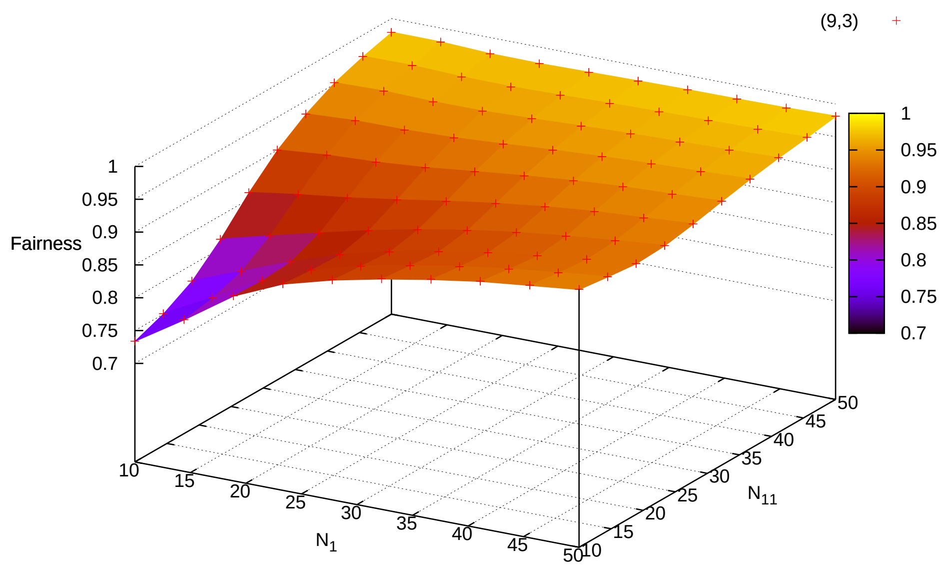


Figure 3.16: Fairness values observed for different number of users for the default retry limit assignment of 7 for all of the users (above); and 3 to high data rate users and 9 to low data rate users (below).

users utilize high data rate choices.

If medium contention in the network is at high levels, the users observe several consecutive collisions and timeout once they reach their retry limits. As the higher data rate users are assigned lower retry limits, they reach their limits earlier avoiding their contention window sizes growing larger. Thus, the higher data rate users have larger probability of accessing the medium at a given generic slot. However, this behavior requires large amount of contention for the medium, which entails large number of users, validating the two observations we have made in the previous paragraph. Notice that increasing the number of users in the network will increase the medium contention regardless of the new users' data rate choices, contributing to the fairness effectiveness in networks with arbitrary retry limit assignments (bottom graph in Figure 3.16).

We summarize the findings we observed so far as the following:

1. Existence of low data rate users in a wireless network affect the fairness of a multi-rate network in a much worse way than existence of high data rate users.

2. Lowering the retry limit for a specific user group in a wireless network causes higher probability of access values for the same user group, assuming saturated conditions.

3. Lowering the retry limit steadily does not assure unlimited station differentiation as assigning very low retry limits violates the purpose of the CSMA/CA mechanism.

4. Assigning low retry limits to more efficient users might increase the total

throughput given that the wireless network is not totally starved (i.e. too many users competing for the channel). If the wireless system is totally starved, lowering the retry limit might cause unnecessary packet drops resulting in lowered throughput.

5. If careful retry limit combinations are chosen in a wireless system, it is possible to acquire highly fair environments with heightened total throughput.

These findings suggest that a careful design that strategically assigns the retry limits of the users in a network might yield improved throughput and fairness. In the next sections, we describe such a system.

3.5 Multi-User Data Rate Adjustment Algorithm (MORAL)

In this section, we define a new algorithm, MORAL, that dynamically adjusts the retry limit of each user in a wireless network according to the channel characteristics observed and the neighbor information gathered from previous communications. MORAL targets a very ambitious problem: distributing the resources available to a group of users in an ethical way, through which each user is assigned as much resources as they can use without causing any suffering for the others [10].

[10] "I have observed that we all naturally desire happiness and not to suffer. I have suggested, furthermore, that these are rights, from which in my opinion we can infer that an ethical act is one which does not harm others' experience or expectation of happiness. ... The first thing to note is that [happiness] is a relative quality. We experience it differently according to our circumstances. What makes one person glad may be a source of suffering to another".

His Holiness the Dalai Lama

During a traditional IEEE 802.11 packet exchange, the nodes involved in the communication have a single way of assessing the channel quality: successful reception of acknowledgements for the packets previously transmitted. In the same way, MORAL utilizes the acknowledgement information as one of its heuristic principles in order to evaluate the channel conditions. In addition, MORAL monitors the traffic on the medium in order to collect information about the channel usages of its neighbors in its vicinity. MORAL combines these different types of information together in order to locally calculate the best retry limit to use for each node to achieve the fairest environment possible.

Because of the broadcast nature of the wireless medium, a packet transmitted might be *overheard* by the users other than the intended destination. In the scope of wireless networks in which energy efficiency is a big concern (such as the Wireless Sensor Networks [11]) overhearing is a major problem as it consumes the battery for an operation that is absolutely unnecessary. There has been several works proposed to mitigate the negative effects of overhearing in energy efficient networks, such as reducing the impact of overhearing during data gathering and dissemination [23] and turning off the nodes that are not involved in transmissions [136]. However, for the networks where energy efficiency is not the main criterion, overhearing can be used for a better use, such as in the case of *cooperative diversity* [95]. According to this principle, distributed users cooperatively get organized in order to jointly transmit information. Among these techniques which utilize the overheard information are

From *Ethics for the New Millenium* [45]

[11] For an excellent survey on Wireless Sensor Networks, please see [13].

resolving the port number conflicts for automatic spatial IP address assignment [51], cooperative relay selection [104] and detecting malicious packets modified in an unauthorized way [138]. In our work, we follow the same principle and utilize the information about the transmission characteristics of the fellow users extracted from overheard packets in order to assess the criteria for creating a fair environment.

In the next sections, we first describe a methodology for assigning retry limits through extending the equations found in the previous sections. Next, we portray the implementation details of our MORAL algorithm; and finally, we report on our empirical results that we gathered through extensive simulations.

3.5.1 Basics

The relative throughput values achieved by different data rate user groups under saturated conditions were analyzed in the previous sections using Equation 3.27; and it was shown that in a fair environment, the throughput values observed are inversely proportional to the duration of the failed transmission attempts by the users (Equation 3.28). In this section, we will focus on the left side of the Equation 3.27 in order to estimate the relative throughput observed by the users.

Let us consider a wireless network in which there are two types of data rate users: 11 Mb/s (fast) and 1 Mb/s (slow). In addition, n_s denotes the number of low data rate users and n_f denotes the number of high data rate users, as in the previous sections. Following Equation 3.27, the ratio of the throughputs observed of a single high data rate user to a low data rate user can be extended using the

Equation 3.15 as the following

$$
\begin{aligned}
\frac{G_f^{n_s,n_f}}{G_s^{n_s,n_f}} \times \frac{\frac{1}{n_f}}{\frac{1}{n_s}} &= \frac{(1-p_{e,f})P_{tr,f}E[S]}{(1-p_{e,s})P_{tr,s}E[S]} \times \frac{n_s}{n_f} && (3.31)\\
&= \frac{\frac{n_f\tau_f\left(\prod_{k\in\mathbb{D}}(1-\tau_k)^{n_k}\right)}{(1-\tau_f)}}{\frac{n_s\tau_s\left(\prod_{k\in\mathbb{D}}(1-\tau_k)^{n_k}\right)}{(1-\tau_s)}} \times \frac{n_s}{n_f} && (3.32)\\
&= \underbrace{\frac{\tau_f}{1-\tau_f}}_{\kappa'_f} \times \underbrace{\frac{1-\tau_s}{\tau_s}}_{1/\kappa'_s} && (3.33)\\
&= \frac{\kappa'_f}{\kappa'_s}. && (3.34)
\end{aligned}
$$

The first equation follows the fact that the expected amount of time spent on the medium is the same for both of the parties, i.e. the denominator in Equation 3.15 is the same for all the data rate users, hence cancelling out in this ratio.

The throughput values observed depends on the channel traffic as well as the channel conditions (thus the packet error rates (p_e) in Equation 3.31). If one of the users is experiencing bad channel characteristics causing severe packet losses, the resulting low throughput observed by this user is not due to the existence of lower data rate users in the system, but merely due to the packet losses observed at the destination. Hence the multi-rate anomaly is not the reason for the low throughput in such a case and the lower data rate users in the network should not be kept responsible for the packet losses observed. Thus, in our calculations of relative fairness, we *assume* that all the users observe best channel conditions, i.e. $p_e = 0$ for all of the users in order to eliminate this confusion. In the next two equations (Equation 3.32 and Equation 3.34), we define a new metric κ', which estimates the relative amount of time spent by the user in transmission with respect to staying idle. Indeed, when the ratio of two κ' values are taken, the numerator denotes the probability that the first user transmits when the second does not; and similarly the denominator denotes the probability that the second user transmits when the first does not.

Notice that κ' can be rewritten as the following, using the Equation 3.24,

$$\kappa' = \frac{2\zeta(p)}{W \times \kappa(p) - \zeta(p)}. \tag{3.35}$$

Combining Equation 3.34 and Equation 3.28 together, we can conclude that in a baseline fair environment, for each different data rate pairs (d_i,d_k), the following equation holds

$$\frac{\kappa'_i}{\kappa'_k} = \frac{T_{f,k}}{T_{f,i}}. \tag{3.36}$$

In a wireless system with k number of different data rates used, if the environment is baseline fair, then the following equation is expected to hold

$$\kappa'_1 \times T_{f,1} = \kappa'_2 \times T_{f,2} = \ldots = \kappa'_n \times T_{f,n} = \kappa^*. \tag{3.37}$$

Then, the best κ^* is the one that minimizes the following function

$$F_k = \sum_{i \in D} (\kappa^* - \kappa'_i \times T_{f,i})^2. \tag{3.38}$$

Taking the first order derivative for finding the maximal point leads to

$$\frac{\partial F_k}{\partial \kappa^*} = 2 \times \Big(\sum_{i \in D} \kappa^* - \sum_{i \in D} (\kappa'_i \times T_{f,i}) \Big) = 0. \tag{3.39}$$

Rearranging this formula yields

$$\kappa^* = \frac{1}{|D|} \times \sum_{i \in D} (\kappa'_i \times T_{f,i}). \tag{3.40}$$

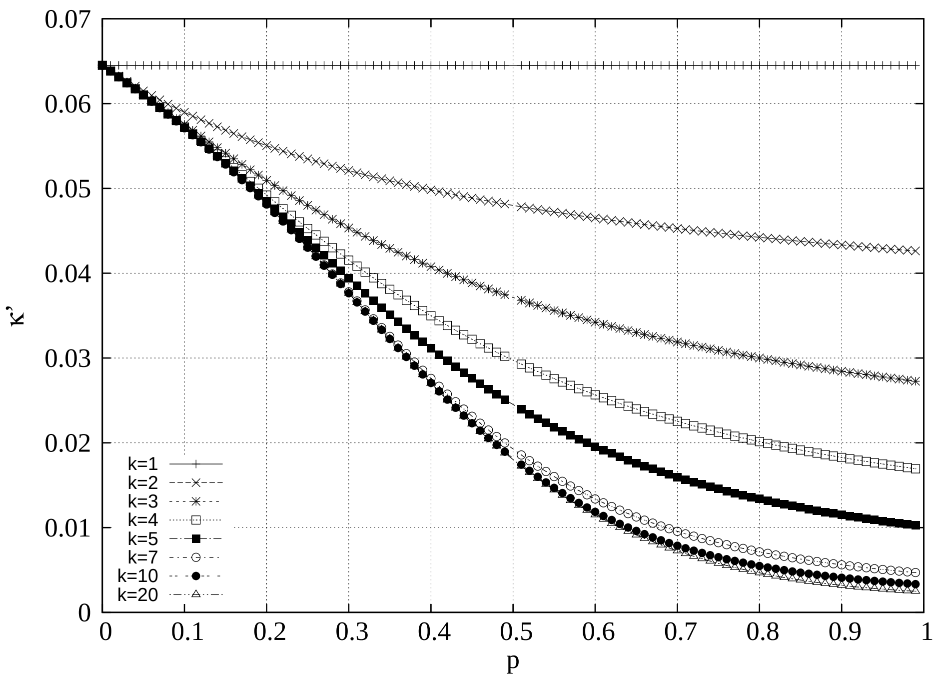


Figure 3.17: κ' function variation for different channel conditions and different retry limits. In networks with high packet loss probabilities, arbitrary retry limit assignments cause differentiation among the users.

Hypothetically speaking, if a wireless node has access to the κ' values of the neighbor nodes around itself, it can easily calculate the desired κ^* using Equation 3.40. As Equation 3.36 holds for each user pair, using the κ^* value calculated, the current node can estimate the κ' value that it has to maintain in this wireless network in order to establish the fairest environment. Consequently, the node can compare this value with its achieved κ' amount and can determine if it accesses the medium more than it should, or vice versa.

The behavior of the κ' function is shown in Figure 3.17 for changing packet loss probabilities. As the channel conditions gets worse, the expected probability of channel access versus staying idle ratio (i.e. κ' function) decreases monotonically. Assigning different retry limits causes variation in the κ' value as the number of backoff states are larger for higher retry limit assignments. This result promotes the validity of our hypothesis that assigning different retry limits to different groups of users (in our case different data rate groups) might create a differentiated environment in which some user groups have higher precedence of medium access than the others.

3.5.2 Algorithm Details

Equation 3.40 describes the optimal instantaneous κ^* value that the users would observe in the baseline fairest environment under the current conditions. If the users would have access to this value, they can easily calculate the κ' value they should utilize in this fair environment using Equation 3.37. Furthermore, as the behavior of the κ' function is already known (illustrated in Figure 3.17), each user can estimate the best retry limit to utilize under the current conditions and deploy this value for the future transmission attempts.

In a controlled network with a centralized entity managing the communication between the wireless users, the information required for the above calculations can be collected and the optimal retry limit can be assigned to all of the users by the central entity. However, such an assumption would narrow the usage of this mechanism only to controlled networks and would require major assumptions beyond the capabilities of the original IEEE 802.11 standard. Instead, MORAL is designed to function in a wide range of wireless network settings including uncontrolled, dynamic environments providing connectivity in an ad hoc manner.

MORAL recognizes the following items as its guidelines:

1. Estimate the retry limit to use through observing other users and their behavior in the current wireless system.

2. Be fair. Take the fair share without hurting other users in the system.

3. Do not violate the distributedness property of the system through any kind of centralized decision making.

4. Do not modify the current IEEE 802.11 schema, including the CSMA/CA mechanism (no packet modification, restructuring or protocol alteration).

MORAL uses the past information observed on the channel in order to estimate the retry limit to use for the future transmissions. Before going further with the algorithm details, we first define the notion of *transmission cycle* that is used as the time interval during which each user monitors the channel and collects neighbor information. A transmission cycle of a packet is the time spent between the operations of the particular packet being fetched from the upper layer buffers for transmission and its dismissal through a successful transmission or a failure. Hence, a transmission cycle can end with a successful transmission or a failure caused by reaching the retry limits. Notice that if the users are assigned low retry limits, then they will observe shorter transmission cycles. This is indeed a fine precaution as the users with very low retry limit assignments would immediately switch to higher retry limits if they cause channel starvation.

MORAL collects two kinds of information per transmission cycle: the number of transmissions observed per data rate[12] together with the identification of each user (e.g. MAC addresses) overheard. As an example, if a user overhears a transmission with data rate d_i from user n_i, then it increments its counter for data rate d_i by one, and notes the id, n_i, of the user. If the current user overhears the same user transmitting another packet, it again increments the data rate counter, however it does not store any additional information as this user's packets are already overheard and the user is registered. Notice that the storage amount is extremely small as it requires one counter to be stored per data rate heard and one id to be stored per user at the worst case with $O(|d| + |n|)$ storage complexity. These counters are all

[12] Data rate information can be calculated using the *duration* and the *length* fields in the IEEE 802.11 MAC and the encapsulated data packet headers respectively.

reset after each transmission cycle.

After each transmission cycle, each MORAL user estimates the wireless system performance and its relative expected medium usage in the current environment compared with the other fellow users. In order to perform this estimation, MORAL modifies Equation 3.31 for expected number of transmissions rather than the channel access probabilities. Notice that the probability of transmission parameters (i.e. P_{tr}) in these equations are defined per slot. Hence, for a time range which consists of m slots, the number of expected transmissions for data rate d_i users can be calculated as $m \times P_{tr,d_i}$. Thus, Equation 3.34 can be further extended through multiplication by m/m to reach the following

$$\frac{\kappa'_f}{\kappa'_s} = \frac{x_f}{n_f} \times \frac{n_s}{x_s}. \tag{3.41}$$

where x_i denotes the number of transmissions observed from data rate d_i users in m timeslots.

Using Equation 3.41, we can redefine Equation 3.37 as the following,

$$c_1 \times T_{f,1} = c_2 \times T_{f,2} = \ldots = c_k \times T_{f,k} = \kappa^*. \tag{3.42}$$

where $c_i = \frac{x_i}{n_i}$ represents the average number of transmissions per user in a group.

Rearranging this formula and taking the first order derivative as in Equation 3.39 yields

$$\kappa^* = \frac{1}{|D|} \times \sum_{i \in D} (c_i \times T_{f,i}). \tag{3.43}$$

As this equation holds for any arbitrary m number of slots, it is independent than the size of the monitoring interval and can be used for any given transmission cycle regardless of its length.

MORAL calculates the average number of transmissions per data rate (i.e. c_i) using the two types of information it collects through overhearing. For each data rate d_i users, c_i can be estimated as

$$c_i = \frac{[Number\ of\ packets\ overheard\ using\ d_i]}{[Number\ of\ unique\ users\ utilizing\ d_i]}. \tag{3.44}$$

Once the c_i values of all the data rate groups are calculated, these values can be used in order to calculate the κ^* value using Equation 3.43. Then, for the current user utilizing data rate $d_{current}$, the *calculated* fair c value that has to be used under these circumstances is $c_{calculated} = \kappa^*/T_{f,current}$.

During a transmission cycle, MORAL collects information from all of the neighbors including the users utilizing the same data rate; and the information gathered from the same data rate users is used in the abovementioned computations together with the others. Hence, conceptually, MORAL regards the current user as a group by itself, separating it from the other same data rate users. Thus, the number of users in this group is 1 and the $c_{calculated}$ computation does not require any knowledge about the number of users in the group.

We further define one additional parameter $c_{current}$ which describes the number of successful medium accesses immediately after each transmission cycle. The $c_{current}$ parameter can retain only two values, as the transmission cycles end either with a success (i.e. $c_{current} = 1$) or a failure (i.e. $c_{current} = 0$). Comparing this parameter with the fair amount, $c_{calculated}$; the user can estimate its closeness to the fair environment and take measures accordingly in order to reach a fairer environment if possible at the next transmission cycle.

$c_{calculated}$ can retain four values:

1. **Case I** ($\mathbf{c_{calculated} = 0}$) **:** No other transmissions were heard during the transmission cycle.

2. **Case II** ($\mathbf{0 < c_{calculated} < 1}$) **:** There are other users in the system which requires higher medium access precedence than the current user (i.e. current user is not the fastest data rate user).

3. **Case III** ($\mathbf{c_{calculated} = 1}$) **:** In the fairest system, current user is expected to transmit only once.

4. **Case IV** ($\mathbf{1 < c_{calculated}}$) **:** There are slower data rate users in the system than the current user and the current user is expected to transmit more.

In the following paragraphs, we elaborate on each of these cases and describe the necessary actions that have to be taken for each condition in order to establish fairer environments. Each action can be one of the following: retry limit increase ($\uparrow$), retry limit decrease ($\downarrow$), no retry limit change ($\leftrightarrow$) or *lean to default.* The last action closes the gap between the current retry limit assignment and the default one; and suggests that if the current retry limit is larger than the default retry limit, it should be decreased; and if it is smaller, it should be increased.

3.5.2.1 Case I

If $c_{calculated} = 0$, this essentially means that no other transmission was heard during the last transmission cycle. If $c_{current} = 1$, then the user could successfully transmit its packet, but no other users could. This situation can occur in two cases: 1) the environment is extremely competitive and even though the current user could successfully transmit, all of the other users' transmissions failed. Hence, the users should collaboratively increase their retry limits in order to overcome the competitiveness in the environment ($\uparrow$); 2) the current user is the only user in the environment and the retry limit choices are redundant as the user will not observe any collisions. In this situation, the retry limits can be assigned arbitrarily and increasing the retry limit does not affect the system. For consistency reasons with the case (1), we choose to increase the retry limit in this case as well ($\uparrow$).

On the other hand, if $c_{current} = 0$, then none of the users in the system could successfully transmit their packets, including the current user. This situation suggests a very competitive environment and the users should collaboratively increase their retry limits ($\uparrow$).

3.5.2.2 Case II

A user will observe $0 < c_{calculated} < 1$ only when there are users in the environment with data rate choices different than the current user (i.e. the environment is multi-rate). If $c_{current} = 1$, the current user had a successful transmission, which is more than its expected fair share. Hence, the user is unfair to the environment and should access the medium less through increasing its retry limit ($\uparrow$).

If $c_{current} = 0$, then the current user could not transmit successfully. Nevertheless, the expected transmission was less than one and there is not enough information about the condition and the retry limit of the current user should be kept constant ($\leftrightarrow$).

3.5.2.3 Case III

If $c_{calculated} = 1$, the system expects the current user to transmit exactly once during a transmission cycle. If $c_{current} = 1$, the current user transmitted its packet successfully as expected and the retry limit should be kept as it is if the environment is multi-rate ($\leftrightarrow$). However, if the environment is not multi-rate, then it is worth a try to probe the environment through leaning towards the default retry limit assignments (lean to default). If the probing results in a less fair environment, the user would automatically return to its previous retry limit choice.

If $c_{current} = 0$, the current user failed to transmit its packet successfully. However, the expected fair share of this user was to transmit exactly once in the transmission cycle and even though it failed to do so in the previous transmission cycle, it should have the same opportunity in the next one. Thus, similar to the previous paragraph, the retry limit of the current user is kept unchanged if the environment is multi-rate ($\leftrightarrow$); or modified through leaning towards default otherwise (lean to default).

3.5.2.4 Case IV

If $c_{calculated} > 1$, then the expected transmission for the current user was more than one. As this situation would suggest the current user to be more aggressive, it can cause the wireless system to suffer from contention if actions taken are decided upon carelessly. We analyze this situation for the multi-rate and the homogenous environments separately.

If the current user has overheard traffic utilizing data rate choices other than its own, the environment is multi-rate and the current user deserves a larger share of the medium than some of the other users in the network. If $c_{current} = 1$, the current user already could transmit its last packet successfully, encouraging a retry limit decrease ($\downarrow$). On the other hand, if $c_{current} = 0$, the last packet transmission attempt was unsuccessful which might have been because of an extremely competitive environment. Even though the fair share calculation suggests a more aggressive transmission schema for the current user, decreasing the retry limit under this situation might worsen the conditions; hence retry limits should be kept constant ($\leftrightarrow$).

If the current environment is homogenous, the parameter $c_{calculated}$ suggests that there are users in the system with the same data rate choice as the current user, but transmitting more aggressively. The reason behind this behavior might be a situation similar to the hidden node problem shown in Figure 2.2. For instance, if nodes A and B utilize high data rate choices whereas node C utilizes a lower data rate choice, node B would transmit more aggressively in order to mitigate the fairness caused by node C's transmission choices. Node A, on the other hand, does not observe node C's traffic and it does not perceive the unfairness caused by the multi-rate anomaly. Hence, in such a case, regardless of the fact that the last transmission was successful or not, node A's retry limit should not be decreased. Notice that this situation is also possible in cases where the low data rate users are

	Successful Transmission ($c_{current} = 1$)		Unsuccessful Transmission ($c_{current} = 0$)	
	Multi-rate	Homogenous	Multi-rate	Homogenous
$c_{calculated} = 0$	$\uparrow$	$\uparrow$	$\uparrow$	$\uparrow$
$0 < c_{calculated} < 1$	$\uparrow$	$\uparrow$	$\leftrightarrow$	$\leftrightarrow$
$c_{calculated} = 1$	$\leftrightarrow$	lean to default	$\leftrightarrow$	lean to default
$1 < c_{calculated}$	$\downarrow$	lean to default	$\leftrightarrow$	lean to default

Table 3.4: MORAL Heuristic Principles. Each action can be one of the following: retry limit increase ($\uparrow$), retry limit decrease ($\downarrow$), no retry limit changes ($\leftrightarrow$) or *lean to default* (please see the text for description).

Procedure 1 MORAL-Initiliazation

//Initialization of the data structures used
PROCEDURE init()
for d in D **do**
  clear usermap[d]
  $occurence[d] \leftarrow 0$
end for

moving away from the core of the network, which suggests reconsideration of the decisions made in the previous transmission cycles. Hence, the environment can be probed in order to validate current retry limit assignments (lean to default).

These principles are summarized in Table 3.4. Procedure 1 through Procedure 3 illustrate the MORAL algorithm implementation details.

Procedure 2 MORAL-Channel Monitoring

```
//Upon hearing a packet with data rate d from user u
PROCEDURE packet_received(user u, data_rate d)
usermap[d].add(u)
occurence[d] ← occurence[d] + 1
```

3.5.3 Experiments

In this section, we report on our findings about the performance of the MORAL algorithm proposed. All of our experiments were constructed using the ns2 simulation environment with version 2.29 on a PC with a 2.4 GHz dual-core CPU with 640 MB of RAM. In addition, we used the multi-rate extension package provided by Fiori, et.al. [54] with realistic channel modeling. In all of the simulations scenarios, each user was assigned a UDP agent with $11Mb/s$ traffic generation rate for uplink traffic. Our experiments assumed two-ray ground propagation model and we used the theoretical BER/SNR curves calculated by Pavon and Choi [112]. Apart from the modifications proposed for the MORAL algorithm, our simulation parameters adhere to the IEEE 802.11b standard. Unless otherwise stated, the architecture consists of a base station and three wired nodes as in Figure 3.3. The wireless nodes communicate with the corresponding wired nodes through the base station. For convenience, wireless nodes are associated with the wired nodes according to their data rate choices, i.e. all of the wireless nodes in the same data rate group communicate with the same wired node.

The analytical results presented in Figure 3.14 and Figure 3.15 suggest that assigning larger retry limits to low data rate users affect the fairness and total throughput of the network in a positive way; however, the amount of this positive

Procedure 3 MORAL-Update

```
//Execute each time CW is reset

//Current user k with data rate d_k, retry limit r_k
//r_min and r_max are minimum and maximum allowable retry limits
//r_d is the default retry limit
//timeout is false if the last transmission was unsuccessful, true otherwise

PROCEDURE calculate_k(boolean timeout)
K ← 0
nonzero ← 0
multirate ← false
for d in D do
   if (occurence[d] > 0) then
      K ← K + occurence[d]/usermap[d].size() * T_{f,d}
      nonzero ← nonzero + 1
   end if
   if (r_k != d) then
      multirate ← true
   end if
end for
K ← K/nonzero
if (K/T_{f,d_k} = 0) then
   r_k ← min(r_k + 1, r_max)
else
   if (K/T_{f,d_k} < 1) and (timeout = false) then
      r_k ← min(r_k + 1, r_max)
   end if
else
   if (1 ≤ K/T_{f,d_k}) and (multirate = false) then
      r_k ← r_k − |r_k − r_d| / (r_k − r_d)
   end if
else
   if (1 < K/T_{f,d_k}) and (timeout = false) then
      r_k ← max(r_k − 1, r_min)
   end if
end if
call init()
```

effect diminishes for larger values. For both of these scenarios, the improvement of the network performance starts to diminish around $k_1 > 10$. Indeed, Figure 3.17 illustrates that the κ' function, which has the same behavior as the throughput function, produces very similar outcomes for retry limits larger than 10. Hence, in our experiments, we fixed the minimum and maximum allowable retry limits as 1 and 10 respectively.

Our experiment scenarios consist of seven cases: In the first scenario, we simulate a static, non-moving wireless network with two groups of users, each deploying different data rate choices for their communications and containing the same number of users. In the second and third scenarios, we repeat the first experiment, only this time with four and three data rate choices respectively, adhering to the other data rate choices provided by IEEE 802.11b standard. The fourth and fifth scenarios focus on unbalanced networks in which different data rate groups contain different number of users. In the sixth and seventh scenarios, we simulate a dynamic network in which there are two groups of users and one of the groups (low data rate users), starting from a far away location, move into the group of high data rate users, pass through the system and leave the communication range of the base station through the end of the simulation. We analyze the *arrival* and *leave* cases separately in the sixth and seventh scenario sections, respectively.

All the simulation scenarios described in this section were conducted five times and the *Student-t Distribution* [139] is used in order to calculate the confidence intervals [76].

3.5.3.1 Scenario 1: Balanced Static 20-20 Case With Two Rates

Our first experiment is an extension of the scenario presented in Section 3.2.4 and Figure 3.9; in which there are 20 11 Mb/s users and 20 1Mb/s users. The distance between the users and the base station is negligible and the users start their transmission shortly after the initialization, around $t = 10$. In this experiment, the users do not move and stay in their initial position throughout the entire experiment. All of the users in the system implement the MORAL algorithm and use the information they gather from other users through overhearing.

In the top graph of Figure 3.18, the cumulative throughput values of the two groups are illustrated. The average throughput values shown in the figure were calculated using the outcomes of all the experimental runs, and the error-bars in the graph represent the 95% confidence interval. Notice that as soon as the simulation starts, the throughput values naturally converge to the average values and tend to stay stable during the rest of the simulation. Using MORAL the high data rate users which are capable of transmitting information in much higher speeds, observe much higher throughput values than the lower data rate users; creating positive differentiation between different data rate groups.

In a homogenous environment with 40 high data rate users, the expected cumulative throughput of 20 users is around 2.4 Mb/s [13]. Similarly, in a network with 40 low data rate users, the expected cumulative throughput of 20 users is around 0.3 Mb/s [13]. It is shown previously in Figure 3.9 that because of the multi-

[13] Value calculated analytically and confirmed empirically.

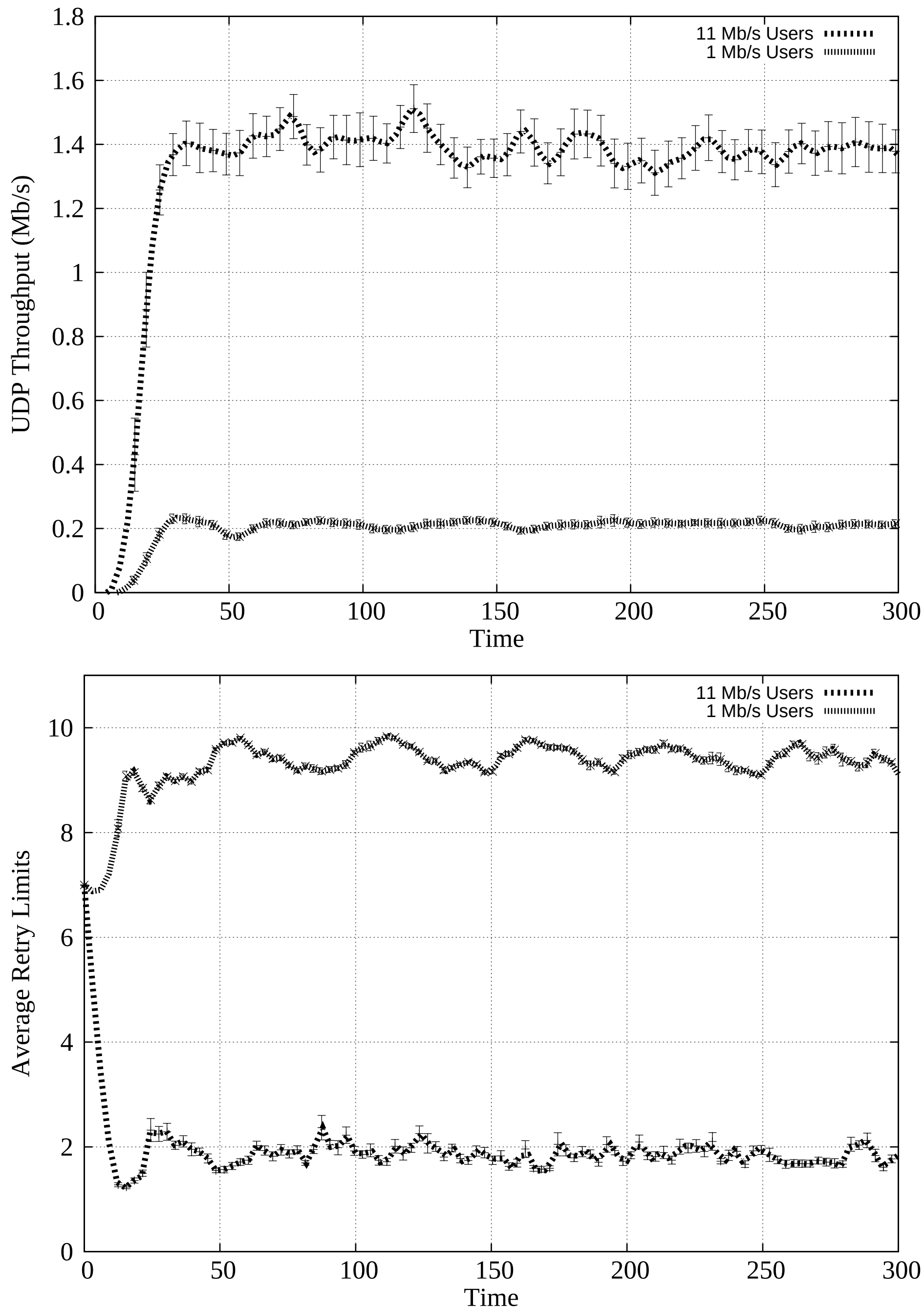


Figure 3.18: Throughput values observed and the average retry limits assigned by MORAL in a network with 20 11 Mb/s users and 20 1 Mb/s users. Fast users are assigned much smaller retry limits than the slow users, creating a throughput differentiated network.

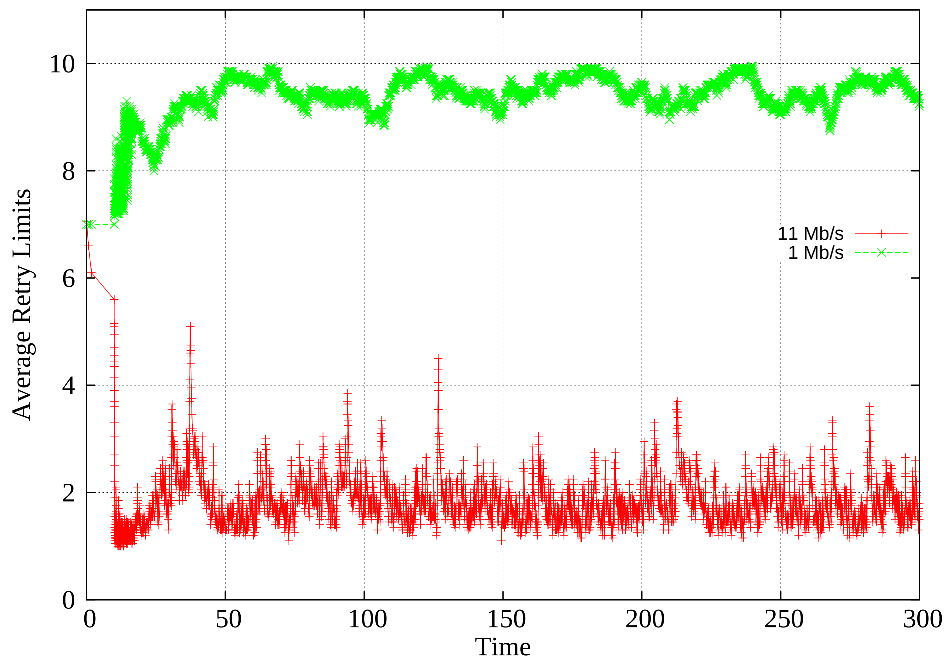


Figure 3.19: Average retry limits assigned by MORAL during a single simulation in a network with 20 11 Mb/s users and 20 1 Mb/s users.

rate anomaly effect, the throughput values observed are much more different than these expected values: around 0.5 Mb/s for each of the groups. This value is 1/5 of the fast users' expectation, whereas it is almost two times the slow users'. MORAL, on the other hand, regulates the throughput share of different data rate groups in a fairer manner. In the MORAL-enabled environment illustrated in Figure 3.18, achieved throughput values are around 1.4 Mb/s for the fast users and around 0.2 Mb/s for the slow users. These values are close to 2/3 of the expected throughput values for each of the user groups.

The bottom graph of Figure 3.18 demonstrates the average retry limits of each data rate groups. Adhering to the standard principles, all of the users start their run by choosing the default value of 7 as their retry limits. As the users start to monitor the environment, they become aware of other data rate users in the system with similar resource allocations (i.e. all the users initially access the medium with the same probability). Each user adjusts its retry limit according to the data rate it uses and the information it gathers from other users, following the MORAL fairness criteria. Fast users start decreasing their retry limits immediately, as they realize that they have to transmit more packets in order to create a baseline-fair environment. Similarly, slow users behave the opposite, realizing that they have to transmit less.

Since CSMA/CA algorithm is probabilistic, it is possible to observe some transmission cycles in which all the overheard packets belong to a single data rate group. After this type of transmission cycles, the users tend to re-adapt their retry limit parameter as they conceive the medium as homogenous and not multi-rate.

For instance, if only high data rate traffic is successfully transmitted during a cycle, the high data rate users predict the environment as homogenous, and they tend to lean towards default, increasing their retry limits. Similarly, if only low data rate traffic is overheard, the low data rate users tend to lean towards default. This phenomena lets the users probe the environment and re-evaluate their previous retry limit decisions. Figure 3.19 illustrates the average retry limits of different data rate users during a single run and shows several sparks identifiable as such *lean to default* instances. Note that the fast users probe the medium more frequently as they deploy smaller retry limits and thus shorter transmission cycles. This acts as a precautionary mechanism preventing the faster users from monopolizing the medium.

Figure 3.20 top graph illustrates the total throughput observed in the system compared with the default, MORAL-disabled scenario. The average total throughput is improved by around 60.0%, from 1.0 Mb/s to 1.6 Mb/s. The bottom graph in the same figure shows the fairness values observed in the MORAL and the default scenario together. The MORAL-enabled scenario produces a much fairer environment with fairness value very close to 1.0 compared with the default scenario with fairness value of 0.62. In this scenario, MORAL algorithm creates an environment with 61.3% increased fairness. Notice that the throughput values seen in these graphs coincide closely with the analytical results found in Figure 3.14 for the average retry limits observed in Figure 3.18 (i.e. 2 for fast users and 9 for slow users).

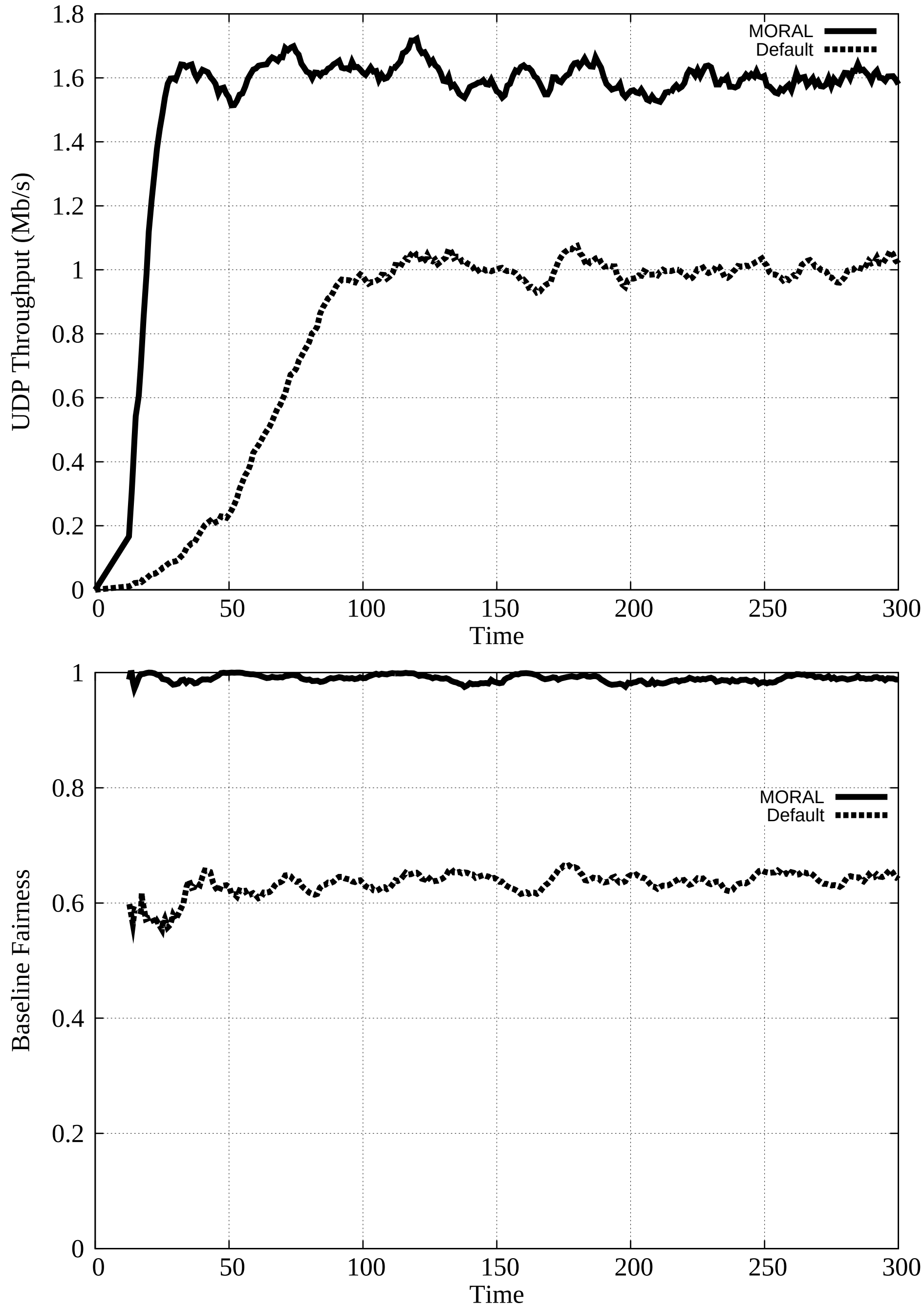


Figure 3.20: Average total throughput and fairness values observed in a network with 20 11 Mb/s users and 20 1 Mb/s users for MORAL and default scenarios. MORAL improves the throughput by 60% and the fairness by 61.3%.

3.5.3.2 Scenario 2: Balanced Static 10-10-10-10 Case With Four Rates

In our next experiment, we created a heterogeneous data rate environment with all four data rates supported by the IEEE 802.11b standard, namely 11, 5.5, 2 and 1 Mb/s. This network consists of 40 wireless users, 10 for each of the data rate choices. Similar to the previous experiment, the distance between the wireless nodes and the base station is negligible and the users start their transmission shortly after the initialization, around $t = 10$.

In Figure 3.21, we show the total throughput values observed by each of the groups in default scenario (top) and the MORAL-enabled scenario (bottom). Notice that in the top graph, all the user groups observe similar throughput values, close to 0.3 Mb/s. This throughput value is almost two times the amount that the 1 Mb/s users expect in a homogenous environment as illustrated previously and it is much lower than the 11 Mb/s users' throughput expectation of 1.2 Mb/s.

In the bottom graph in Figure 3.21, the throughput values observed by all the groups in a MORAL-enabled scenario is shown. Notice that, contrary to the default scenario, MORAL enables users to observe throughput values in accordance with the transmission speeds they are capable of. For instance, 11 Mb/s users and 1 Mb/s users observe throughput values of 0.9 Mb/s and 0.1 Mb/s respectively. These results promise a much fairer environment (with 75% of 11 Mb/s users' expectation and 67% of 1 Mb/s users' expectation) than the default scenario (with 25% of 11 Mb/s users' expectation and 200% of 1 Mb/s users' expectation). Consequently,

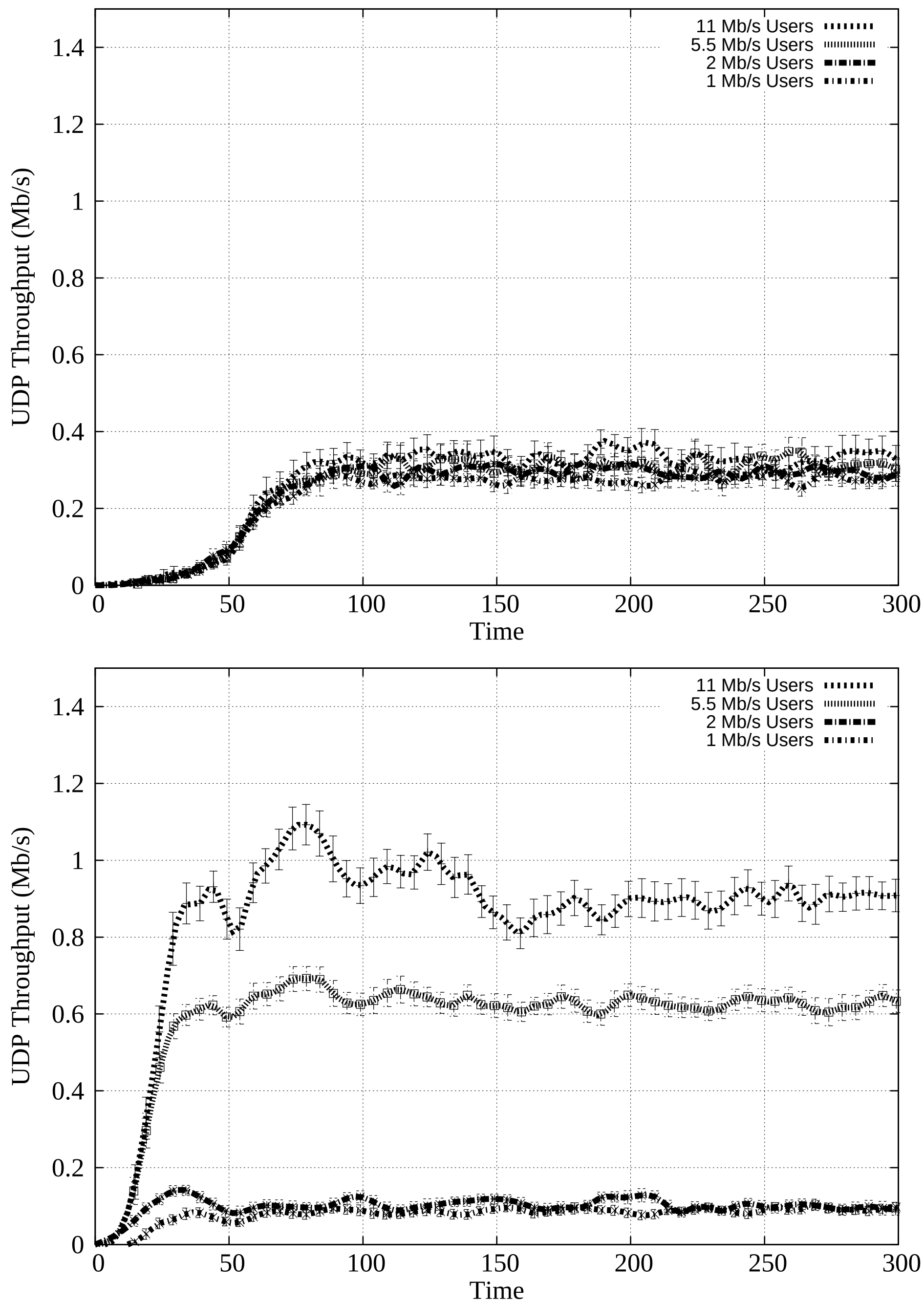


Figure 3.21: Throughput values observed in a network with 10 users per data rate groups of 1 Mb/s, 2 Mb/s, 5.5 Mb/s and 11 Mb/s, in MORAL disabled (top) and enabled (bottom) systems. MORAL creates throughput differentiation between different data rate groups and lets faster users achieve increased throughput levels.

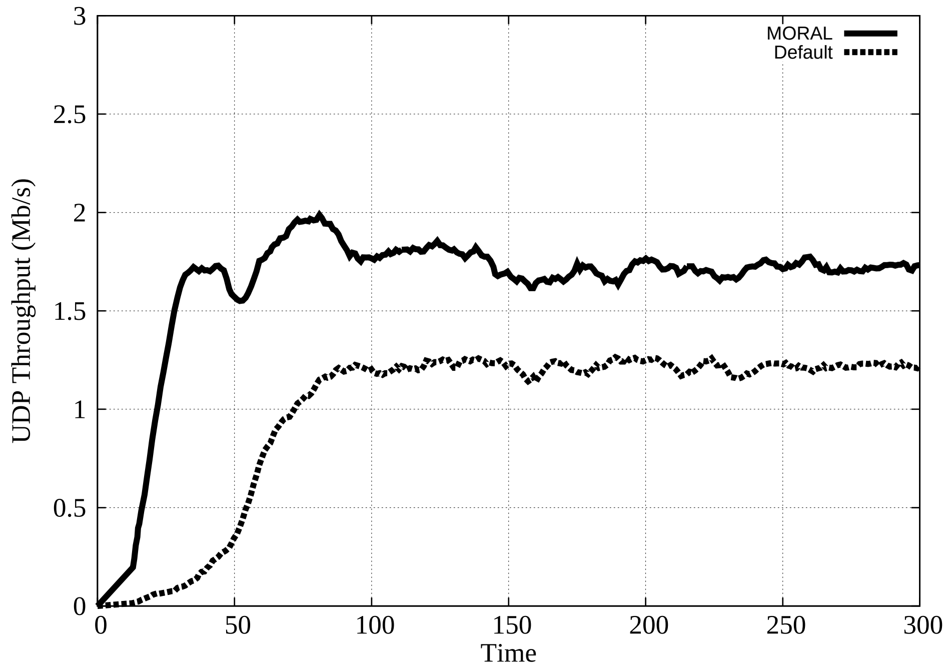


Figure 3.22: Total throughput values observed in a network with 10 users per data rate groups of 1 Mb/s, 2 Mb/s, 5.5 Mb/s and 11 Mb/s. MORAL improves the throughput by 43.0%.

the total throughput value increases 43.0% from 1.21 Mb/s to 1.73 Mb/s as can be seen in Figure 3.22.

In Figure 3.23, we report about the baseline fairness and the retry limits observed in the top and bottom graphs, respectively. Notice that as the system starts, the users immediately tend to admit different retry limits observing the other uses in the system. As expected, the lowest data rate group (1 Mb/s users) admits the highest data rate with average value of 9.6, and the other groups follow with 7.8, 1.8 and 1.3, correspondingly. One appealing result is that as the retry limit adjustment starts creating a differentiated environment right from the start,

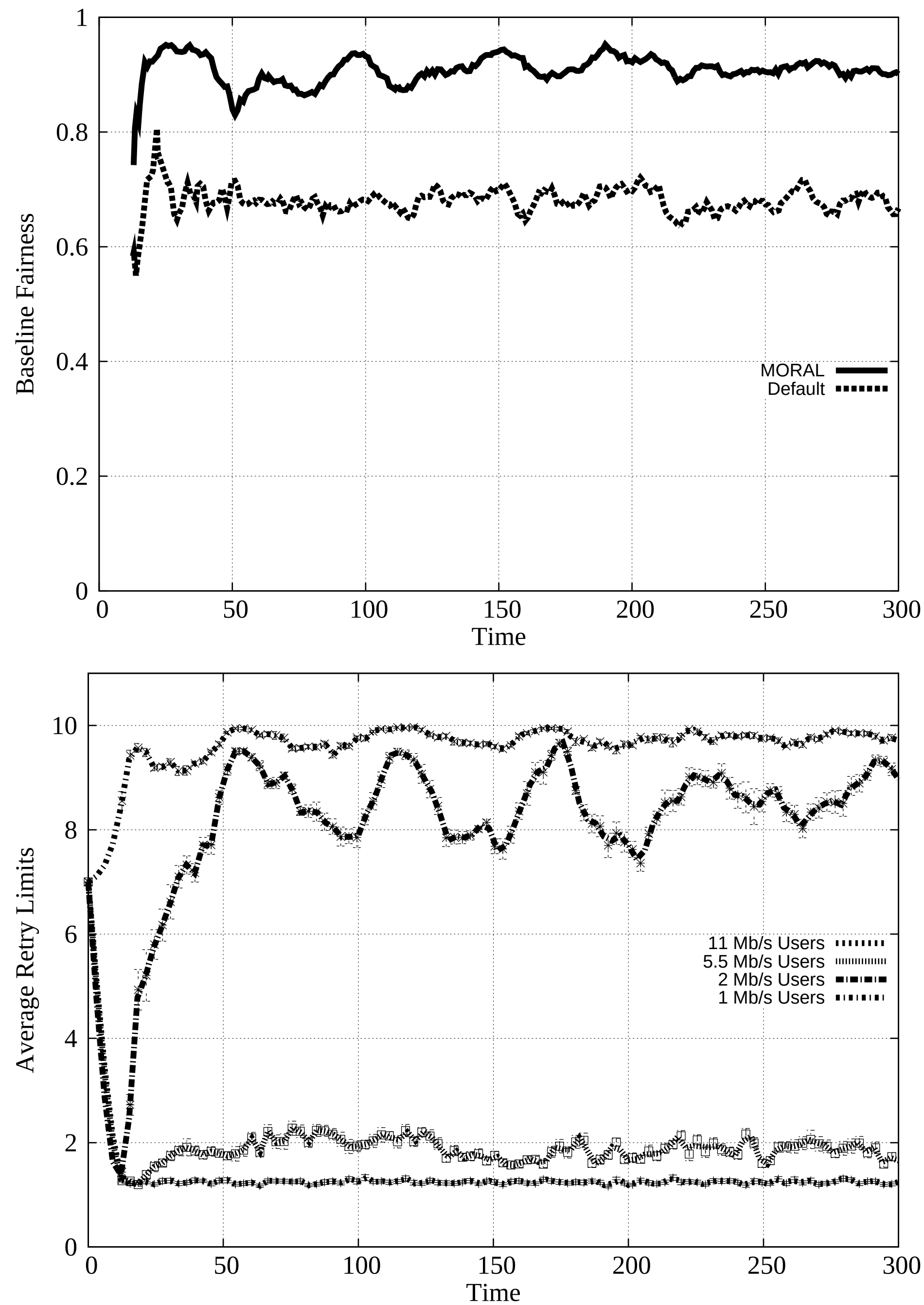


Figure 3.23: Average baseline fairness with the average retry limits assigned per user group by MORAL in a network with 10 users per data rate groups of 1 Mb/s, 2 Mb/s, 5.5 Mb/s and 11 Mb/s. MORAL improves the fairness by 31.8%.

the throughput increase in the MORAL-enabled case is more rapid as can be seen in Figure 3.22. In this figure, the MORAL-enabled scenario converges to the levels of equilibrium throughput in a faster manner (around $t = 40$), whereas it takes longer time for the default situation to reach the equilibrium level (around $t = 100$). This is because of the fact that the higher data rate users, which constitute the bigger share of the medium throughput, are immediately more favored even from the beginning of the simulation. Evidently, the MORAL-enabled case provides a better start-up process, promising an increased number of bits transferred successfully.

In a vastly heterogeneous environment with four data rates, MORAL constructs a highly fair environment with an improved baseline fairness of 31.8% from 0.68 to 0.90 and improved total throughput of 43.0% from 1.21 Mb/s to 1.73 Mb/s.

3.5.3.3 Scenario 3: Balanced Static 10-10-10 Case With Three Rates

Our third experiment is similar to the previous simulation with one difference: it only consists of three different data rate user groups, namely 1 Mb/s, 2 Mb/s and 5.5 Mb/s. As in the previous experiment, each user group contains 10 users. The expected default throughput calculated analytically is 0.35 Mb/s for each user group and coincides with our empirical results presented in the top graph of Figure 3.24. The bottom graph in the same figure shows the total throughput values for each group when MORAL is in effect. Notice that MORAL differentiates each group according to its capacities as expected, and as observed before.

The total throughput values achieved are 1.0781 Mb/s and 1.2433 Mb/s for

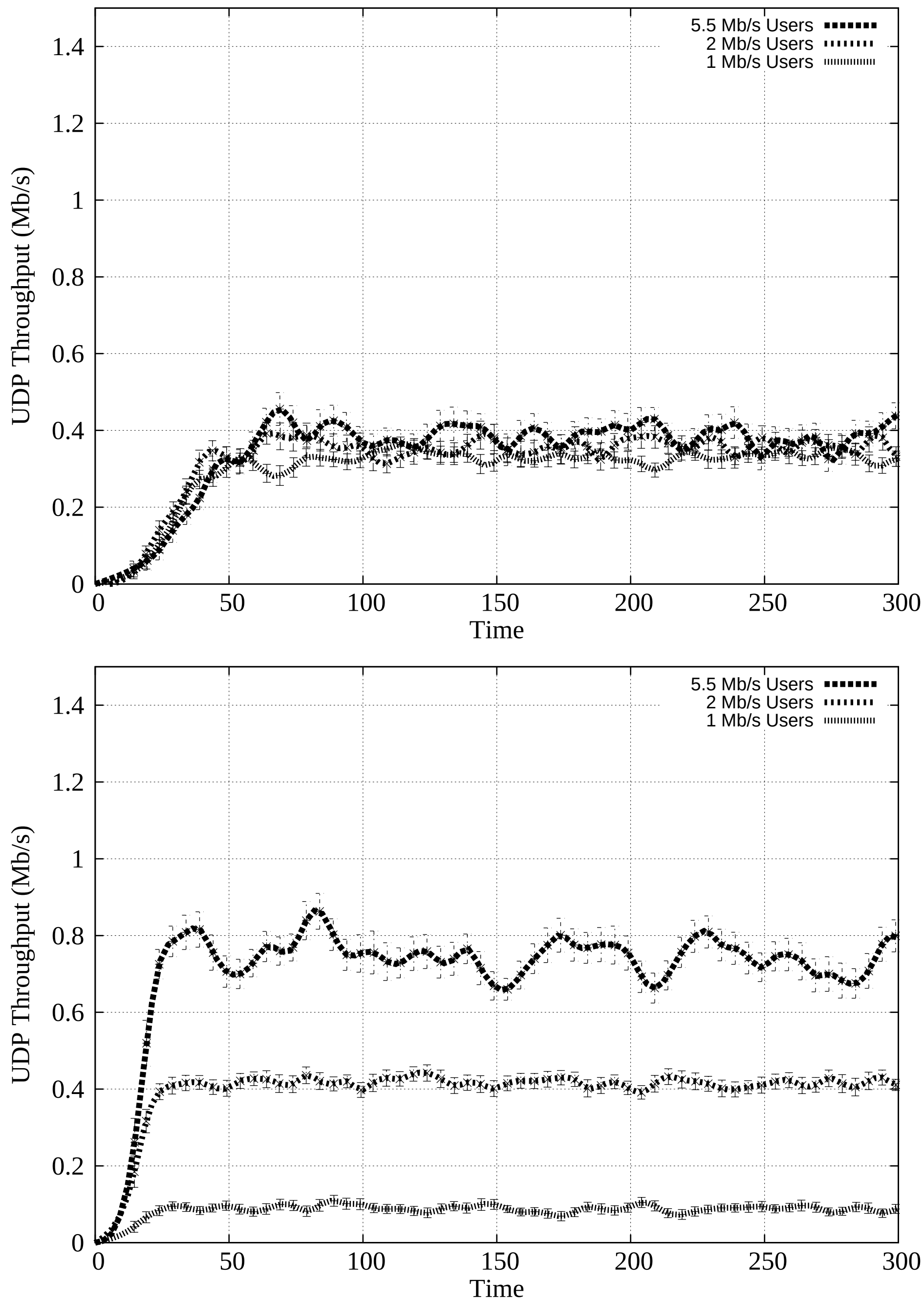


Figure 3.24: Throughput values observed in a network with 10 users per data rate groups of 1 Mb/s, 2 Mb/s and 5.5 Mb/s, in MORAL disabled (top) and enabled (bottom) systems. MORAL lets faster users achieve increased throughput levels.

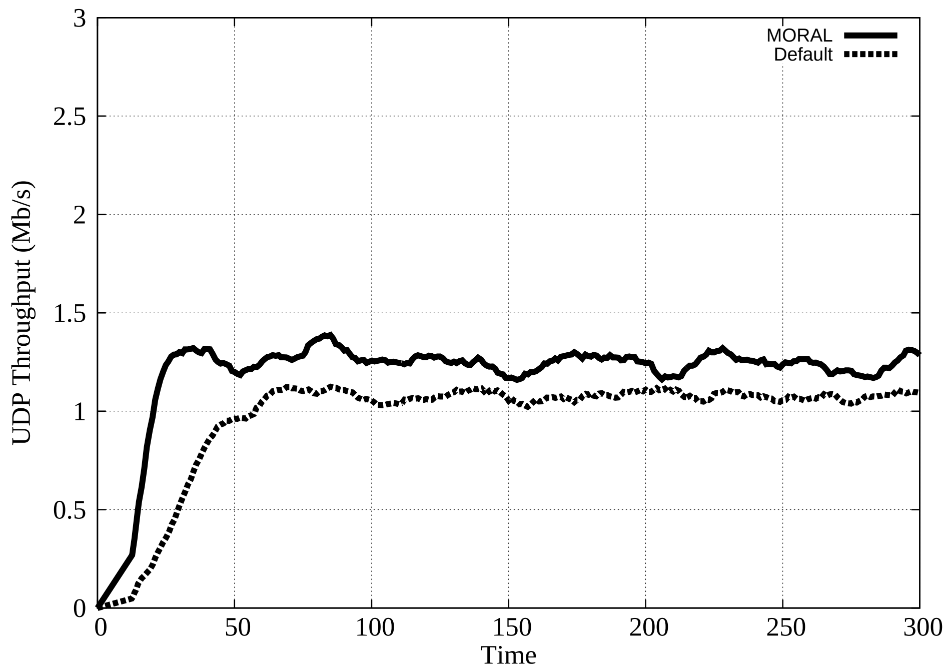


Figure 3.25: Total throughput values observed in a network with 10 users per data rate groups of 1 Mb/s, 2 Mb/s and 5.5 Mb/s. MORAL improves the throughput by 15.3%.

the default and MORAL-enabled cases respectively. MORAL improves the total throughput by 15.3% and the fairness by 12.8% from 0.79 to 0.89 as can be seen in the top graph of Figure 3.26. Bottom graph of the same figure demonstrates the average retry limits assigned per data rate group. MORAL assigns the retry limits of 9.64, 2.83 and 1.26 to the 1 Mb/s, 2Mb/s and 5.5 Mb/s users respectively.

Compared to the Scenario 2, the fairness value observed in the default scenario for this experiment is much better, close to 0.8. As the multi-rate anomaly is more damaging to the higher data rate users, observed fairness naturally improves in networks in which the speeds of lowest and highest data rate users are alike. As the speed difference of the users in a balanced network increases, fairness is expected to diminish further.

3.5.3.4 Scenario 4: Unbalanced Static 30-10 Case With Two Rates

In our next experiment, we consider an unbalanced network with 30 11 Mb/s users and 10 1 Mb/s users. Figure 3.27 presents the cumulative throughput values achieved for the default (top) and MORAL-enabled (bottom) scenarios. Since the number of users in each data rate group is different, the cumulative throughput values vary. However, the default scenario generates identical normalized throughput values per user and regardless of the data rate they are utilizing, each user achieves around 0.038 Mb/s throughput values for the default scenario. In the MORAL-enabled scenario, on the other hand, 11 Mb/s users achieve 0.067 Mb/s normalized throughput while the 1 Mb/s users achieve 0.0161 Mb/s, creating a fairer overall

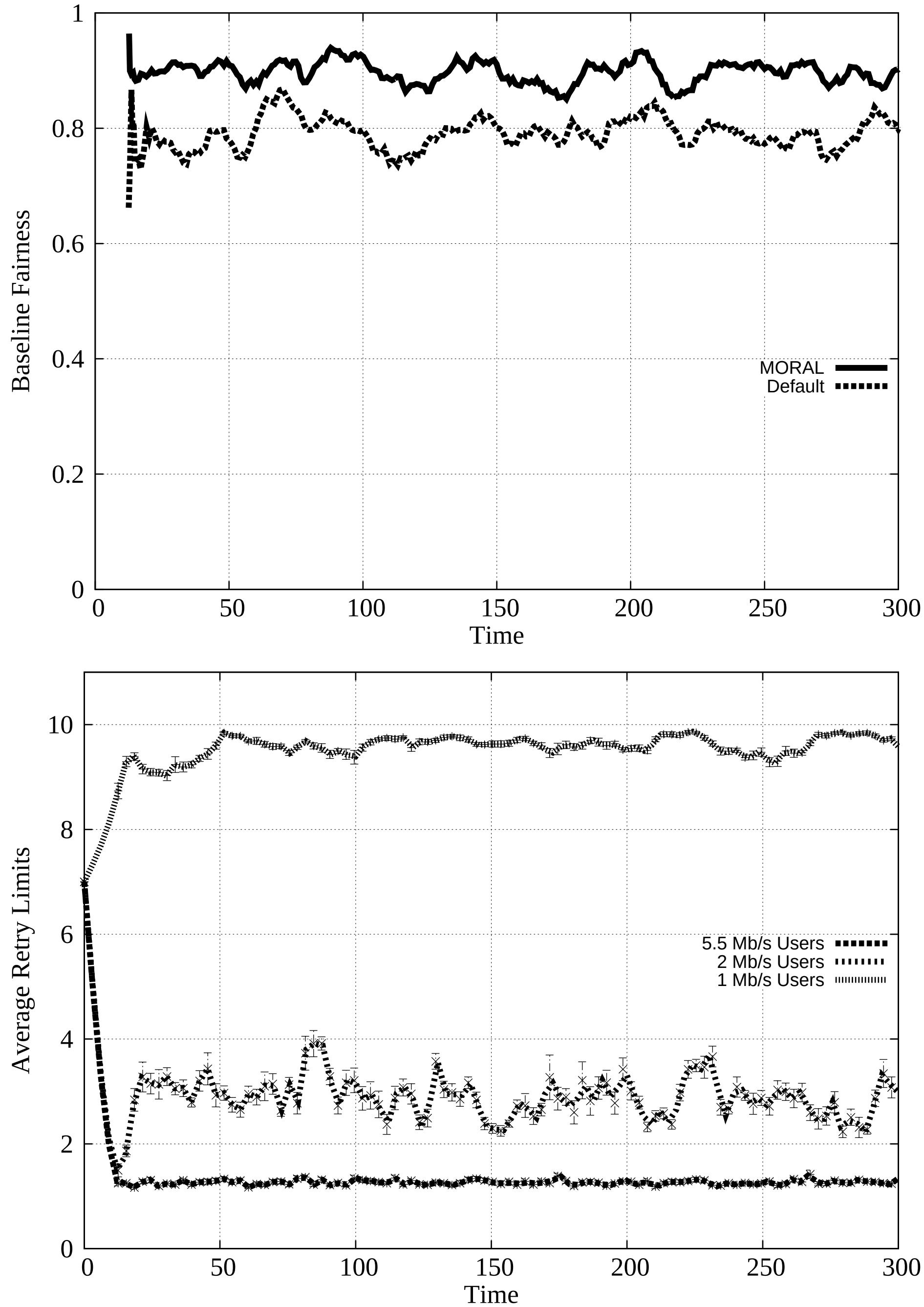


Figure 3.26: Average baseline fairness with the average retry limits assigned per user group by MORAL in a network with 10 users per data rate groups of 1 Mb/s, 2 Mb/s and 5.5 Mb/s. MORAL improves the fairness by 12.8%.

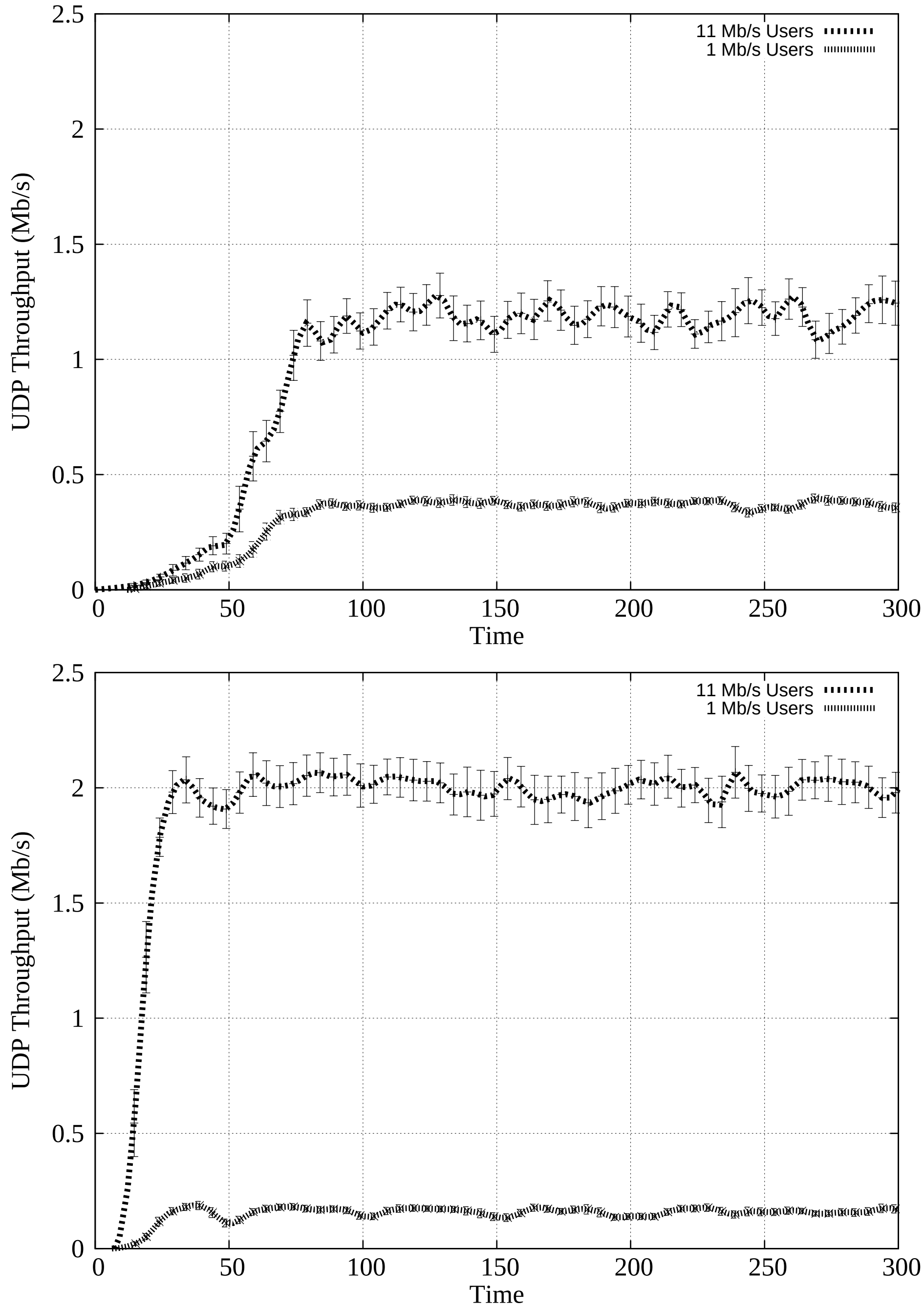


Figure 3.27: Throughput values observed in a network with 10 1 Mb/s and 30 11 Mb/s users in MORAL disabled (top) and enabled (bottom) systems. MORAL lets faster users achieve increased throughput levels.

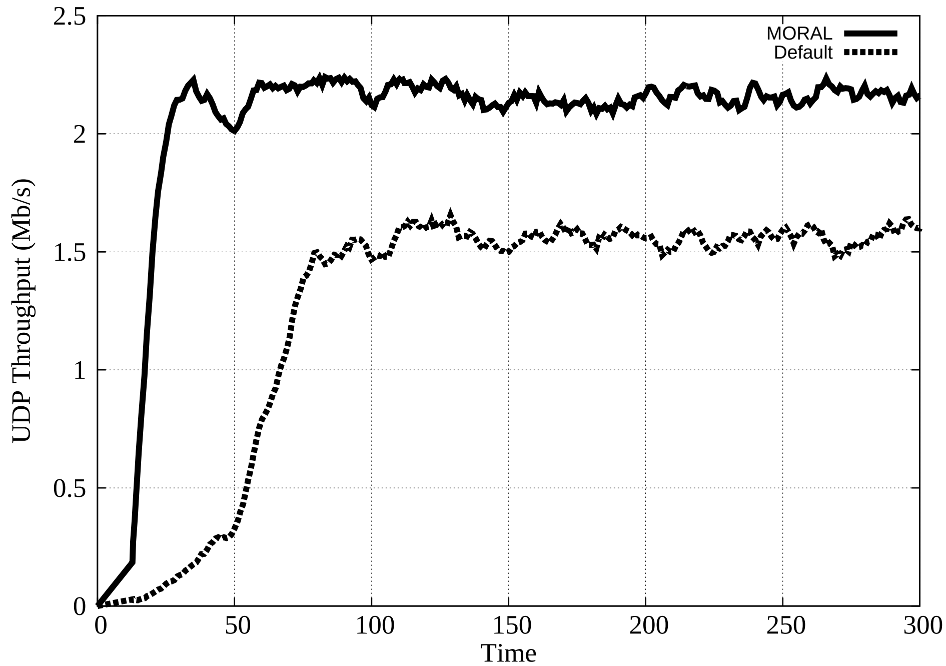


Figure 3.28: Total throughput values observed in a network with 10 1 Mb/s and 30 11 Mb/s users. MORAL improves the throughput by 37.9%.

network.

The total throughput of the network is improved by 37.9% from 1.57 Mb/s to 2.16 Mb/s using MORAL (Figure 3.28). Fairness is also improved vastly from 0.47 to 0.90 by 93.8% (Figure 3.29). Notice that as the network is unbalanced with majority of the users belonging to the high data rate group, the fairness values achieved in the default case are much lower than the values observed in the balanced networks. Multi-rate anomaly damages the fairness of the majorly high data rate unbalanced networks more severely as majority of the users observe very unfair throughput levels. For balanced networks, on the other hand, the effect of multi-rate anomaly lessens (compared to the current scenario) as the fairness calculation is weighted towards the larger groups. In extreme conditions in which a single data rate group constitutes a large portion of the entire user set, the fairness achieved is expected to be very high as the users from the largest data rate group almost entirely represent the whole network. This phenomena has been discussed before in Section 3.4 and it has been argued that as the networks become closer to homogenous systems, fairness naturally improves.

3.5.3.5 Scenario 5: Unbalanced Static 10-30 Case With Two Rates

Similar to the previous scenario, our next experiment focuses on an unbalanced network as well. However, unlike to the previous scenario, the slower users constitute the bigger portion of the network with 30 1 Mb/s users and 10 11 Mb/s users. Figure 3.30 illustrates the cumulative throughput values observed in MORAL-disabled (top) and enabled (bottom) scenarios. Notice that the cumulative throughput of 1 Mb/s users is larger than the 11 Mb/s users for the default case. This is an expected behavior as the number of slow users is larger than the number of fast users. However, MORAL changes the network vastly, increasing the total throughput observed by the fast users (with 10 users) to levels more than the slow users (with 30 users)!

MORAL increases the total throughput by 48.0% from 0.75 Mb/s to 1.11 Mb/s (Figure 3.31). Fairness is also increased by 20.8% from 0.82 to 0.99 (Figure 3.32).

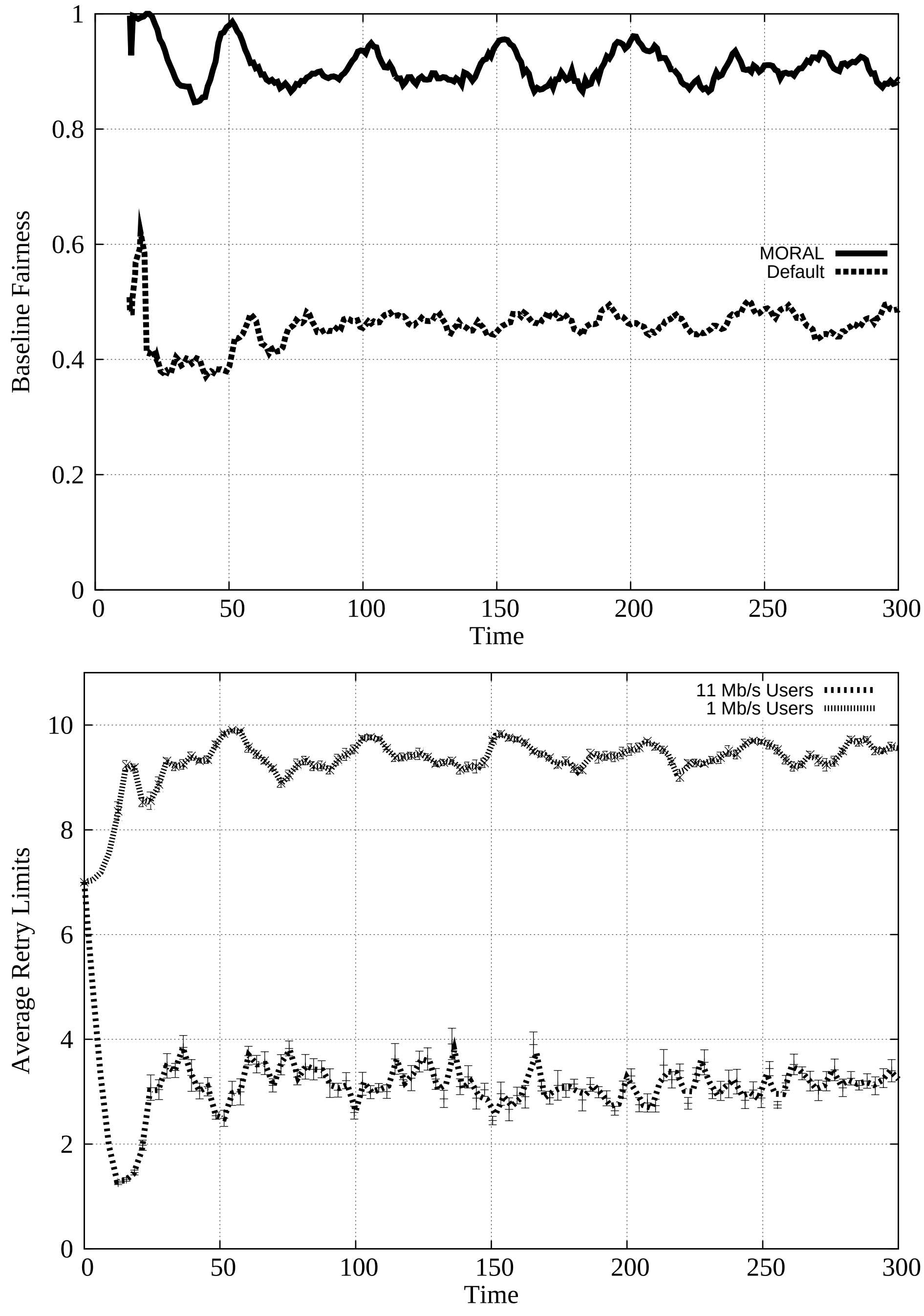


Figure 3.29: Average baseline fairness with the average retry limits assigned per user group by MORAL in a network with 10 1 Mb/s and 30 11 Mb/s users. MORAL improves the fairness by 93.8%.

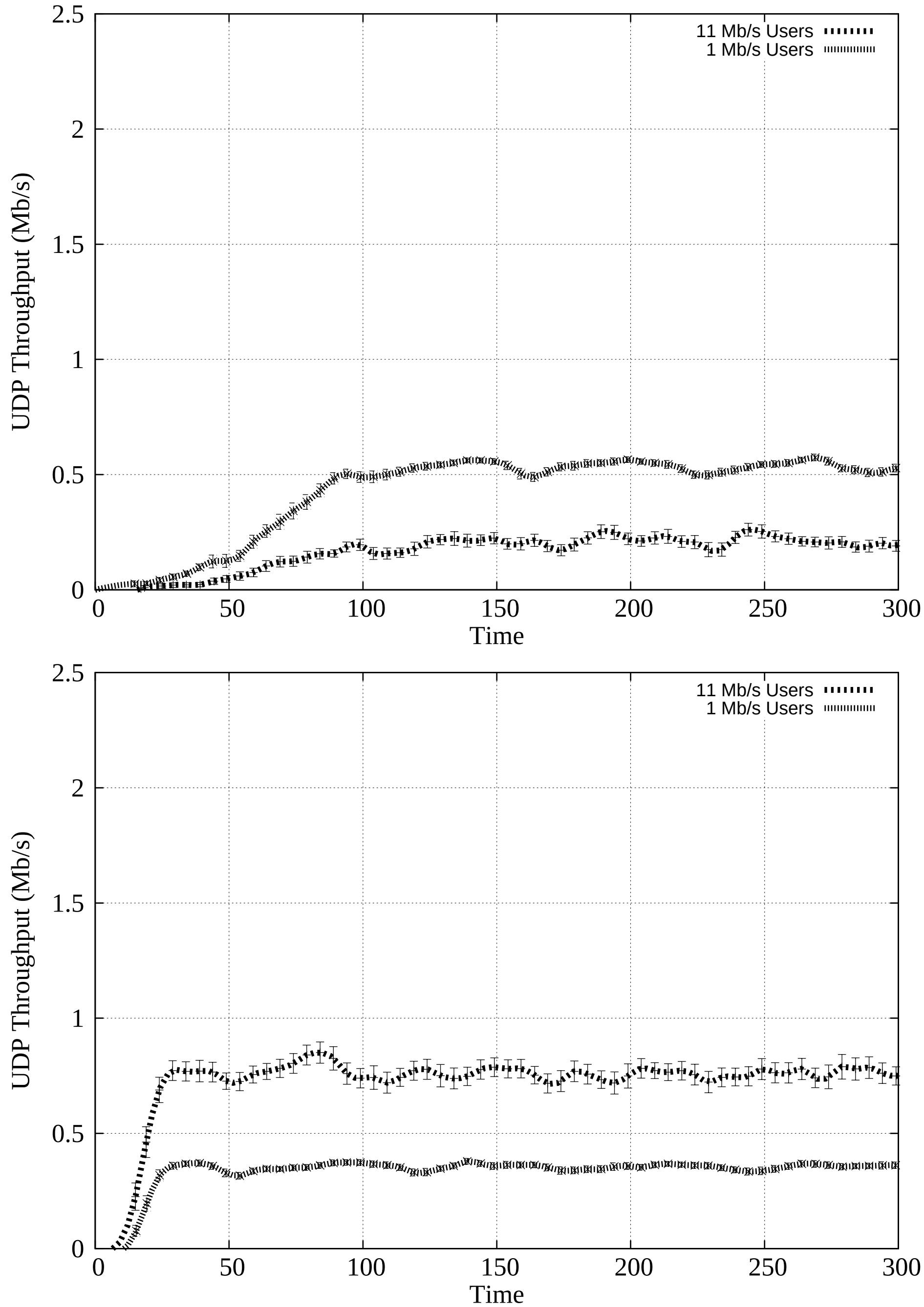


Figure 3.30: Throughput values observed in a network with 30 1 Mb/s and 10 11 Mb/s users in MORAL disabled (top) and enabled (bottom) systems. MORAL lets faster users achieve increased throughput levels.

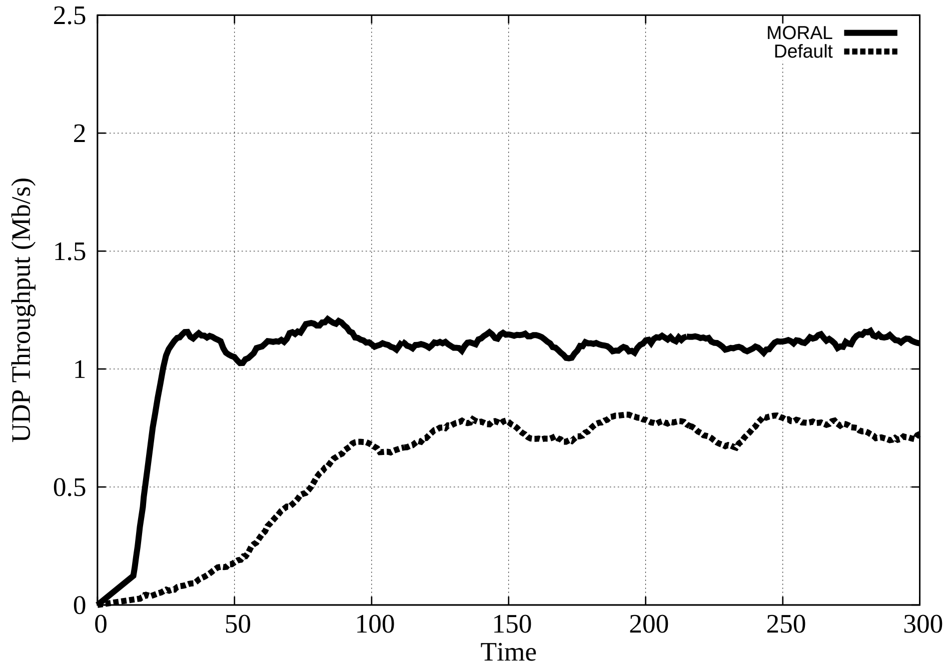


Figure 3.31: Total throughput values observed in a network with 30 1 Mb/s and 10 11 Mb/s users. MORAL improves the throughput by 48.0%.

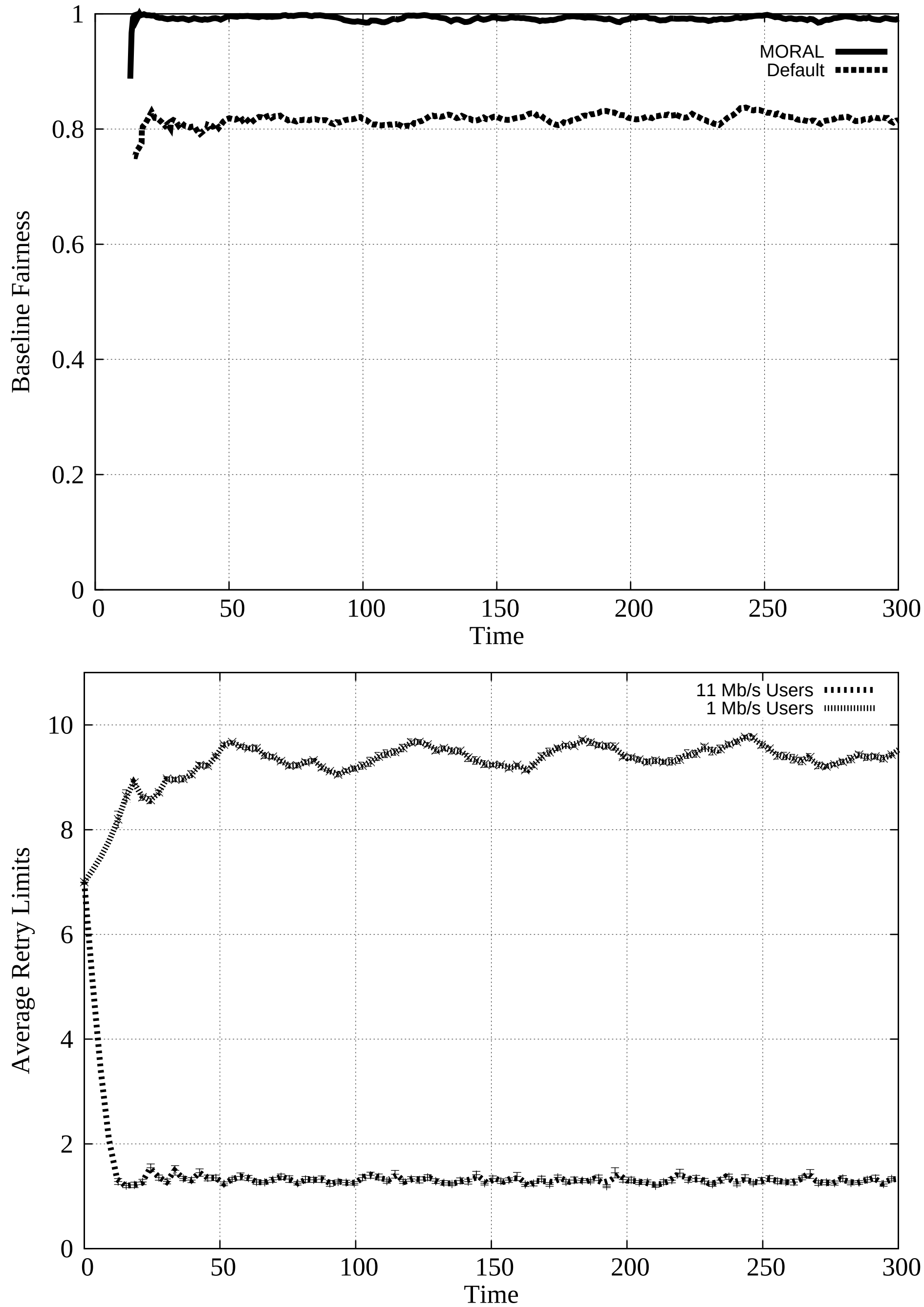


Figure 3.32: Average baseline fairness with the average retry limits assigned per user group by MORAL in a network with 30 1 Mb/s and 10 11 Mb/s users. MORAL improves the fairness by 20.8%.

3.5.3.6 Scenario 6: Balanced Dynamic 20-20 Case With Two Rates, Case of Arrival

Our next experiment focuses on the performance of the MORAL algorithm in dynamic environments where nodes are not placed at specific locations but rather move in certain directions. As in the previous examples, the system consists of 20 11Mb/s users and 20 1Mb/s users. In contrast to the previous examples, this time fast users are located very close the base station and static; whereas slow users are far away from the base station at the beginning of the experiment and moving towards the base station with constant speed. At around $t = 600$, moving nodes reach the destination and do not travel afterwards.

The link adaptation mechanisms have the main aim of improving the throughput for bad channel conditions. As the distance between the transmitter and the receiver affects the quality of the channel vastly, a thorough link adaptation mechanism would lower the data rate in case of large transmission distances. Hence, the scenario presented in this section is expected to be observed frequently in networks armed with link adaptation mechanisms.

The throughput values seen in this dynamic environment are shown in Figure 3.33, in which the top graph shows the throughput values seen by each data rate group using the default retry limits and the bottom graph shows the MORAL-enabled scenario, as in the previous experiments [14] . Just before $t = 300$, the

[14]The effective throughput of IEEE 802.11b is lower than the maximum physical throughput because of the overheads such as 1) the requirement of transmitting specific parts of the header using basic lower data rates for backward capability reasons 2) acknowledgement packet required

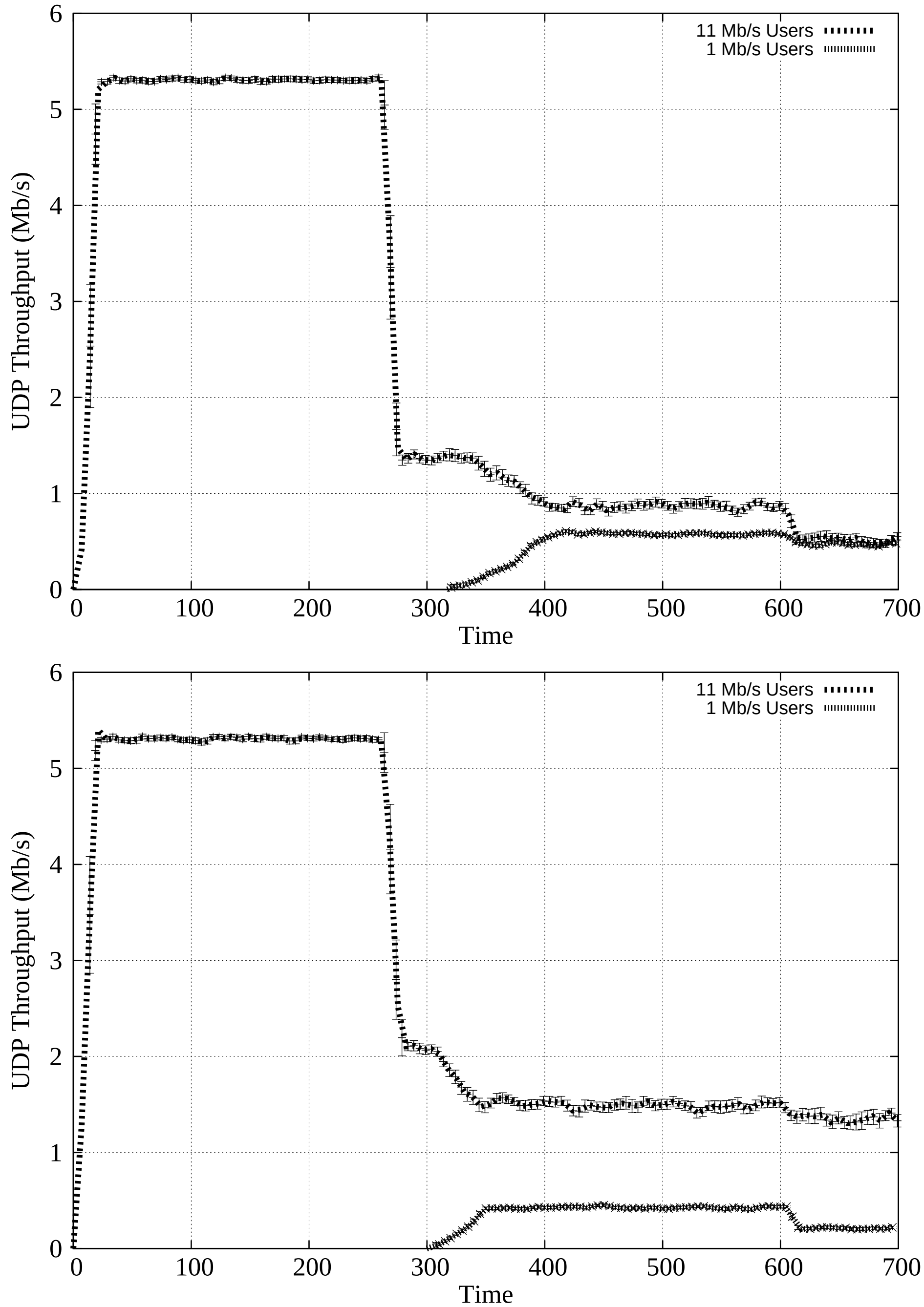


Figure 3.33: Throughput values observed in a network with 20 1 Mb/s and 20 11 Mb/s users in MORAL disabled (top) and enabled (bottom) systems. Slow users are initially away from the fast users and are in motion towards the core of the network. MORAL lets faster users achieve increased throughput levels.

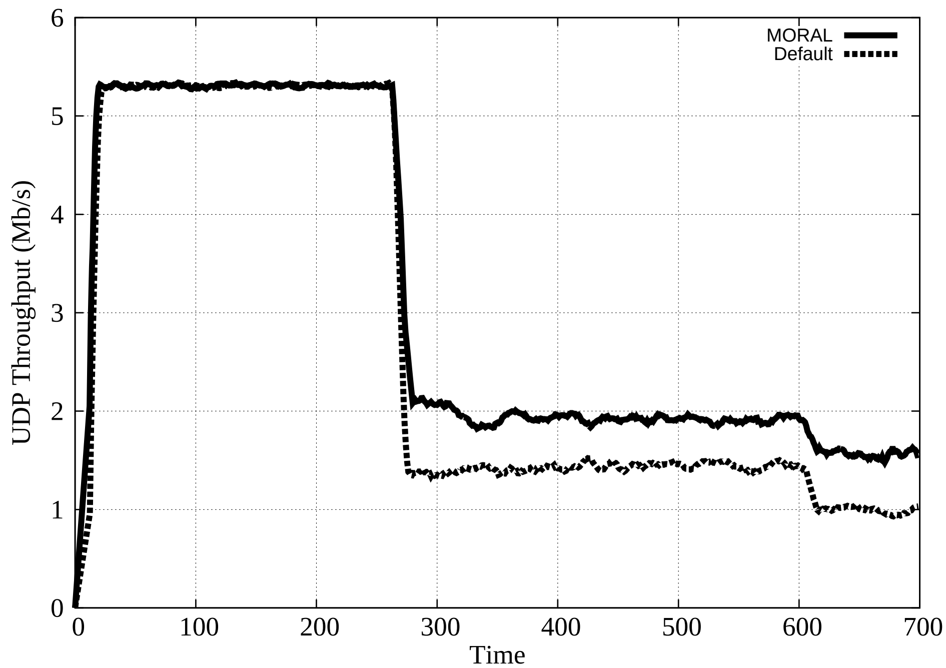


Figure 3.34: Total throughput values observed in a network with 20 1 Mb/s and 20 11 Mb/s users. Slow users are initially away from the fast users and are in motion towards the core of the network. MORAL improves the throughput by 40.2% for the critical time range of $[300, 650]$.

attempted transmissions of the lower data rate users start to affect the higher data rate users, even though this traffic does not convert into actual throughput for the lower data rate users since these users are still far away from the base station. After $t = 300$, the lower data rate users start enjoying better channel characteristics. Having the same medium access privileges with their higher data rate counterparts, the lower data rate users' traffic pulls the observed throughput of the higher data rate users close to its level. Notice that, in the default scenario, throughput achieved by high data rate users is slightly larger than the throughput achieved by the lower data rate users in the $t \in [300, 600]$ interval. This because of two reasons: 1) since the fast users are close to the base station, the received signal strength of their transmission are larger than the other and they capture the physical channel and observe higher throughput [17]; 2) the slow users observe high levels of packet loss ratios and consequently utilize their backoff states longer than the faster users, thus letting the faster users observe more channel idle times. These phenomena are precisely what MORAL exploits with its arbitrary retry limit assignments. After the slow users reach the destination, the impact of these phenomena fades and all the users observe same throughput levels starting at $t = 600$.

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for each packet successfully transmitted 3) the inter-frame spacing times leaving the medium idle and unused; and 4) the channel contention and collided transmissions. Inter-frame spacing and the acknowledgment requirements reduce the achieved throughput to 5.6 Mbps [58] with the theoretical maximum throughput efficiency of $\sim$ 50% for 11 Mbps transmissions [81]. Similar results are achieved through DCF analysis [35, 155] and real-life measurements as well [29, 113]. Hence the throughput achieved in Figure 3.33 in the homogenous environment (i.e. $t \in [50, 250]$) is around 5.5 Mb/s, which is the maximum achievable throughput of 11 Mb/s IEEE 802.11 transmissions.

Once MORAL is enabled, the fast users start to decrease their retry limits immediately when they detect slower users in their vicinity, as in Figure 3.35 bottom graph. This action avoids the extreme throughput decrease for the fast users and forces the system to reach the equilibrium faster at around $t = 350$, compared with the equilibrium time of $t = 400$ of the default case. Notice that, even though in interval $[300, 400]$ slow users do not overhear any multi-rate transmissions, and hence keep their retry limit at the default value of 7, the immediate action of the fast users is sufficient to keep the higher data rate users maintain a larger throughput value. In fact, this outcome is in accordance with the analytical analysis drawn in previous sections. In Figure 3.14, it is shown that, even for $k_1 = 7$ default value, the fairness and total throughput values increase vastly if low k_{11} values are chosen.

After $t = 600$, the outcomes of capture effect lessen and the fast users start to observe more collisions. Consequently, they start to observe more Case I instances (Section 3.5.2.1) with no successful transmissions overheard (i.e. $c_{current} = 0$ and $c_{calculated} = 0$) and attempt to increase their retry limits more frequently. As a result, the dominance of the fast user traffic on the medium lessens, and the slow users more frequently perceive the network as homogenous, attempting to decrease their retry limits more often. Hence, after $t = 600$, the average retry limit increases for fast users and decreases for the slow users.

Finally, for the experimental Scenario 6, our simulations produce a total throughput increase of 40.2% from 1.38 Mb/s to 1.88 Mb/s (Figure 3.34) and a fairness increase of 23.9% from 0.71 to 0.88 (Figure 3.35), for the critical time range of $[300, 650]$ which is the approximate time interval between the moment the network

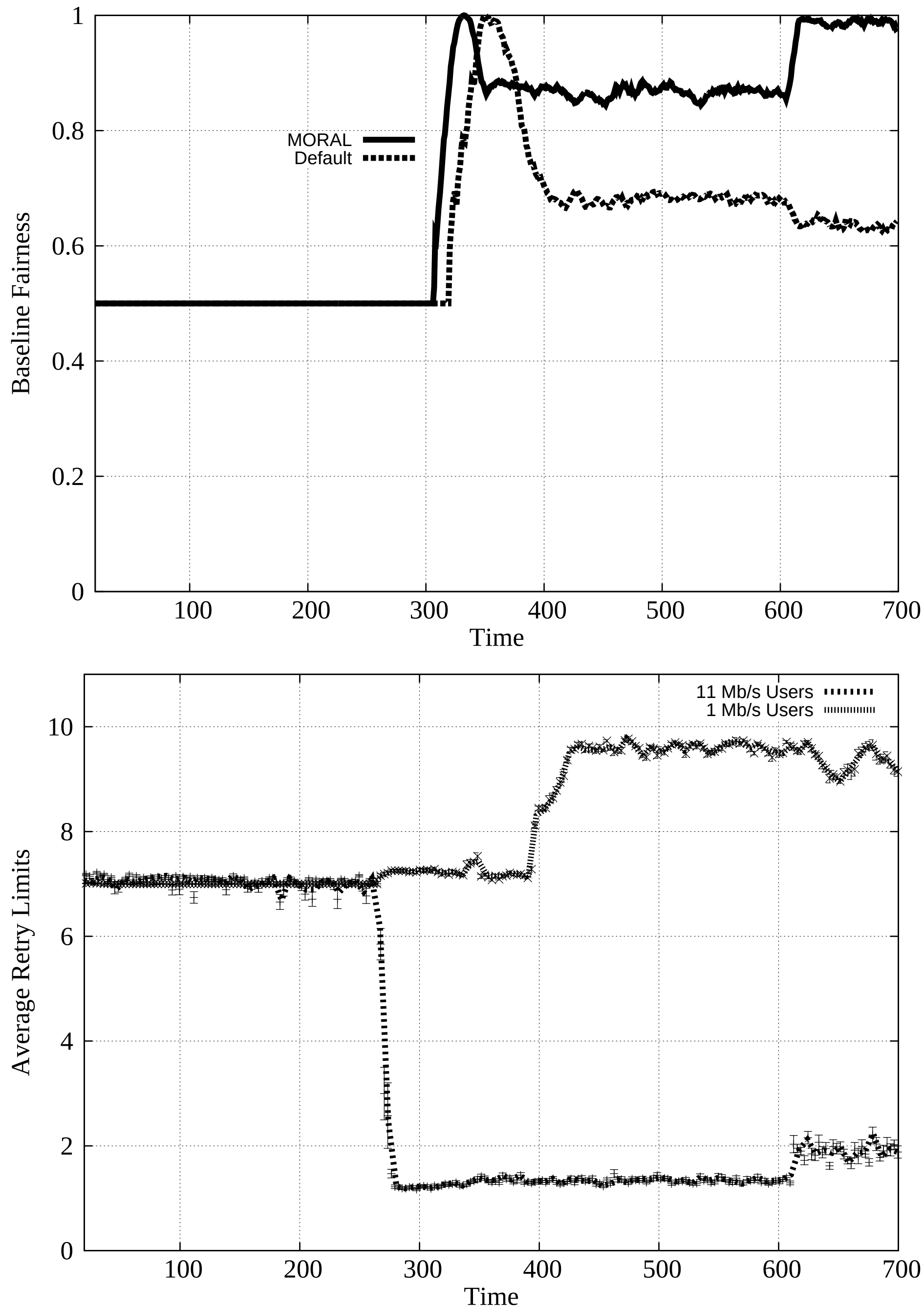


Figure 3.35: Average baseline fairness with the average retry limits assigned per user group by MORAL in a network with 20 1 Mb/s and 20 11 Mb/s users. Slow users are initially away from the fast users and are in motion towards the core of the network. MORAL improves the fairness by 23.9% for the critical time range of $[300, 650]$.

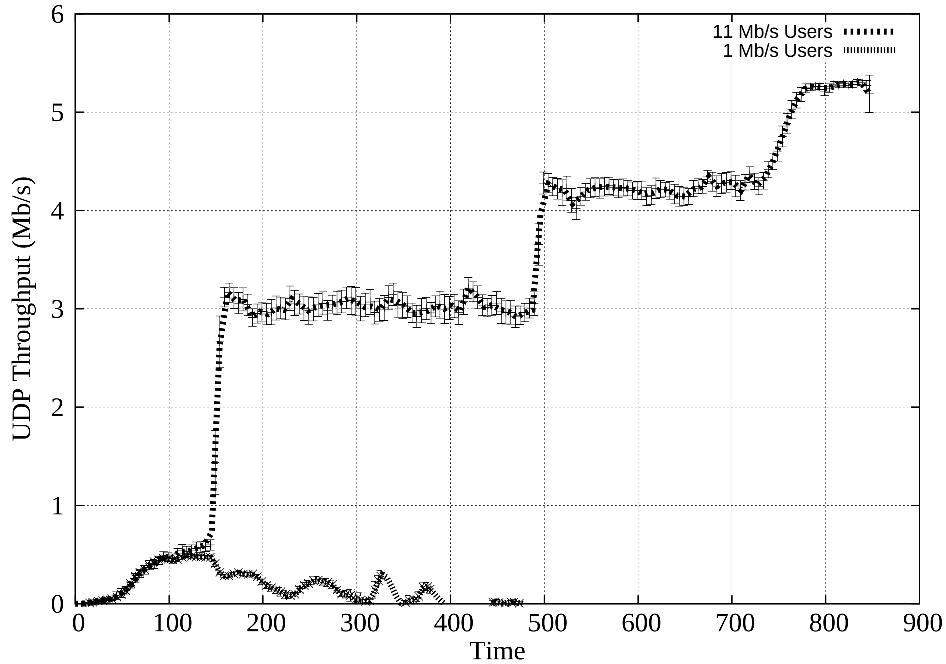
becomes multi-rate and the moment the slow users reach the base station.

3.5.3.7 Scenario 7: Balanced Dynamic 20-20 Case With Two Rates, Case of Leave

Our last experiment is the continuation of the previous dynamic scenario presented as Scenario 6 in Section 3.5.3.6, and targets the situation where a system at equilibrium is disturbed by the leave of the lower data rate users from the system. We start this simulation exactly at the point where the previous scenario is left at and start the leave process of low data rate users, i.e. Group 2, at $t = 150$.

Once the slow users start moving, the throughput values of the higher data rate, fast users immediately increase with the increased packet error rate of former because of two reasons: 1) Since the slow users now become away from the basestation further than the fast users, they observe more packet losses caused by frame errors than the others. 2) Because of the physical capture effect studied before [96], packets generated by fast is going to have a higher change of being successfully received. Increased packet error rate implies longer time in the backoff stages for slow users, exacerbating the effects of the short-term fairness characteristics of the wireless media access protocols [92], creating an effect similar to the Ethernet capture effect [125].

MORAL, on the other hand, does not let the fast users take over the whole medium by taking advantage of the poor channel characteristics of slow users. Observing the decreased channel access times of the slow users, faster users re-adjust



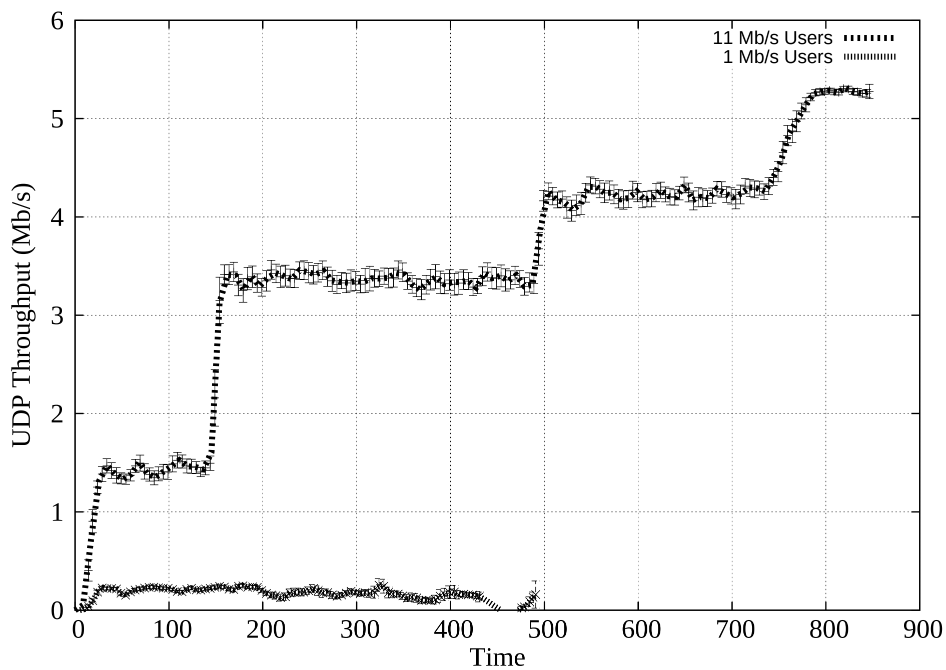


Figure 3.36: Throughput values observed in a network with 20 1 Mb/s and 20 11 Mb/s users in MORAL disabled (top) and enabled (bottom) systems. Slow users are initially very closely located to the fast users and move away from the core of the network. MORAL lets faster users achieve increased throughput levels.

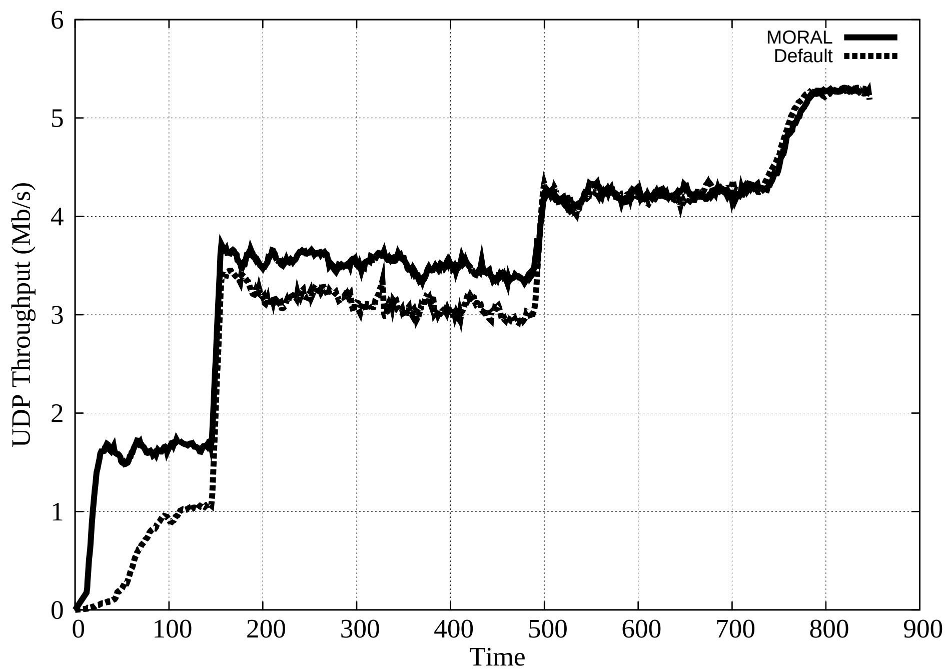


Figure 3.37: Total throughput values observed in a network with 20 1 Mb/s and 20 11 Mb/s users. Slow users are initially very closely located to the fast users and move away from the core of the network. MORAL improves the throughput by 11.9% for the critical time range of $[150, 500]$.

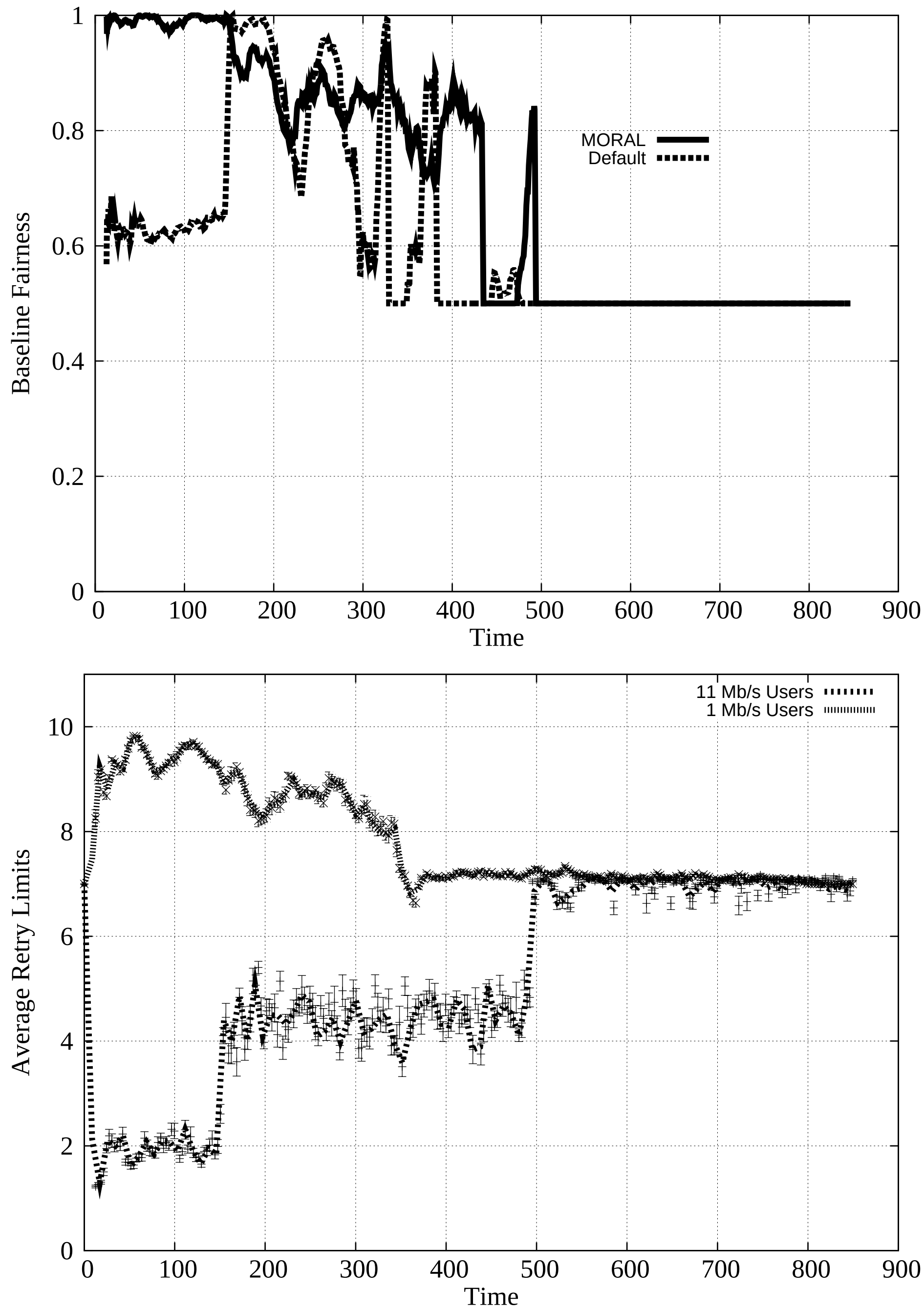


Figure 3.38: Average baseline fairness with the average retry limits assigned per user group by MORAL in a network with 20 1 Mb/s and 20 11 Mb/s users. Slow users are initially very closely located to the fast users and move away from the core of the network. MORAL improves the fairness by 12.6% for the critical time range of $[150, 500]$.

their access characteristics via increasing their retry limits, as can be seen in Figure 3.38 bottom graph. Notice that when the system was in equilibrium before the slow users start moving, fast users admit a very low retry limit (around 2, from Figure 3.38). However, once the movement starts, the average retry limits of the fast users immediately increases to values close to 4.5. As the slow users are just about to leave the stage completely (around $t = 400$), there is a blackout period for the slow users as they cannot hear the others' traffic, which forces them to lean to the default value of 7 as they conceive the environment as homogenous. Once the slow users are totally out of the range, fast users immediately lean to the default value.

The fairness values observed in this scenario is among the trickiest as the MORAL-enabled case favors the higher data rate user. Because of the physical and Ethernet capture effects defined above, fast users enjoy higher throughputs than even expected! Notice that even the default, MORAL-disabled case produces peak fast throughput values just around 3 Mb/s, much higher than the expected value of 2.4 Mb/s. As the higher data rate users hold this high throughput values, the fairness seems to be negatively affected. However, the overall fairness is still improved by MORAL, even though the amount of improvement is smaller compared to the other scenarios.

MORAL increases the total throughput by 11.9% from 3.15 Mb/s to 3.52 Mb/s and fairness by 12.6% from 0.70 to 0.79 during the leave process in the time interval $[150, 500]$ (Figure 3.37).

Figure 3.39 summarizes the level of throughput and fairness increase gained

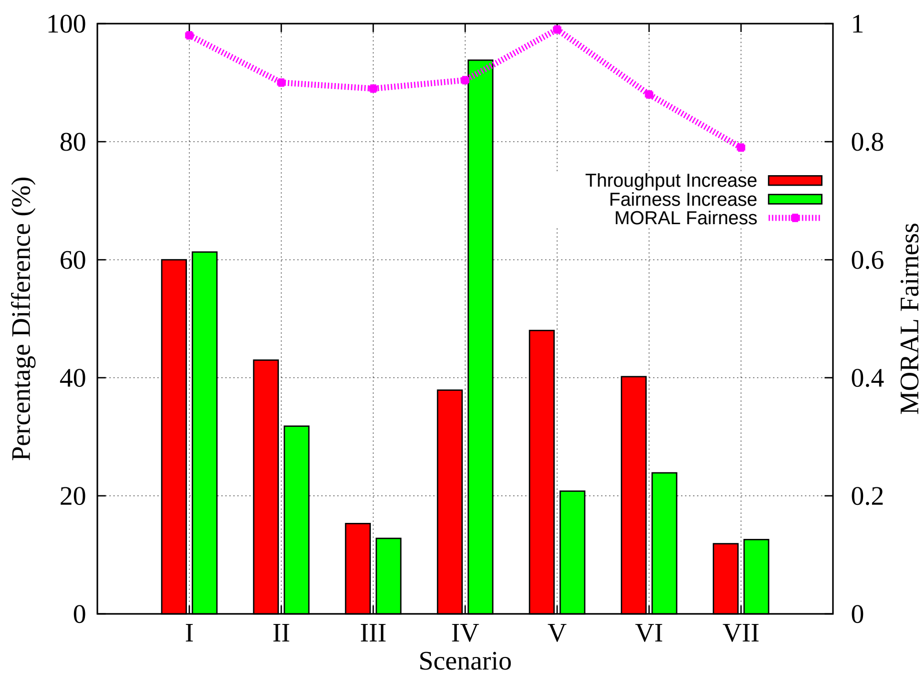


Figure 3.39: Average throughput and fairness changes for the entire experiment set. MORAL improved both the throughput and the fairness values observed in all of the simulation scenarios.

through MORAL, shown for all of the seven scenarios. MORAL promises better performance in both aspects and maintains very high levels of baseline fairness, more than 0.8 for all of the above scenarios.

3.6 Discussions

3.6.1 Packet Losses are MORAL-ly Preferred

MORAL targets at maintaining differentiation between users through arbitrary retry limit assignments. Even though such assignments provide enough differentiation for highly saturated conditions, the amount of improvement observed depends largely on the competitiveness of the environment. In order for the retry limits to be influential, the users need to observe several consecutive collisions and reach their retry limits causing failure notifications. If the users do not observe any packet losses, they will not utilize the BEB algorithm advertised, causing the retry limits assigned to be irrelevant.

However, collisions are not the sole reason for the packet losses and a successful transmission of a packet heavily depends on the channel characteristics as well. Indeed, the link adaptation mechanisms stated in the previous sections (Section 2.2) aim to resolve such bad conditions. Hence, if a user has chosen a lower data rate than the usual, it is highly likely that s/he is observing bad channel characteristics and has decided to lower his/her transmission rate in order to overcome these difficulties. Thus, it is highly likely that either this user's transmission link is very noisy or the user is relatively far away from the destination. Under these conditions, the probability that the user will observe collisions is still higher than a channel with perfect conditions (minimal noise and low distance between transmitter and the receiver). Notice that even if these conditions are not met, i.e. all of the users are observing perfect channel conditions and the collision rates are very low, MORAL cannot hurt the performance of the existent wireless system as the users will not reach their retry limits and will be able to transmit their packets without packet losses.

As an interesting side note, notice that through assigning lower retry limits, MORAL might contribute to the competitiveness of the environment as well. However, because of its preventive approach of eliminating potential starvation scenarios as soon as possible, MORAL accepts a more conservative role and does not explore this feature.

Hence, even though MORAL promises better fairness and total throughput levels for competitive environments; the performance improvement for wireless systems with good channel characteristics and low collision rates is not as promising as otherwise. However, even under such conditions, MORAL does not cause any harm to the system.

3.6.2 Transmission Cycle Length and MORAL

The transmission cycle is the time interval during which each user collects information about its neighbors. Each transmission cycle starts when the next packet to be transmitted is fetched from the upper layer buffers and ends when it is transmitted or discarded. Thus, each transmission cycle represents the time interval between two contention window reset operations.

Variation in the retry limits implies variation in the transmission cycle lengths. Lower retry limit users will monitor the environment for shorter times compared to the other users in the system; but they will act faster and re-compute their retry limits more rapidly. This self-timing mechanism of the transmission cycles has the following advantages:

1. Assigning very low retry limits to the users might push the system into starvation as the *CA* part of the CSMA/CA algorithm will not be able create enough randomness for the users competing for the channel, causing extremely severe channel conditions as discussed before in Section 3.4. Under these circumstances, the users are expected to recover fast from starvation and adapt better to the channel conditions through accepting higher retry limits. As smaller retry limits also mean shorter transmission cycles, the users are forced to re-evaluate their short retry limit choices quicker; and roll back to higher retry limits if they observe starvation on the channel.

2. Larger retry limits, on the other hand, imply existence of other, faster users in the network. In such conditions, decreasing the retry limit of such slow users will affect the fast users' throughput levels and the fairness of the overall system negatively. Since larger retry limits mean longer transmission cycles, the slow users will observe the channel long enough to ensure that there is no fast users in the vicinity that can be harmed by a potential retry limit

decrease.

The self-timing property of MORAL algorithm contributes to the correct network operation through eliminating unnecessary retry limit reduction, avoiding starvation and improving fairness.

3.6.3 Potential Link Adaptation Backfire and Solutions

MORAL relies on packet losses in order to create differentiation among users, but at the same time it increases the competitiveness of the network through assigning lower retry limits to specific user groups. This, consequently, might increase the number of consecutive packet losses observed and can trigger the physical link adaptation mechanisms in use.

In the literature, several link adaptation mechanisms utilize the packet losses as their main criterion for adjusting the transmission parameters [82, 94, 40, 77, 2]. In these works, if several packet losses occur, the wireless nodes attempt to improve the link performance through data rate adjustments.

MORAL, on the other hand, might increase the instantaneous packet losses observed through assigning lower retry limits to faster data rate users; and might tempt the link adaptation mechanisms to consider lowering the data rate in a premature way. SNR-based link adaptation mechanisms [69, 62, 63, 112, 120, 123], on the other hand, rely on the quality of the packet received and do not expose this problem.

This problem arises because of the lack of synchronization between the physical link adaptation mechanisms and MORAL. The following proposes two tactics to mitigate this issue:

1. It can be assumed that the physical link adaptation mechanisms in effect have a way of differentiating between packet losses caused by collisions and by the channel conditions. Several past studies already propose mechanisms in order to detect the cause of packet losses [89, 105, 69, 111].

2. The link adaptation mechanisms can be closely synchronized with MORAL in order to better estimate the packet loss ratio. The threshold-based mechanisms such as [82, 94] can consider the current retry limit in use for their calculations. For instance, ARF [82] lowers the transmission rate if two consecutive acknowledgement packets are missed. As an alternative to this methodology, for arbitrary retry limit r_i, user n_i might consider a normalization function ($f(r_i)$) and consider lowering its transmission rate according to a new criterion (for example, $f(r_i) * 2$). The details of such a normalization function are out of the scope of our work.

Using these techniques, the physical link adaptation mechanisms and MORAL can be integrated in a better manner, avoiding possible unnecessary physical adaptation attempts.

3.6.4 TCP and MORAL

Much like the physical transmission choices affecting the MAC layer (which is our main motivation for MORAL), the MAC layer choices affect the layers above as well. We discuss one of such effects in Chapter 4 for the network layer and propose improvements. In this section, we elaborate on the interaction between MORAL and the transport layer, specifically TCP. As TCP is a very complicated and vigorous mechanism, we do not delve into the protocol details and suggest the interested readers to consult to several excellent sources on the subject [116, 14, 74, 132].

The foremost aim of the transport layer is to provide the services of reliable data transfer, flow control, congestion control and multiplexing for multiple application processes. One of the earliest of transport protocols, User Datagram Protocol (UDP) [118], is a connectionless, best-effort service which provides simple and fast transport functionality, and often used for streaming applications. UDP provides rudimentary reliability through checksums and does not offer any congestion control mechanisms.

Transport Control Protocol (TCP) [14], on the other hand, is a connection-oriented protocol promising reliable, in-order data delivery with congestion and flow control. TCP controls the size of the simultaneous transmissions and avoids the receivers to observe buffer overflows through flow control and lessens the network congestion through its self-throttling congestion control mechanism. TCP identifies losses through keeping track of Round Trip Time (RTT) of the previous transmissions and timeouts observed.

Performance of TCP over wireless links has been an active research area [21, 101, 31, 100]. These works focus on the intermittent connectivity issues frequent in wireless networks and propose techniques such as TCP flow differentiation through scheduling techniques [31, 101] and intermediate caching [21].

The transient nature of the wireless connectivity raises the issue of *spurious timeouts* which are defined as the timeouts that would not have occurred had the sender waited "long enough" [100]. Spurious timeouts trigger two undesirable responses [100, 90]: Firstly, TCP interprets the timeout as being caused by losses and unnecessarily retransmits segments. Secondly, the congestion avoidance mechanism gets falsely triggered causing the window size to decrease, yielding lower throughput than the capacity.

Even though there are some works elaborating on potential correlations of MAC layer retry limits and TCP behavior [90, 41, 55], the topic is largely untouched in the research community. One common intuition is that the retry limits should be assigned as high as possible in order to eliminate the effects of delay variation and spurious timeouts. However, this action might have consequences such as bad channel utilization (both on MAC layer because of high number of trials per packet and on transport layer because of increased timeout values), increased queue lengths and delayed detection of link breakage.

Intuitively, MORAL might affect the TCP behavior in two ways: Firstly, if retry limits that are higher or lower than the default value are assigned, TCP RTT values (thus timeout values) will reshape accrodingly. Secondly, the number of segments lost will increase for lower retry limits as the link layer packets will be attempted to be transmitted fewer times. We will address the latter issue first as it affects TCP more rigorously.

The second issue mentioned above arises because of the increased competitiveness of the environment by MORAL. As mentioned before, MORAL relies on

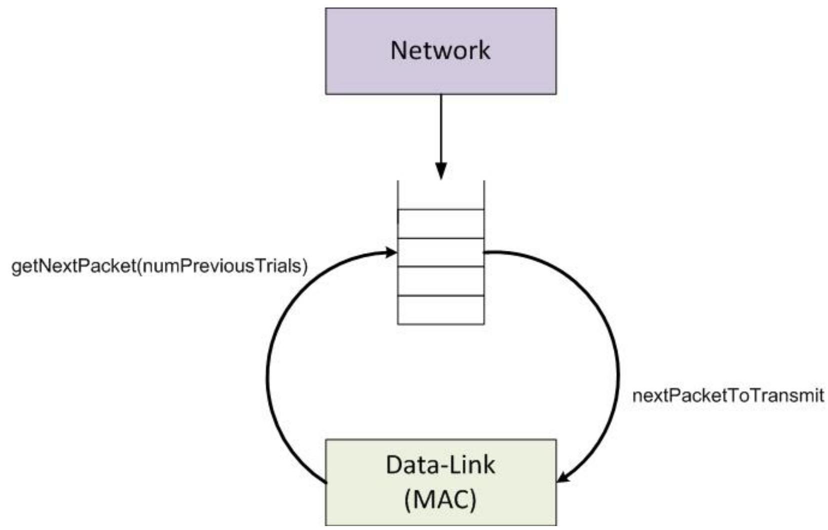


Figure 3.40: Alternative solution for improving the association between MORAL and TCP. Actual number of transmission attempts at the link-layer is kept higher than the default retry limit through an intermediate buffer control mechanism.

the packets losses to function properly. However, this feature makes the transport layer segments to be timed out rapidly if very low retry limit values are chosen. We propose a simple modification to alleviate the effects of this problem. Our mechanism extends the capabilities of the buffer between the MAC and the Network layer such that the number of times the previous packet was attempted to transmit was additionally transferred to the buffer control mechanism (Figure 3.40). Through a simple integer comparison, the control mechanism of the buffer can calculate how many times the current packet was attempted to transmit in total and discard the current packet only if this number exceeds the default retry limit. If the current packet has not been attempted to be transmitted at least the default retry limit number of times, the same packet will be returned to the MAC layer for further transmission attempts. This mechanism can be implemented in the scope of either of the adjacent layers, or as an intermediate pseudo-layer between the two layers. The details of such an implementation are beyond the scope of our work and left as a future exercise.

The other potential effect of MORAL on TCP (timeout variation) is due to the RTT deviation caused by retry limit assignments. The users with shorter retry limit assignments might observe shorter RTT values; and similarly, the users with longer retry limits might observe longer RTT values. Clearly, once a retry limit assignment is done, TCP timeouts will react to the changes and are expected to become steady afterwards. This is a complementary action as MORAL actually *modifies* the expected link behavior by creating differentiated environments. These implications of MORAL should be considered from the perspective of multi-rate networks. When a homogenous network becomes multi-rate, the effects on the RTT values observed are drastic as the fast users have to wait for the slow users to complete their transmissions. MORAL, on the other hand, lessens this behavior through assigning shorter retry limits (thus smaller wait times) to faster users. Hence, MORAL is expected to improve the TCP performance issues caused by increased RTT values in multi-rate networks.

MORAL's rule of thumb of keeping the relative probability of successful transmissions proportional (Equation 3.41) might cause minor variation in the retry limit assignments at different times. However, these adjustments are purely done to keep the throughput levels consistent and as long as the spurious timeouts are handled correctly (through a mechanism such as Figure 3.40), they are expected to contribute to the TCP RTT consistency.

In the light of the discussions above, we expect MORAL and TCP to function well together with minor modifications; and leave the validation of this claim as a future work.

3.7 Summary

We conclude our chapter by emphasizing the importance of link characteristics on the performance above the physical layer, specifically at the MAC layer. This chapter has shown that utilizing the physical layer information at the MAC layer in a cross-layer manner can enhance the performance immensely.

Conventional link adaptation techniques only consider adjusting the physical transmission properties according to different channel conditions. These techniques function as one-step mechanisms and expect the higher layers to operate perfectly with their decisions. Ignoring the potential aftereffects of these mechanisms can acutely impact the higher layer performance and can cause calamities such as the multi-rate anomaly. In this chapter, we have shown that relaying the encapsulated physical layer information to the MAC layer can lessen the effects caused by such calamities.

Our contributions in this chapter are twofold. Firstly, we extend a well-known analytical CSMA/CA model, for multi-rate networks and arbitrary retry limits per data rate groups; and validate our model for several different simulation scenarios. Secondly, we propose a new algorithm, MORAL, which passively collects information from its neighbors and re-aligns its MAC layer parameters according to the observed conditions. MORAL improves the fairness impaired by the multi-rate anomaly and promises increased total throughput through granting larger medium shares to faster users. Through extensive simulations, we have shown that MORAL outperforms the default static retry limit assignment scenario vastly.

MORAL is a link adaptation mechanism operating on the MAC layer, according to which the scope of a *link* is beyond simple interactions between two end-users and involves interactions between all of the neighbor users in the network. MORAL monitors the MAC-link and collects information continuously in order to create a fair, high throughput system.

MORAL is highly adaptable and totally distributed; furthermore, it does not require any modifications in the standards. These features make MORAL easily deployable in any real-life situations with minimal effort.

Chapter 4

Network Layer Enhancements for Multi-Rate Networks

Wireless links, as opposed to their wired counterparts, are much more dependent on good physical characteristics of the environment in order to operate properly. Successful delivery of transmission on wireless links heavily depends on the interference levels in the vicinity of the receiver, distance between the transmitter and the receiver, and other physical disturbances such as multipath fading [65]. Transient nature of these physical elements makes each link perform differently at different times, causing variation in the achieved performance [25, 149, 47]. In-motion transmissions [56] and in-door radio propagation [65] further complicate the transmission dynamics, making the performance of a link often unpredictable. For instance, Figure 4.1 shows the packet delivery ratio of a certain link in the Roofnet wireless mesh network [12] illustrating the fast changing behavior of a single link [1]. These properties make the widely-known *number of hops* metric [2] inadequate for distinguishing the high performance routing paths.

In this chapter, we propose a new routing mechanism, *Data Rate and Fragmentation Aware Ad-hoc Routing*, which considers two majorly used link adaptation parameters, the data rate and the fragmentation size (Section 2.2.1, Section 2.2.2 and the refereces therein) together in order to create energy efficient routing paths.

[1]Image taken from [91].

[2]For an excellent survey on routing technologies in the Internet, please see [73].

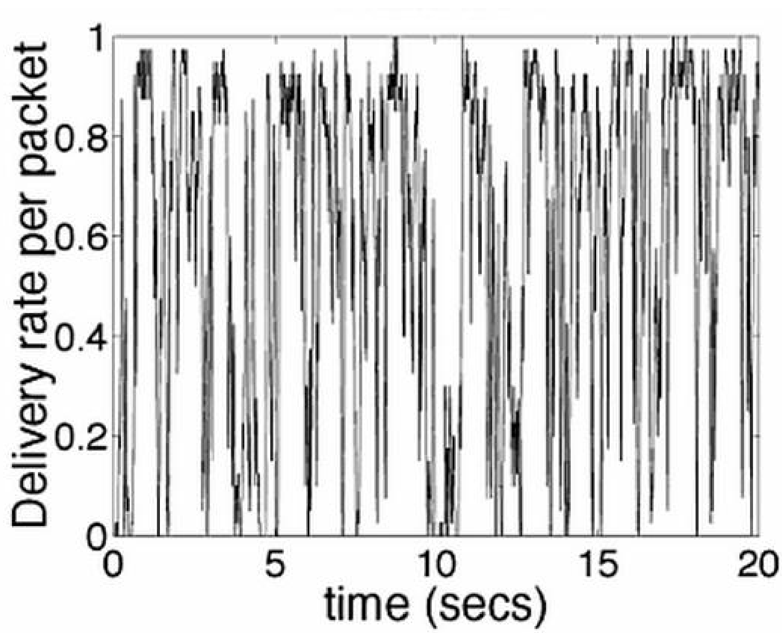


Figure 4.1: Short-term packet delivery rate variation of a single link in the Roofnet wireless mesh network [12], illustrating a highly variant behavior. Image taken from [91].

There have been several efforts in the literature which propose alternative routing metrics in order to estimate the link conditions during the path discovery process. However, these works either do not consider the link adaptation techniques used [46, 50, 133, 11, 49] or their analysis are based on rather simpler approximations (e.g. no retransmission concerns [18]). In contrast, our routing metric estimates the likelihood of a packet being successfully transmitted in a more accurate way taking not only the physical interference levels but also the parameter choices of the transmissions into account.

In addition, we also provide a novel link adaptation mechanism, *Combined Data Rate and Fragmentation Size Link Adaptation*, which calculates the best (data rate, fragmentation size) pair to use for given link conditions. Using our two mechanisms jointly, wireless nodes can adapt to physical conditions in a healthier manner and estimate the impact of this adaptation on the routing layer in a better fashion.

The rest of this chapter is organized as follows: We first describe the physical factors which affect the success rate of a transmission link. Then, we present a new energy-efficient routing metric for wireless networks which takes the physical characteristics of each link along a path into account during the path creation process. Next, we define a new link adaptation mechanism which adjusts both fragmentation size and data rate parameters according to the physical link characteristics. Finally, we report on our experimental results.

4.1 Characteristics of Wireless Communication

There are two main factors that might lead to an erroneous packet reception: ambient noise at the receiver side and received signal strength of the packet received. First factor is a property of the environment the node is in and its quantity cannot be reduced through adjusting communication link properties. The second factor depends on the physical characteristics of the path the packet is following and the transmission power level of the sender. Even though there are some mechanisms which adjust the transmission power level at the sender as mentioned in the previous sections; since most current wireless cards do not provide any mechanisms for adaptively choosing the transmission power level for each packet [106], we do not consider this parameter in our studies and focus on other transmission related parameters.

Although the quantities of the two factors defined above cannot be reduced externally, their effects on a communication link depend on the characteristics (i.e. parameter choices) of the corresponding transmission link. For instance, high complexity modulation techniques are more vulnerable to bad environmental conditions since they are utilized to send maximum amount of data under ideal conditions and do not possess enough redundancy to recreate the original data in case of errors. Similarly, longer packets are more susceptible to errors since the expected number of errors increase per number of bits transmitted.

The error of a given link can be portrayed through bit error rate analysis. Bit Error Rate (BER) is defined as the number of erroneous bits received for the total number of bits sent [33] for a time interval. Hence, for a given time interval,

$$BER = \frac{[Number\ of\ Erroneous\ Bits]}{[Total\ Number\ of\ Bits]}. \tag{4.1}$$

In a contention-free channel, the probability that a single bit will be received by the dedicated destination without any error, will solely depend on BER. It is expected that as the receiver's environment becomes noisier, BER increases. Error rate of a single packet depends on the size of the packet and the BER in the environment.

The calculation of BER depends on the modulation and coding techniques employed together with the physical characteristics of the medium in use. For example, for BPSK (Binary Phase-Shift Keying) and QPSK (Quadrature Phase-Shift Keying) [3], if stationary network nodes and independent bit error rates among multiple bits are assumed, one way of computing the bit error rate can be described as follows [119]:

$$BER = 0.5 \times erfc(\sqrt{\frac{P_r \times W}{N \times f}}) \tag{4.2}$$

where P_r is the received signal strength, W is the channel bandwidth in Hz, N is the noise signal power and f is the transmission bit rate. $erfc$ is the complementary error function and defined as $erfc(z) = \frac{2}{\sqrt{\pi}} \int_z^\infty e^{-t^2} dt$ [148]. Assuming free space propagation model for lower distances and two-way reflection model for larger distances, the received signal strength, P_r is calculated as follows [126]:

[3]IEEE 802.11b uses BPSK for 1 Mb/s transmissions and QPSK for 2, 5.5 and 11 Mb/s transmissions as shown in Table 2.1.

$$P_r = \begin{cases} \frac{P_t \times G_t \times G_r \times \lambda^2}{(4\times\pi)^2 \times D^2 \times L} & D \le D_{cross} \\ \frac{P_t \times G_t \times G_r \times h_t^2 \times h_r^2}{D^4 \times L} & D > D_{cross} \end{cases}$$

where P_t is the transmitted power, G_t is the transmitter gain, G_r is the receiver gain, h_r is the receiver height, h_t is transmitter height, L is the system loss factor and λ is the wavelength in meters. Also, D_{cross} can be calculated as

$$D_{cross} = \frac{4\times\pi\times h_r \times h_t}{\lambda}.$$

BER calculated above gives the probability that a single bit is received erroneously and can be used easily to calculate the probability of a packet received in error. Latter probability is known as Packet Error Rate (PER). Assuming that the probabilities of different two bits arriving erroneously are independent [4], for a packet with length l, PER can be calculated as

$$PER = 1 - (1 - BER)^l. \tag{4.3}$$

In one of the previous studies [106], abovementioned equations are used in order to find an optimal fragment size for packets using IEEE 802.11 fragmentation mechanisms as described in [5]. The authors also use this information for finding energy-efficient routes by minimizing the number of retransmissions. We also follow a similar approach and use these formulas in order to find optimal data rates and fragmentation sizes for IEEE 802.11b links, extending their work by enabling a second parameter dimension.

[4] This assumption can be seen in several works in the literature as in [44, 103, 88, 106].

In the next section, we explain how the physical characteristics of each link on a network path contribute to the performance of the overall network routing algorithm, and propose our new metric which takes these factors into account during the path creation process for better energy efficiency.

4.2 Data Rate and Fragmentation Aware Ad-hoc Routing

4.2.1 Network Layer Link Adaptation on IEEE 802.11b

Our network layer link adaptation schema is an extension of Nadeem et al.'s previous work [106]. In this work, authors find optimal fragmentation sizes to use per link according to the Signal-to-Noise ratios (SNR) seen and exploit this information in order to find energy efficient routing paths using AODV [114, 115] routing strategy. Their main intent is to reduce the power consumption through minimizing the number of retransmissions. Their philosophy is explained in the following paragraphs.

Let p_l represent the bit error rate over link l, and k_l represent the packet size in bits for the same link. It is straightforward to see that the number of transmissions needed (including the retransmissions and the actual packet) is $1/(1-p_l)^{k_l}$. Hence, the mean energy cost, C_l, for a successful packet transfer is

$$C_l = \frac{E_l}{(1-p_l)^{k_l}}$$

where E_l is the energy consumed by the sender and constant per bit, and p_l can be estimated using the SNR readings of the wireless card. In case of fragmentation, two types of overhead bits should be considered: extra bits (o_1) that have to be repeated per fragment and not considered as the data bits, such as PLCP preamble and header; and extra bits (o_2) that are transmitted within the fragment such as frame header and CRC field. Thus, if fragmentation size is k_l, then the total associated cost with a successful transmission of a packet can be denoted as:

$$C_l = \frac{L}{k_l - o_2} \times \frac{(o_1 + k_l) \times v}{(1 - p_l)^{k_l}}$$

where L is the original packet size and v is the transmission bit energy. [106] uses this metric as the cost function for their routing schema and, finds optimal fragment sizes per link and associated costs per path.

Above formula represents the total energy consumed in order to successfully transmit a packet, but ignores different physical modulation and coding selection among candidate data rates. Hence, we modify this formula for enabling different physical data rate selection.

Choosing different data rates results in two changes: 1) Bit error rate increases as the modulation technique gets more complicated, i.e. data rate increases; 2) As the power that can be generated by a radio chip is constant, energy per bit decreases as data rate increases. Hence, we end up with the following formula:

$$C_{l,f} = \frac{L}{k_l - o_2} \times \frac{(o_1 + k_l)}{(1 - p_{l,f})^{k_l}} \times \frac{v}{f_n}. \tag{4.4}$$

Here fn represents the normalized data rate selected, i.e. $f_n = \frac{f}{f_{base}}$, $f_{base} = 1$ Mbps . $p_{l,f}$ is the bit error rate of link l with data rate f using Equation 4.2.

Ideally, if the SNR level of a link is known, the BER and PER values for the same link can be calculated using the Equation 4.1 and Equation 4.3 respectively; and can be used in Equation 4.4 in order to calculate the expected energy cost of the associated link. The cost value calculation is orthogonal to the other network layer choices and can be integrated with any routing mechanism. In our proposed mechanism, we chose the well-known AODV algorithm because of its popularity and *on-demand* nature.

4.2.2 AODV and its Proposed Modifications

Wireless routing protocols are categorized as either *table-driven* or *on-demand* according to the techniques they follow for maintaining the routing state information [128]. In table driven protocols, users maintain consistent, up-to-date routing information from each node to every other node in the network. In these protocols, changes are propagated throughout the whole network in order to maintain a consistent network view.

The on-demand routing protocols, on the other hand, only create routes when desired by a source node. When a node requires a route to a destination, it initiates a route discovery process in order to establish a connection with the intended destination. Once the connection is established, it is kept alive as long as it is desired. When the route is no longer in use, it ceases to exist and the route discovery phase has to be repeated for future communication requests between the same source-destination pair.

As the table-driven mechanisms are always up-to-date, there is no path discovery mechanism needed for new connections and the source nodes can immediately start the communication with the intended destinations. However, this practical aspect comes with a large cost of frequent route maintenance and update messages transferred between the nodes. Hence, in networks where the traffic volume is expected to be large, table-driven mechanisms might be a better choice over on-demand routing mechanisms.

For our system, we have chosen the popular *Ad-hoc On-Demand Distance Vector* (AODV) protocol because of the following reasons: 1) as the physical characteristics of a link are transient and can change very rapidly as the physical environment changes, the cost calculation presented in Equation 4.4 has to rely on the recent channel conditions in order to represent the channel quality as close as possible. The on-demand nature forces the routes to be constructed just before the actual communication starts and the routes discovered reflect the most recent view of the network. 2) AODV protocol's route discovery mechanism is based on the sender's initiation and each node receiving the RREQ packets can estimate the quality of reception at the current position. This allows the channel quality to be estimated on the receivers' side for each link through the entire route. In contrast, the table-driven mechanisms rely on the periodic *hello* packets transmitted per node which are registered as potential route destinations for the routes to be constructed.

The following paragraphs summarize the AODV routing protocol in more detail. In [115], authors summarize their routing schema as follows:

AODV builds routes using a route request / route reply query cycle. When a source node desires a route to a destination for which it does not already have a route, it broadcasts a route request (RREQ) packet across the network. Nodes receiving this packet update their information for the source node and set up backwards pointers to the source node in the route tables. In addition to the source node's IP address, current sequence number, and broadcast ID, the RREQ also contains the most recent sequence number for the destination of which the source node is aware. A node receiving the RREQ may send a route reply (RREP) if it is either the destination or if it has a route to the destination with corresponding sequence number greater than or equal to that contained in the RREQ. If this is the case, it unicasts a RREP back to the source. Otherwise, it rebroadcasts the RREQ. Nodes keep track of the RREQ's source IP address and broadcast ID. If they receive a RREQ which they have already processed, they discard the RREQ and do not forward it.

As the RREP propagates back to the source, nodes set up forward pointers to the destination. Once the source node receives the RREP, it may begin to forward data packets to the destination. If the source later receives a RREP containing a greater sequence number or contains the same sequence number with a smaller hopcount, it may update its routing information for that destination and begin using the better route.

As long as the route remains active, it will continue to be maintained. A route is considered active as long as there are data packets periodically traveling from the source to the destination along that path. Once the source stops sending data packets, the links will time out and eventually be deleted from the intermediate node routing tables. If a link break occurs while the route is active, the node upstream of the break propagates a route error (RERR) message to the source node to inform it of the now unreachable destination(s). After receiving the RERR, if the source node still desires the route, it can reinitiate route discovery.

Our proposed mechanism requires the following modifications on the original AODV mechanism:

1. RREQ messages are modified so that the cost calculated using Equation 4.4 is included for the partial path traversed from the source node to the current node. Similarly, the cost of the path to the destination node from the current node is included in the RREP messages. In addition, RREP packets contain the (data rate, fragmentation size) pair to use along the particular link, as will be explained in the next sections.

2. The routing table structure is modified such that for each potential destination node, the cost of the associated link is also included.

3. The route discovery mechanism of the original AODV mechanism is largely unchanged, but it is modified so that it includes the cost metric instead of the hop metric during the route discovery phase.

4.2.3 Combined Data Rate and Fragmentation Size Link Adaptation

In the previous section, we have shown that the choice of the data rate and fragmentation size to use influences the energy efficiency of a single link. Furthermore, we have developed a routing cost function which discovers the network routes according to the expected energy expenditure considering the choices on these two parameters. Notice that the routing mechanism we propose is independent than the physical link adaptation techniques in use and only assumes that the wireless nodes are armed with some basic functionality of transferring the (data rate, fragmentation size) information in use to the network layer. In this section, we extend our system further and propose a physical link adaptation mechanism which adapts these parameters automatically during the path discovery process.

Our *Combined Data Rate and Fragmentation Size Link Adaptation* method is a natural extension of the routing mechanism we propose and extends the functionalities of each node such that during the route discovery process, in addition to the cost metric provided, the best (data rate, fragmentation size) pair to use for the given link is calculated and recorded as well. Consequently, this information is added to the RREP messages flowing back to the source node. Hence, each transmitter node along a route knows what data rate and fragmentation size to use for the next hop. Our method utilizes the SNR of each RREQ packet received in order to compute this information [5].

In the following paragraphs, we first analyze the behavior of the Equation 4.4 for varying SNR, data rate and fragmentation sizes; and afterward, we describe our physical link adaptation mechanism.

We simulated the equations above in Matlab in order to investigate performance differences caused by these choices. We used SNR as a parameter in our

[5]SNR is known to be a good estimator for the channel quality [37, 84].

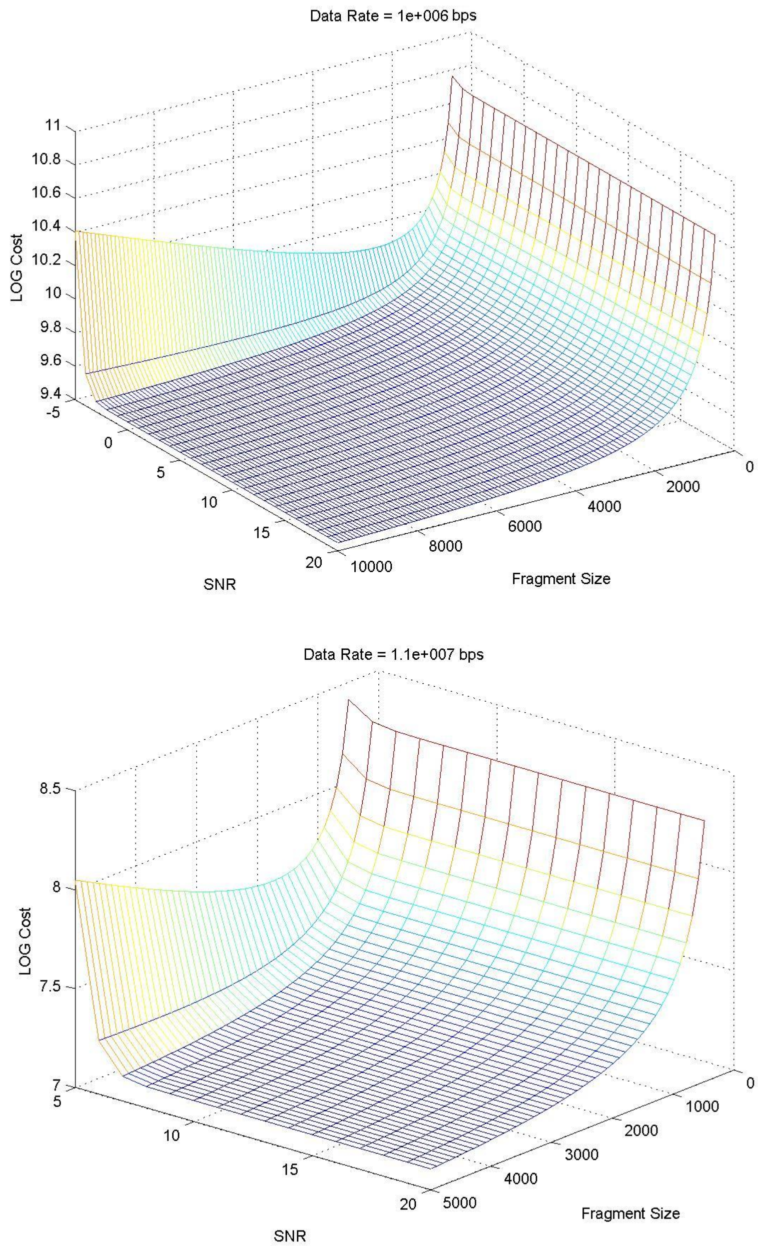


Figure 4.2: Impact of the data rate and fragmentation size choices on the energy efficiency cost for varying SNR levels. Lower data rates can provide better energy efficiency for a wider range of environmental conditions. Choosing too low or too high fragmentation sizes affect the energy efficiency negatively.

experiments and used the following approximation:

$$SNR = 10 \times log(\frac{P_r}{N})$$

and assigned channel bandwidth as $W = 22 \times 10^6 Hz$.

Figure 4.2 portrays SNR and fragment size effect on the cost function outcome for different data rate choices . These results are in accordance with the experiments conducted in [106]. Generally, the system is not very sensitive to SNR changes for high values of SNR, as the system is already functioning well. However, as can be seen from the figures, after a certain threshold, SNR starts to affect the overall cost intensely. As expected, for greater data rates, this threshold value is larger, implying greater sensitivity to noise.

Fragment size, on the other hand, influences the cost function significantly as well. But, unlike SNR, there is a local minimum point for the fragment size; and choices that are larger or smaller than the minima point result in suboptimal outcomes. Thus, it is clear that the fragment size and data rate should be carefully selected and adjusted according to link conditions.

Even though the previous analysis gives us insight about the effects of transmission parameters selection on the overall performance, it is still not clear how to choose our parameters for given link conditions. Hence, we conducted a second set of experiments in order to find the optimal fragment size and data rate for given SNR values. We looked at all of the data rate and fragment size possibilities and computed the combinations that yield the minimal cost per SNR. Since we have discrete number of possibilities (4 for data rate, $[o_2, 12000]$ and integer for fragment

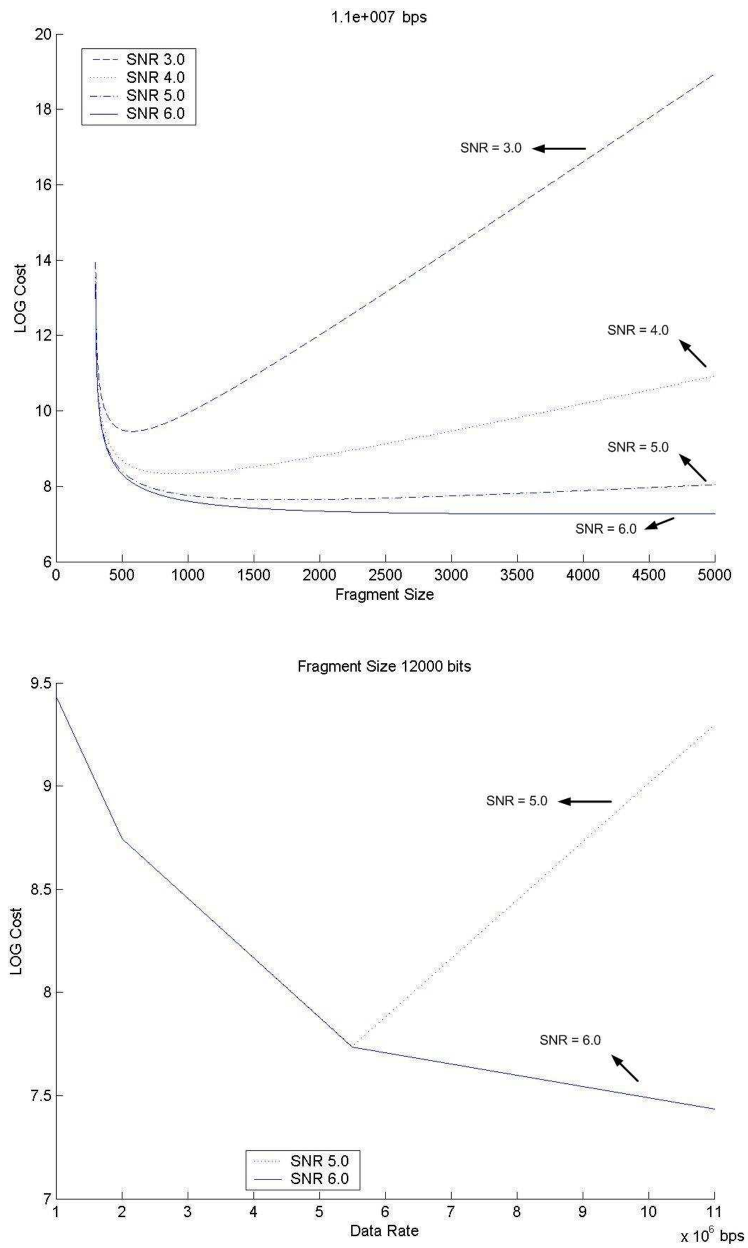


Figure 4.3: Impact of the data rate and fragmentation size choices on the energy efficiency cost for varying SNR levels. As SNR levels decrease, optimal data rate and fragmentation size shift to smaller values.

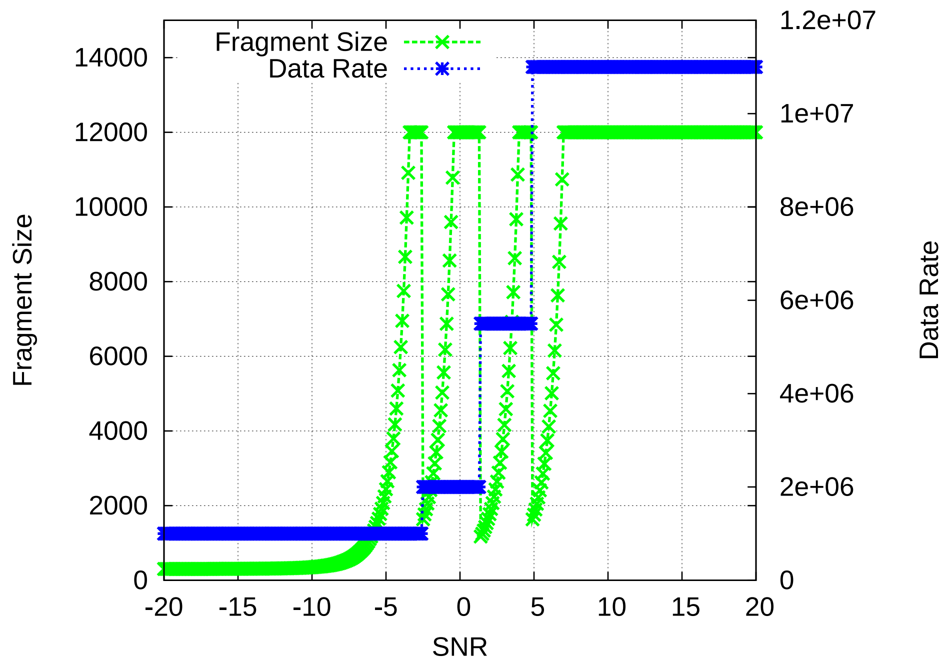


Figure 4.4: Optimal fragment size and data rate to use for a given SNR value for IEEE 802.11b.

size), our experiment was plausible.

We present our result in Figure 4.3. Bottom figure shows the effects of different data rate choices if fragmentation size is kept constant. As can be seen, for good SNR values, using the highest data rate possible gives the lower cost results, but as the SNR value decreases, highest data rates become cost inefficient and switching to a lower data rate yields lesser costs. In this figure, for $SNR = 5$, optimal data rate is 5.5 Mbps whereas for $SNR = 6$, optimal data rate is 11 Mbps. A similar observation can be made with the top figure regarding the fragmentation size. As the SNR readings get worse, optimal cost results can be reached with lower fragmentation sizes.

Figure 4.4 presents the optimal fragment sizes and data rates per SNR readings we calculated through the analysis above. Using this figure, the users can choose the best (data rate, fragmentation size) pair to use according to the environment conditions they are in. For instance, for $SNR = 0$ dB, the best data rate to use is 2 Mb/s and the best fragmentation size to use is 12000 bits. This figure summarizes that as the link conditions worsen (thus lower SNR values are being observed), it is reasonable to decrease the fragment size first, keeping the current data rate unchanged. However, after some critical point, decreasing the fragment size is not sufficient and it is better to switch to a lower data rate and start over with the maximum fragment size. As the link condition further worsens and shortening the fragments is not sufficient, it is best to drop the data rates even lower, setting the fragment size to maximum after each data rate decrease.

Using previous results, we created a table which maps SNR values to optimal data rate and fragmentation size choices. Then, we used these results in order to find the costs associated with each link and applied this cost as our routing function to find energy efficient routes.

4.3 Experiments

We constructed a 5x5 mesh network where the sender is located at the left topmost position and the receiver is at the right bottommost position. We assumed that only adjacent nodes (excluding the diagonal ones) can communicate with each other. Clearly, in this architecture, assuming that all the nodes are static and active

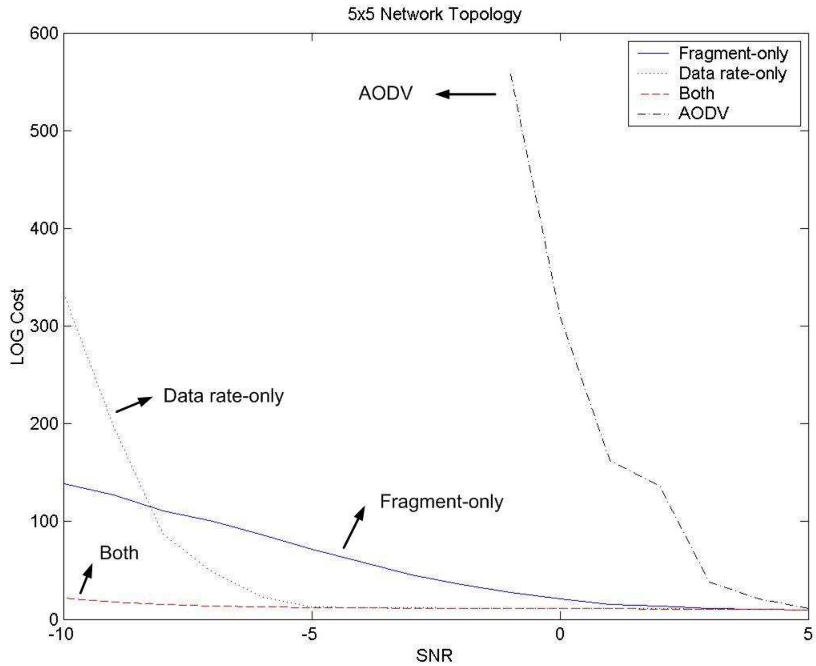


Figure 4.5: Energy efficiency achieved for different link adaptation schemas in a 5x5 network. Data Rate and Fragmentation Aware routing protocol outperforms the original AODV protocol immensely.

all the time, minimum number of links that need to be traversed is 8. Hence, pure AODV, omitting any SNR properties of the system, chooses one of many available 8-hop links in the network.

In addition to Pure-AODV with the classical number of hops metric, we looked at three different link-adaptation mechanisms: 1) only fragmentation is used as an adjustable parameter and data rate is kept constant at maximum 11 Mbps, 2) only data rate is adjusted and fragmentation size is kept constant at 1500 bytes, and 3) both of these parameters are adjusted.

In this experiment, we generated normally distributed SNR values for each link independently and vary the mean of the distribution function in order to create arbitrarily erroneous environments. Figure 4.5 exhibits our results, in which the x-axis represents the mean of the normal distribution. Pure-AODV, omitting any environmental characteristics in its route-finding schema, performs very poorly as the SNR values worsen. Since they deploy some kind of a link-adaptation schema, all of the other mechanisms perform much better than pure-AODV.

The mechanisms stated above have different sensitivity levels to low SNR degrees. We observe that changing the data rate parameter has a more radical effect on cost, compared to the fragment size parameter. This observation supports our initial analysis as well. As expected, when we use both of the adjustable parameters together, the performance of the system is much better compared to when only single parameter is used. This effect gets much clearer as SNR decreases.

Notice that the data rate only mechanism usually produces better performance levels than the fragment only mechanism. However, for extremely low SNR values (less than -7 dB in Figure 4.5), data rate only mechanism reaches the limits of the lowest data rate (1 Mb/s in this example). Figure 4.2, top graph, also illustrates that for very bad channel characteristics, the cost associated with 1 Mb/s data rate choice increases rapidly if maximum fragment size is chosen. Fragment only mechanism, on the other hand, can adapt to the channel in a better fashion, through deploying extremely small fragment sizes.

4.4 Summary

We conclude our chapter by emphasizing the importance of link characteristics on the performance above the physical layer, specifically at the network layer. This chapter has shown that utilizing the physical layer information at the network layer in a cross-layer manner can enhance the performance immensely.

Our contributions in this chapter are twofold: We have first described a novel routing methodology, *Data Rate and Fragmentation Aware Ad-hoc Routing* protocol which considers the physical characteristics of each link during the path discovery process and produces energy-efficient network paths. Secondly, we have proposed a new physical link adaptation mechanism, *Combined Data Rate and Fragmentation Size Link Adaptation* which adjusts two physical layer parameters (data rate and fragmentation size) together in order to better adapt to changing environmental conditions. We have shown through simulations that using both of our proposed mechanisms together improves the routing performance significantly over the classical AODV routing algorithm with the number of hops metric.

Data Rate and Fragmentation Aware Ad-hoc Routing protocol is a link adaptation mechanism operating on the network layer, according to which the scope of a *link* is beyond simple interactions between two end-users and involves interactions between all of the users in the network contributing to the path created between the source and the destination. Our protocol utilizes the physical layer information in a cross-layer manner and adapts to the Network-link through creating energy-efficient paths.

Our methods proposed require minimal changes in the original AODV protocol and can be used in conjunction with any other on-demand protocol of choice.

Chapter 5

Conclusions and Future Directions

Wireless networks pose additional challenges compared to their wired counterparts. Quality of a wireless communication link vastly depends on the physical environmental characteristics and the behavior of the fellow wireless users. Conventional wireless standards do not differentiate between different network elements according to their competence levels and, generally, assume that each wireless user and each wireless link are identical in their capabilities.

The work presented in this dissertation has demonstrated the advantages of utilizing the physical transmission choices in a cross-layer manner beyond the scope of the physical adjustments. Such information transfer enables each layer to better adapt to channel conditions and distinguish the capabilities of different network elements in a better fashion.

5.1 Summary of Contributions

Wireless networks, today, are armed with *Link Adaptation Techniques* which adaptively choose the best suitable transmission parameters according to the observed link characteristics. These techniques are essential for the overall performance of the networks, especially for environments where ambient noise level is high and/or noise level changes rapidly.

Link adaptation techniques are by definition one-step mechanisms. They answer the questions of *What to change?* and *When to change?* in order to improve the present layer performance. Once these decisions are made, other layers are expected to function perfectly with the new communication channel conditions. In this dissertation, we have shown that this assumption does not always hold; and provide two mechanisms that lessen the negative outcomes caused by these decisions.

The proposed mechanisms augment the current link adaptation techniques for higher layers, extending the notion of the *link* in *link adaptation*. At the physical layer, a link represents the physical interaction between a transmitter and a receiver. At the MAC layer, a link represents not only the interaction between two users, but also the neighboring users in the vicinity of the two users. Finally, at the network layer, a link represents all of the physical links traversed along an end-to-end transmission, i.e. a path. Our dissertation consists of two link adaptation mechanisms designed for MAC and network layers which adapt to the new link conditions caused by the physical layer adjustments.

Our first mechanism, MORAL, is a MAC layer link adaptation technique which utilizes the physical transmission information in order to create differentiation between wireless users with different transmission capabilities. MORAL passively collects information from its neighbors and re-aligns the MAC layer parameters according to the observed conditions. MORAL improves the fairness and total throughput of the system through distributing the mutually shared network assets to the wireless users in a fairer manner, according to their capabilities.

Our second mechanism, Data Rate and Fragmentation Aware Ad-hoc Routing protocol, is a network layer link adaptation technique which utilizes the physical transmission information in order to differentiate the wireless links according to their transmission capabilities. The proposed mechanism takes the physical transmission parameters into account during the path creation process and produces energy-efficient network paths. Our protocol improves the energy efficiency of the system through avoiding the power consumptive links, otherwise not considered as a criterion.

The mechanisms that have been described in this dissertation are highly adaptable and totally distributed. Furthermore, they do not require any modifications in the current standards and can be used on the currently available devices without any hardware changes.

5.2 Future Directions

There is still much to do within the framework of cross-layer link adaptation mechanisms towards construction of fully link adaptable wireless networks. Our work presented in this dissertation has demonstrated two examples in the direction of full-stack link adaptation mechanisms from the perspective of MAC and network layers; leaving the other layer interactions for future work.

The transport and physical layer interaction is one of the foremost research topics that we have left for future investigations. Sudden physical layer adjustments performed by the link adaptation mechanisms might cause several different types of consequences at the transport layer, such as lost packets (e.g. caused by channel probing through higher data rate packets [82]) or varying round trip times (e.g. caused by switching to different data rate choices). Transport layer mechanisms can benefit from the cross-layer interactions with the physical layer, through fine-tuning their internal decision mechanism to utilize the physical layer information, if conveyed in a timely manner.

The application layer can also benefit from the physical layer information through estimating the network conditions in a better fashion. Higher layer applications can consider modifying their end-to-end requirements [66], if they perceive the upcoming transmission changes in advance. For instance, multimedia applications that adapt their video streams according to the network status [151, 64] can gain better insight about the channel conditions and revise their video quality related parameters in a healthier way.

Finally, integration of the two proposed mechanisms is also left for future investigations. Our intuition is that our MAC and network layer techniques can be better integrated with additional cross-layer interaction. To begin with, MORAL can be extended to consider additional transmission related parameters in its decision mechanism, such as the fragmentation size and power level. In addition, our network layer protocol can be modified such that its energy-efficiency metric takes the adjustments made by MORAL into account. We conceive these future work items as our next step towards our vision of full-stack link adaptation mechanisms.

5.3 Epilogue

The research demonstrated in this dissertation contributes to our understanding of link adaptation techniques and broadens the scope of such techniques beyond simple, one-step physical adjustments. Such a broader perspective requires better interaction between the physical layer and the layers above, in order to re-tune the optimization algorithms utilized within each layer. We have designed and implemented two cross-layer mechanisms that utilize the physical layer information to better adapt to the varying channel conditions caused by physical link adaptation mechanisms. These mechanisms has shown that even though *Link Adaptation* concept starts at the physical layer, its effects are by no means restricted to this layer; and the wireless networks can benefit considerably by expanding the scope of this concept throughout the entire network stack.

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