HIGH PERFORMANCE RATE ADAPTATION ON IEEE 802.11 NETWORKS

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VITA

Shaoen Wu, son of Daju Wu and Xiuying Tu, was born on July 18, 1976, in Sichuan, China. He attended Qingdao University of Science and Technology in Qingdao, China, and graduated with a Bachelor of Science degree in Automation and System Control, 1998. He immediately entered into graduate school at University of Electronic Science and Technology of China and received a Master of Science degree in Automation Theory and Engineering in March, 2001. He was then hired by Bell Laboratories, Lucent Technologies Inc. (China), where he worked as a Member of Technical Staff I. He married Lingyan Wang, daughter of Daji Wang and Fengzhen Hu, on Jan 1, 2002. In Spring of 2005, he was admitted into the Ph.D. program in Computer Science and Software Engineering at Auburn University. In July, 2008, he and his wife Lingyan became the proud parents of their son David (Si-Xing) Wu.

DISSERTATION ABSTRACT

HIGH PERFORMANCE RATE ADAPTATION ON IEEE 802.11 NETWORKS

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IEEE 802.11 standard has evolved from the basic transmission rates in early days to multiple rates today with advanced encoding and antenna techniques. The performance of IEEE 802.11 wireless networks is supposed to benefit from the support of multiple transmission rates. To take advantage of multiple available rates, a strategy is required to choose the most appropriate rate in transmission: rate adaptation in wireless networks is the selection of the optimal transmission rate for data frames under current channel conditions. For this purpose, a rate adaptation scheme must assess the channel condition and then accordingly adjust the data rate if necessary. In this dissertation, we extensively survey the rate adaptation schemes in literature. We progressively proposed three strategies to address some open problems in rate adaptation. First, we exploit the periodical mandatory beacon frames in IEEE 802.11 networks to estimate the initial rate for a stream of data frames and yield the scheme Beacon Assisted Rate Adaptation (BARA). In IEEE 802.11 networks, particularly with basic CSMA/CA access, the assessment of channel condition may become complex because frame losses likely result from channel condition degradation or transmission collision, and each type of frame loss requires a different response in rate adjustment. To diagnose the cause of a frame loss, we propose the basic rate retransmission technique and a rate adaptation scheme Loss

Differentiated Rate Adaptation (LDRA), based on Signal-Noise-Ratio (SNR). However, SNR and Received Signal Strength Indication (RSSI) are deemed not to be good channel condition indicators. Then we investigate rate adaptation schemes based on frame losses with loss differentiation ability. We discuss some anomalies observed in implementing a Linux based rate adaptation testbed. Based on the observations, we propose Effective Rate Adaptation (ERA) to effectively adapt rates in mixed frame lossy environments. ERA judiciously exploits the IEEE 802.11 fragmentation mechanism to accurately diagnose the cause of a loss and responds accordingly. We analytically prove that the fragmentation mechanism incurs less overhead and is more effective than using RTS/CTS at the basic (lowest) rate for rate adaptation in a collision dominated environment. ERA also takes effective actions to overcome abnormalities observed in other rate adaptation schemes. For performance evaluation, we first simulate all our three schemes on ns-2. Moreover, we implement ERA and four selected most recent representative rate adaptation schemes on a Linux based testbed and evaluate them with extensive experiments in both controlled and public field tests. Experiment results demonstrate that our proposed schemes outperform their peers in most scenarios.

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TABLE OF CONTENTS

LIST OF FIGURES

INTRODUCTION

The "last mile" access networks are undergoing tremendous migration from wired technologies to wireless. Among these wireless access technologies, IEEE 802.11 [1] has gained wide popularity. Data transmission on wireless medium suffers frame losses from unstable channel conditions, which fluctuate because of signal degradation (like fading, interference, fast time variation). When a channel can support high data rates, transmission at lower rates definitely underutilizes the scarce wireless resource. However, transmission at an overoptimistic data rate results in more frame losses. In IEEE 802.11 networks, frame loss is exacerbated by collision because multiple stations running with CSMA/CA protocol may transmit data simultaneously. A station can improve the probability of successfully delivering a frame in channel degradation with a more robust modulation scheme, which yields a lower data rate. But, it does not increase the chance in collision dominated environments. Instead, the larger transmission range of a lower data rate worsens collisions, and consequently degrades the overall network performance. Different causes of frame loss require different reaction. Therefore, we need a strategy to appropriately adjust data rates based on channel conditions. Rate adaptation is a strategy that determines the optimal rate most appropriate for the current wireless channel conditions. Rate adaptation generally consists of two functions: channel assessment and rate adjustment. Channel assessment estimates the channel condition or variation trends whereas rate adjustment determines the most appropriate rate based on the assessment.

Rate adaptation on IEEE 802.11 networks has been extensively studied in the past years and many schemes [2–13] have been proposed. These schemes can be generally categorized into two

generations. A first generation of these schemes consider adjusting data rate primarily due to channel degradation. They [2–7] assume that frame losses are mostly caused by channel fading because RTS/CTS control frames are *believed* to minimize or eliminate collisions. Some rate adaptation schemes [6, 7] even count on the exchange of RTS/CTS control frames running at the basic rate to estimate the practical rate for data frames. Therefore, most first generation rate adaptation schemes systematically respond to frame loss by decreasing the data rate. However, based on IEEE 802.11 standard and common practice, the use of RTS/CTS control frames is rare: because RTS/CTS are overhead, they are optional and are recommended only for large data frames. Without the presence of RTS/CTS control frames, RTS/CTS based rate adaptation schemes are not effective for deployment and also the collision is not so sparse to neglect. Even with RTS/CTS control frames, congestion losses may still occur and mislead these rate adaptation schemes. While decreasing rate is appropriate for channel degradation, it is not for collision for two reasons: 1) a lower rate may exacerbate medium congestion because of longer frame transmission duration and wider effective communication range (more interference and higher collision probability); 2) a lower rate unnecessarily underutilizes network resource because the channel is still able to support higher data rates. Therefore, if granted the ability to differentiate frame loss causes, a rate adaptation can tremendously benefit network performance.

Thus, recently, researchers proposed the second generation rate adaptation schemes [8, 9] to appropriately react to each type of loss: channel degradation or collision. Some schemes explicitly diagnose the cause of a frame loss [8, 9] while others implicitly adjust the data rate to maximize the throughput. Although they differ in methods to assess channel condition and to differentiate losses, they all agree that the transmission rate should be decreased only in case of channel degradation and rather remain constant for a collision.

Moreover, there is no effective strategy to estimate the initial data rate at the very beginning of a transmission when a wireless station just starts up or has been staying idle for a long time. In general, existing proposed protocols just simply suggest using the median rate in the supported rate set, or the basic (lowest) rate, or the rate used in the last successful transmission. However, these strategies do not depend on current channel conditions. Therefore, it is desirable to design an effective strategy to address this challenge.

Another challenge is the loss recovery (or retransmission) strategy after a frame loss. Since no algorithm can perfectly estimate the channel, a frame loss happens when the channel condition is overestimated. Traditional rate adaptation algorithms unduly retransmit the lost frame at the failing rate until the retransmission succeeds or the rate is lowered to the next level after a timer expires. Such a strategy does not take advantage of the information exchanged in the transmission and is not efficient.

The contributions of this dissertation are represented by three new rate adaptation schemes proposed along the way to identify an effective and efficient strategy. The key features of these contributions are:

- We address the estimation of the initial rate for a stream of data frames. We reach this goal with the scheme *Beacon Assisted Rate Adaptation* (**BARA**) that exploits the periodical mandatory *beacon* frames available in IEEE 802.11 networks.

- We tentatively propose the basic rate retransmission technique to differentiate frame losses in combinational loss environments and yield the scheme *Loss Differentiated Rate Adaptation* (**LDRA**).

- We present some anomalies observed among other recent rate adaptation schemes with loss differentiation ability. Also, we identify the inefficiency of loss differentiation in the basic

3

rate retransmission technique used in *LDRA*. Finally, we propose a complete strategy *Effective Rate Adaptation* (**ERA**), to accurately diagnose the cause of losses, effectively adapt rates and promptly recover losses by taking the advantage of the *fragmentation* mechanism in the IEEE 802.11 standard in channel assessment and loss differentiation. As shown later, *ERA* performs effectively and efficiently in *both* channel degradation and collision dominated environments. We provide a simple analytical proof that the fragmentation mechanism is more effective than RTS/CTS control frames in diagnosing the cause of a frame loss, adapting the data rate, and recovering losses.

- To evaluate *ERA* and other most recent, most representative rate adaptation schemes, we implement them on a Linux based testbed. We conduct extensive controlled and field experiments on the testbed under the same conditions for these rate adaptation schemes, in addition to the performance evaluation with numerous simulations.

From the experiment results gathered from both simulations and the implemented testbed, our solutions outperform their peers in most scenarios. Particularly, the outstanding performance of *ERA* benefits from: 1) its accurate and prompt diagnosis of the cause of a frame loss (channel degradation or congestion) and appropriate response; 2) its faster recovery of a frame loss. Whenever a frame is lost for the *first* time, *ERA* splits the lost frame in two fragments in retransmission: a very short fragment and the remainder. After the cause of a loss is analyzed, *ERA* maintains the rate unchanged for collisions and judiciously adapts the rate for channel degradation based on a halving-rate strategy described later.

The remaining sections of this chapter (Chapter 1) present the background information, the review and our classification of rate adaptation schemes in literature, the motivation followed by

the design principles and the rationale of those techniques in our schemes. In the rest of this dissertation, we will chronologically present each rate adaptation scheme (*BARA, LDRA* and *ERA* in order) and the corresponding performance evaluation as an individual chapter. Next, in Chapter 2, we present our first rate adaptation work *BARA* and its performance evaluation. Then, in Chapter 3 we discuss the rate adaptation scheme *LDRA* that can differentiate frame losses with the basic rate retransmission technique. Afterwards, Chapter 4 presents the design of a complete rate adaptation solution *ERA*, related implementation and performance evaluation. Finally, Chapter 5 concludes this dissertation and provides hints to future work.

1.1 Background

Research in wireless network can barely be carried out without investigating the characteristics of wireless channels. This section presents the literature investigation on IEEE 802.11 channels, followed by the review of related IEEE 802.11 standard information.

1.1.1 Investigation on IEEE 802.11 Network Channels

Existing IEEE 802.11 link measurements [15–24] can be categorized upon different criteria. This section groups them according to the environment in which these assessments are conducted. Most of the measurements on IEEE 802.11 network link were completed in outdoor environments. More extensive assessment is needed for indoor environments.

Aguayo *et al.* [18] performed extensive and nearly the most complete link-level measurement in a 802.11b mesh network named Roofnet [14] with hundreds of node pairs with antenna mounted at the top of different campus buildings. From their experiments in an urban-like network, the interesting observations are:

- Most of the links experience mild loss rates regardless of the communication distance between node pairs. Namely, the frame loss rate is not closely correlated to the communication distance.

- Although links with strong signals are likely to have low frame loss rates, signal strength is in general not predictive of the loss rate.

- The fact that most links experience mild loss rates is probably due to multi-path fading in outdoor environments.

- When the loss rate increases heavily due to collisions, high data rates might result in much better performance than low data rates. This implies that rate adaptation algorithms should *not* take the loss rate as the *only* indicator to adjust data rate.

Although Aguayo *et al.* carried out extensive experiments and analyzed in detail the factors impacting IEEE 802.11 networks, their results are limited to outdoor environments and IEEE 802.11b [25] only. As the physical coding is different between IEEE 802.11b and IEEE 802.11g [26], they might have different characteristics even in the same environment, which is partially demonstrated by Bianchi, Formisano, and Giustiniano [17].

Chebrolu, Raman, and Sen [23] monitored several links in a long distance 802.11b rural network, where interference is rare. Their observation on this network setting somehow contrasts with Aguayo's observations [18]. They found:

- The relationship between the packet loss rate and the signal-to-noise (SNR) is close to the theory. The loss rate varies in a very narrow band of SNR. This is confirmed by Barsochhi, Oligeri and Potort [19].

- Transmission rate (modulation) has a significant impact on the frame loss rate.

- External interference can almost destroy the communication in the link.

One possible explanation for the contrast between Chebrolu [23] and Aguayo [18] is the network environment. Although both measurements are gathered from 802.11b outdoor networks, one is with heavy communication congestion (collision), the other is not. In [23], most frame losses are caused by channel fading. Therefore, more robust modulation (lower data rate) incurs less frame loss.

Rodrig *et al.* [15] collected real time traces from the conference SIGCOMM 2004. From the data they gathered, almost 35% of whole transmission time is spent in retransmitting. And almost 28% of data frames had to be retransmitted. This observation implies the ineffectiveness of current rate adaptation algorithms implemented in commercial wireless adaptor drivers. This work also shows that the frame loss from CSMA/CA contention has a profound impact on the performance of the rate adaptation.

Bianchi, Oligeri and Potort [17] found that the link behavior of 802.11b is totally different from that of 802.11g due to the physical coding variance. Some signal environments favorable to 802.11b throughput are not for 802.11g's.

Souryal *et al.* [27] assessed the 802.11 link in an indoor environment. They claim that the SNR is a good indication of link robustness. But the trace is very limited and the IEEE 802.11 variant used in experiment is not specified.

One conclusion upon these assessments of IEEE 802.11 networks is that although SNR can be a good indicator to adapt rate in some cases particularly in outdoor wide space, frame loss from contention (collision) is worth more attention in rate adaptation.

1.1.2 Review of Relevant IEEE 802.11 Features

In this section, we firstly present the support of multiple data rates on IEEE 802.11 networks. Then we briefly introduce the basic CSMA/CA access mechanism in IEEE 802.11 Distributed Coordination Function. Finally, possible frame loss scenarios are discussed.

Support of Multiple Rates in IEEE 802.11

Since 1999, IEEE 802.11 standard has evolved through several variants. The support of multiple rates becomes a mandatory requirement in the physical layer with more and more advanced encoding and modulation techniques for all these variants. For example, IEEE 802.11g [26] offers 1, 2, 5.5, 11, 6, 9, 12, 18, 24, 36, 48, 54 Mbps; IEEE 802.11a supports 6, 9, 12, 18, 24, 36, 48, 54 Mbps; IEEE 802.11b consists of 1, 2, 5.5, and 11 Mbps. Most recently, IEEE 802.11n draft has recommended high data rates up to 540 Mbs with MIMO [28] technology. However, the IEEE 802.11 standard group does not mandate or recommend any specific rate adaptation strategy even though a rate adaptation strategy is critical to the performance in multiple-rate networks.

Basic CSMA/CA Access

In IEEE 802.11 standard, there are two access methods for Distributed Coordination Function. One is CSMA/CA with optional RTS/CTS. However, the basic CSMA/CA access is mandatory. With CSMA/CA, each station has to sense a channel before it transmits. To reduce collisions, the station does not transmit immediately if the channel is sensed idle, but backs off a contention window of time slots whose size is randomly picked up from a range. After the contention window time passes and if the channel is sensed idle, the station will transmit its data immediately. If the channel is still busy at the end of the contention window, the station picks up another random

contention window size from a doubled range and repeats the backoff procedure. The transmission occurs as a stop-and-wait mechanism: an ACK frame is the proof of successful receipt of the data frame at the receiver station. Although channel sensing and random backoff are introduced to reduce transmission collision, two transmissions are still likely to collide with each other, especially when a hidden terminal problem exists: a station receives two transmissions from two stations that can not sense each other. The collision in an environment with basic CSMA/CA access is exponentially increased with the number of transmitting stations.

Loss Scenarios

Unguided wireless medium is vulnerable to distance fading, physical characteristic change of the air, external signal interference and collision from simultaneous transmissions. These disturbances cause frequent frame losses in wireless communication. IEEE 802.11 networks are generally exposed to two kinds of loss environment: strictly channel degradation dominated or the combination of degradation and collisions. Most wireless home networks are channel degradation dominated environments, where there is only one wireless user (laptop or other IEEE 802.11 equipped device). In such an environment, whether the user is mobile or not, frame losses are mostly due to channel degradation (e.g. fading due to distance or fast channel variation). Another degradation scenario is the long distance mesh network backbone deployed in rural regions [29]. Since IEEE 802.11 supports multiple non-overlapping channels, these sparse wireless backbone nodes generally connect to one or two neighbors, each link on a different channel, thus eliminating transmission collisions. More frequently, IEEE 802.11 networks, such as corporate and campus networks, suffer from both channel variation and transmission collisions. Normally, there are multiple wireless client stations associated with an access point in these networks. It is inevitable that some transmissions collide.

To minimize data frame collisions, short RTS/CTS control frames are recommended to precede data frames. However, due to the overhead caused by RTS/CTS to data frames, IEEE 802.11 standard dissuades their use for those data frames with a length smaller than the *RTS threshold*. Garg and Kappes [30] showed that, for VoIP traffic with 160-byte packets, the data frame efficiency drops to about 12% in IEEE 802.11b networks at 11 Mbps when RTS/CTS control frames are used. Thus, RTS/CTS frames are rarely used in practice. This low utilization of RTS/CTS was observed by Rodrig *et al* [15]: RTS/CTS control frames account for only 297 frames compared to 5540 data frames (in Table 2 of [15]). Therefore, it is fairly reasonable to expect frequent frame losses from collision in the real world.

1.2 Literature Review

Rate adaptation in IEEE 802.11 networks has been studied for years. This section surveys the most typical and latest relevant schemes. We begin with the first generation rate adaptation schemes, which do not differentiate losses due to channel degradation from those due to collision. Then the second generation schemes with loss differentiation are reviewed.

1.2.1 First Generation: Rate Adaptation without Loss Differentiation

This section presents the rate adaptation schemes without loss differentiation. They are categorized in two groups: frame loss based and signal strength based.

Rate Adaptation Based on Frame Loss

Auto Rate Fallback (**ARF**) by Kamerman and Monteban [2] is the earliest rate adaptation scheme for IEEE 802.11 based wireless networks. Kamerman and Monteban proposed it for the

Lucent Wave-II wireless LAN adapters. It is simple and intuitive. A sender starts transmission at the basic (lowest available) data rate (2 Mpbs in IEEE 802.11b) and triggers a timer. If either the timer expires or the sender succeeds for N (a constant threshold) consecutive transmissions, the sender increases its data rate r_{old} to a new data rate r_{new}, and the timer is reset. If the first transmission at the new rate r_{new} fails *immediately* after the data rate is increased, the sender falls back to the prior rate r_{old}. The data rate is also decreased when the sender fails in transmission twice consecutively. *ARF* considers the frame loss as the indicator of channel conditions. It adjusts the rate based on the number of consecutive successful transmissions.

Since *ARF* [2] the earliest rate adaptation for WLAN, several rate adaptation schemes have been proposed to refine it. One of them is the *Adaptive Auto Rate Fallback* (**AARF**) [3] proposed by Lacage *et al.*. *ARF* suffers from periodical rate fluctuation between the best rate it can support and the higher rate even in fairly stable channel condition due to its *constant* rate increase threshold. To tackle with this shortcoming, *AARF* introduces an *adaptive* rate increase threshold N. Just as *ARF* does, *AARF* records the number (M) of consecutive successfully transmitted frames. If M is no less than the threshold N, the transmission rate is increased from r_{old} to a new rate $r_{new}$with the assumption that the channel is likely stable enough to support a higher data rate. However, *AARF* differs from *ARF* in that it adaptively adjusts this threshold N after a frame loss at a newly increased rate. More particularly, consider that the very first transmission at this new rate r_{new} fails, then the sender falls back on the prior rate r_{old} immediately (with assumption that the channel is not actually good enough to support rate r_{new}). At the same time, *AARF doubles* the threshold to $2N$ for the next cycle to in crease rate unless the threshold reaches the upper bound (50). Otherwise, i.e. the first transmission at the new rate succeeds, the threshold is reset to the lower bound (10). This threshold is also reset and the rate is decreased if a transmission fails twice consecutively. The benefit of such
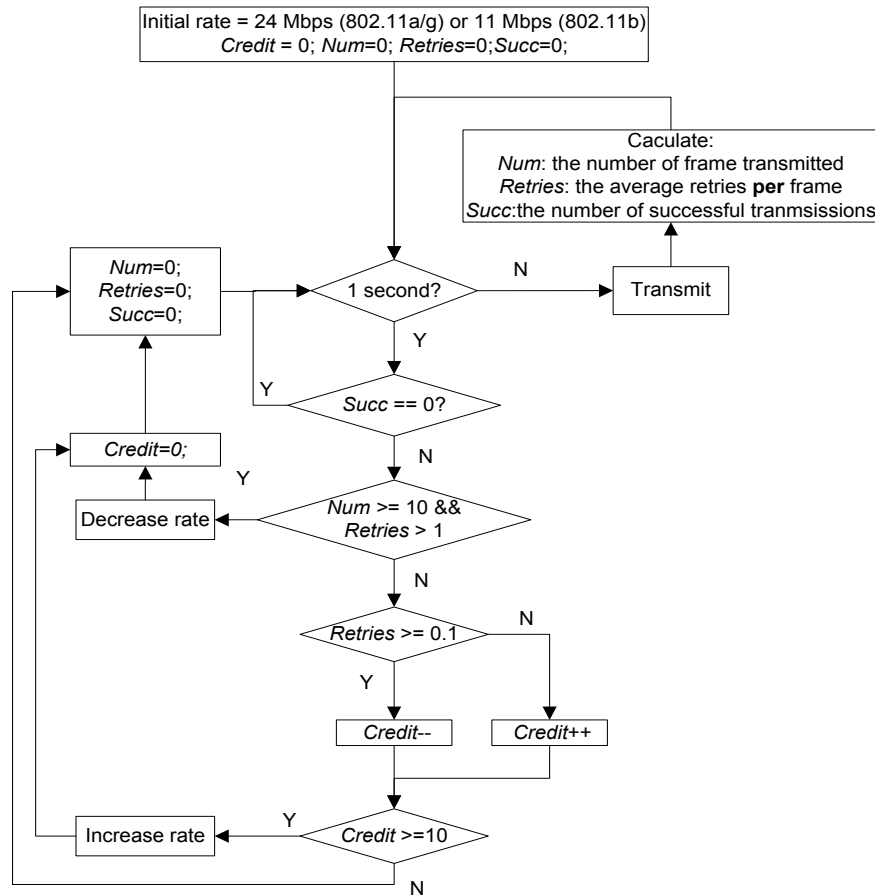
Figure 1.1: Onoe Flowchart

adaptive threshold update is that the interval between two successive rate increases over a stable channel is exponentially extended and fewer rate fluctuations are incurred than *ARF*.

As one the earliest implemented open source rate adaptation schemes, **Onoe** [10] was developed by the MadWifi organization for WiFi adapters with Atheros chips. It is a credit based algorithm and tries to find the best data rate whose loss ratio is less than 50%. *Onoe* adjusts the rate at the end of each 1000 ms cycle based on the collected transmission statistics. Therefore, *Onoe* is insensitive to burst losses but irresponsive to fast changes in wireless channels. The detailed algorithm *Onoe* is illustrated in the flowchart on Figure 1.1.

Rate adaptation scheme **SampleRate** [11] by Bicket is based on periodical transmission statistics. The size of a transmission window is constant (10 seconds) in time. *SampleRate* attempts to identify the bit-rate with the smallest *average* transmission time in the last transmission window. The transmission time for a frame is defined as the time frame from sending it to the receipt of its acknowledgement, which includes the time spent on retransmission and backoff if applicable. In the beginning, *SampleRate* tries the initial transmission at the highest rate. If four consecutive transmissions fails at this rate, the rate is decreased immediately. In each window, a rate is designated as the primary rate. To probe potential better channel conditions in each window, *SampleRate* randomly "samples" one of those rates whose *lossless* transmission time is less than the average transmission time of the rate in use for every tenth frame. Then, *SampleRate* calculates the average transmission time *per* frame for different rates used in transmission at the end of each transmission window. The particular "sample" strategy is illustrated in following example. In IEEE 802.11b, for a packet of 1500 bytes, the lossless transmission times are about 1.873, 2.976, 6.834, and 12.995 ms for the four data rates 11, 5.5, 2 and 1 Mbps, respectively (from Figure 5-1 in [11]). Suppose the rate at some transmission window is 11 Mpbs. After a couple of transmissions, the transmission time is averaged as 3.276 ms (including retransmission time resulted from frame losses), which is larger than the *lossless* transmission time (2.796 ms) of rate 5.5 Mbps. Then, *SampleRate* transmits the tenth packet at 5.5 Mbps hoping that such a transmission might take less average transmission time without any frame loss. Using the rates with smallest average transmission time, *SampleRate* seeks to achieve the best average throughput performance in the long term.

With *SampleRate*, we close this section on rate adaptation schemes that are based on frame loss to estimate channel conditions. In the following, we focus on rate adaptation schemes based on SNR or the received signal strength indication (**RSSI**).

Rate Adaptation Based on Signal Strength/SNR

Receiver Based Auto Rate (**RBAR**) [6] by Holland, Vaidya, and Bahl is the first rate adaptation that takes advantage of the control frames RTS/CTS transmitted at the basic rate. *RBAR* modifies IEEE 802.11 standard in two aspects: 1) the channel reservation in the header of RTS/CTS is represented by packet size and rate, instead of the standard transmission time; 2) a proposed message RSH precedes the data frame to finalize the tentative reservation information in CTS. In *RBAR*, a source station Src selects a heuristic rate (for instance, the rate of the last successful transmission) and includes it and the data packet size in a RTS frame. When the destination station Dst gets the RTS, it retrieves the SNR from physical layer and translates it to a data rate that can be supported by current channel conditions. Then Dst embeds the selected data rate and the data packet size in the CTS frame header. All stations sensing this CTS frame can calculate the *tentative* channel reservation time from the packet size and the rate. Receiving the rate information in CTS, Src makes the final decision about the rate to use. This proposal relies on the use of RTS/CTS.

Heusse *et al* [31] observed a performance anomaly in IEEE 802.11 multi-rate networks: all stations achieve almost the same throughput despite different rates they can support. This is due to the equal probability of all stations with CSMA/CA to access the shared wireless channel in spite of their perceived channel conditions. This is the so called throughput fairness. But this fairness hurts the performance of those stations with high rates and the overall network. Another option is to achieve temporal fairness among stations: each station gets almost equal transmission time rather than equal throughput. *Opportunistic Auto Rate* (**OAR**) [7] by Sadeghi *et al* tackles this challenge. *OAR* probes the rate through the exchange of RTS/CTS exactly as *RBAR* does. However, it differs from *RBAR* in its opportunistic transmission of data frames. After a source station Src selects the rate, *OAR* transmits multiple consecutive frames depending on the selected rate: 5 for 11 Mbps, 3 for

5.5 Mbps and 1 for 2 Mpbs. *OAR* takes advantage of the fragmentation mechanism in IEEE 802.11 to transmit these multiple consecutive frames. Fragmentation allows to keep the channel until all fragments are sent. *OAR* delivers time fairness and also improves the throughput performance with less average overhead of contention time and RTS/CTS per frame.

Scheme *Full Auto Rate* (**FAR**) [12] by Li *et al.* attempts to achieve full data rate adaptation. The authors contend that, as for receiver based protocols, like *RBAR*, if the RTS/CTS is transmitted at a higher data rate when possible, rather than the basic rate, better performance should be achieved because the transmission at the basic rate underutilizes the wireless channel. In *FAR*, an idle station overhears frames from its neighboring stations. Based on the received signal strength, it estimates the rate to each neighboring station. Then, when this station needs to transmit a RTS, it uses the pre-estimated rate. If RTS is transmitted successfully, *FAR* follows the strategy in *RBAR* to estimate the rate for the data frame with the exchange of RTS/CTS. If RTS fails, it is retransmitted at a lower rate.

Received Signal Strength Link Adaptation (**RSSLA**) [5] by Pavon and Choi is a table-driven scheme. It estimates channel conditions based on the *Received Signal Strength* (RSS). A station monitors all frames it can sense and stores the adapted RSS for each neighboring station. The RSS is adaptively updated with a constant low pass filter coefficient:

$$RSS_n = (1 - \alpha) * RSS_{n-1} + \alpha * r \tag{1.1}$$

where RSS_n is the nth adapted RSS and r is the instant RSS retrieved from the last received packet. In transmission, the station retrieves the RSS of the destination station and maps it to the corresponding rate.

15

Theoretically, the modulation scheme is closely associated with SNR [32]. Therefore, the rate should be able to be determined from SNR measurement. However, recent studies of the signals in realistic environments [11, 18] uncover that generally neither SNR nor RSS exhibit a strong correlation with the frame delivery probability at a given rate. These observations limit the effectiveness of SNR/RSS based schemes in practice.

1.2.2   Second Generation: Rate Adaptation with Loss Differentiation

The first generation rate adaptation schemes are effective in environments without collision loss. But, without the ability to diagnose the cause of a loss, they also unduly decrease rates in response to collision losses. Therefore, they can not perform effectively in collision dominated environments or in presence of losses mixed from channel fading and collision. Since most traffic in practice does not use the *optional* RTS/CTS frames to clear the channel [15], losses from collisions are likely to occur. To respond accurately to a frame loss, recent rate adaptation research focuses on the loss differentiation and appropriate loss recovery. In this section, these schemes are chronologically presented.

Pang, Leung and Liew proposed a rate adaptation scheme called loss-differentiating-ARF (**LD-ARF**) [13] for IEEE 802.11 WLANs by combining *ARF* with a loss-differentiating MAC [33] they developed. In *LD-ARF*, loss differentiation is performed at the receiver. *LD-ARF* assumes there is no hidden terminal problem in a WLAN: all stations can hear each other. The authors argue that the frame header is short and thus resilient to wireless channel fading; therefore, if a received frame header can be decoded while the payload can not, this frame corruption is attributed to the channel fading. Otherwise, the frame loss is inferred from a collision because two stations might transmit in the same slot although they both are equipped with carrier sense: the collision is assumed to

destroy both the frame header and body. If the frame loss is diagnosed due to channel degradation, a negative ACK (*NACK*) is sent back to the source station to lower down its rate: the frame source address is assumed available in the decoded frame header. All other operations are the same as in *ARF*.

To introduce the ability of differentiating frame loss between from channel degradation and collision into *ARF* in environments *without* RTS/CTS, Kim, *et al.* proposed *Collision-Aware Rate Adaptation* (**CARA**) [8]. However, *CARA* does rely on RTS/CTS to be "aware" of collision in case of a frame loss. In rate increase, *CARA* does the same as *ARF* does: by counting the consecutive successfully transmitted data frames. However, when a data frame is lost, *CARA* sends a RTS before the retransmission of the lost data frame. The motivation of *CARA* is that the RTS (always sent at the basic rate) is resilient to channel fading. Therefore, if the RTS preceding retransmission of data frames also fails, the data frame loss is likely resulted from collision. Essentially, what *CARA* does is to use RTS/CTS to *suppress* collisions from hidden terminals with implicit assumption that the data frame is lost due to congestion. To mitigate the overhead caused by using RTS/CTS frames, *CARA* suggests that a transmitting station switches its adapter to sense the channel immediately after a transmission is over. If its transmission gets lost and the channel is sensed busy, this loss is attributed to collision without the probing with RTS. It should be noted that the busy channel sensed at the source station does not necessarily result in a collision at the destination station.

Wong, *et al.* proposed the *Robust Rate Adaptation Algorithm* (**RRAA**) [9] and demonstrated its outstanding performance with implementation of it and some other schemes at an access point in a testbed. *RRAA* targets at environments without RTS/CTS. But, it also relies on the use of RTS/CTS in loss differentiation after a frame loss and further to avoid collisions. *RRAA* consists of two elements: rate adaptation (loss ratio estimation and rate selection) and collisions elimination

if collision domination is inferred. *RRAA* gathers the loss ratio from recent transmission statistics over windows. Each transmission window designates the number of data frames to transmit, instead of the time. The size of a window depends on the transmission rate to be used in this window. Therefore, the sizes are not always equal for different rates. A station running with *RRAA* initializes transmission at the maximum rate. The rate is used for all frames in a window. At the end of each window, the frame loss ratio p is calculated to select the rate for the next window. To select a proper rate for the next window, *RRAA* introduces two thresholds: P_{MTL} and P_{ORI}. If $p > P_{MTL}$, the next lower rate is chosen for the next window transmission with the assumption that the channel is degraded and the lower rate might produce better performance. If $p < P_{ORI}$, the rate is increased with the belief that the channel condition is not fully exploited. If ($P_{ORI} <= p <= P_{MTL}$), the rate remains unchanged, but the window *slides* forward to continuously compute the loss ratio for the current rate, rather initialize a new window. Once the rate is decided, it is then used to index a table to extract the corresponding window size. Beyond the rate adaptation, *RRAA* presents a strategy called *Adaptive RTS* (**A-RTS**) to reduce possible collision losses. *A-RTS* maintains two variables RTS_{wnd} and $RTS_{counter}$. RTS_{wnd} essentially "predicates" the trend of channel condition variation. It indicates the number of *potential* consecutive data frames to be transmitted with a preceding RTS. RTS_{wnd} is adjusted as follows: when a frame without RTS is lost, RTS_{wnd} is incremented by one with the assumption that this loss is probably due to collision; when a frame preceded by RTS is lost, or a frame without RTS succeeds, RTS_{wnd} is halved. However, it is $RTS_{counter}$ that controls if a RTS should be really used before a data frame. If $RTS_{counter}$ is less than RTS_{wnd}, a RTS/CTS handshake is pursued before a data frame is transmitted. When integrating the rate adaptation and *A-RTS*, *RRAA* does not consider the loss of RTS into the loss ratio computation. Although *RRAA* uses RTS frames to mitigate collisions, it does not promptly decrease its rate even if the loss is

inferred from channel fading. To respond to channel variation more quickly, *RRAA* also suggests an optimization: adjust the rate if necessary in the middle of a window, instead of at the end of the window.

1.2.3   Rate Adaptation Schemes Classification

In this section, before the categorization, we first present the criteria to be used. Rate adaptation generally contains two essential operations: the assessment of channel condition and the corresponding rate adjustment. The categorization criteria are identified from these two operations.

- **Channel Condition Indicator**: Most of the early rate adaptation schemes (e.g. *RBAR* [6], *OAR* [7], *FAR* [12], and *RSSLA* [5]) consider **Signal-to-Noise Ratio** (SNR) or **Received Signal Strength Indication** (RSSI) as an indicator of the channel conditions. Such motivation intuitively originates from the wireless signal propagation principle. Other early schemes (*ARF* [2], *AARF* [3], *Onoe* [10]) consider **frame loss** as the channel conditions indicator. Since Aguayo *et al.* [18] and Bicket [11] observed that neither SNR nor RSSI demonstrates a strong correlation with delivery probability at any given data rate, all recent schemes, such as *SampleRate* [11], *CARA* [8], and *RRAA* [9], use **frame loss** as the channel conditions indicator. For those schemes using frame loss as indicator, they can be further characterized as either *winning streak* or *statistics* based. A "winning streak" scheme relies on a given number of **consecutive** successfully transmitted data frames. A "statistics" based scheme gathers transmission statistics over the most recent history to increase or decrease its data rate.

- **Loss Differentiation**: Rodrig *et al.* [15] discovered that most of the traffic is transmitted without presence of RTS/CTS to reduce overhead. Therefore, frame losses in a WLAN network now can be contributed by both channel degradation and collision. Consequently, the

Schemes	Channel Condition Indicator		
	Frame Loss		RSSI/SNR
	Winning Streak	Statistics	
LD	*CARA*, LD-ARF	**RRAA**	LDRA
Non-LD	ARF, **AARF**	*Onoe*, **SampleRate**	RBAR, FAR, RSSLA

Table 1.1: Categorization of Rate Adaptation Schemes

ability to differentiate frame loss between these two contributors is critical because each of them requires different response in rate adjustment. Loss differentiation consists of 1) diagnosing the cause of a frame loss between channel degradation or collisions, and 2) taking appropriate actions for each cause of loss. Although rate adaptation in IEEE 802.11 networks has been studied for years, most schemes do not explicitly diagnose a frame loss, particularly those schemes based on RSSI/SNR. Recently proposed rate adaptation schemes (e.g. *CARA* [8] and *RRAA* [9]) react to collisions by dissipating the congestion and react to channel degradation by lowering the data rate.

According to above criteria, we classify typical rate adaptation schemes in Table 1.1, where LD stands for *Loss Differentiation*, Non-LD for *Non Loss Differentiation* (i.e., schemes that do not differentiate the cause of a frame loss).

1.3 Motivation

In a multi-rate wireless network, when a mobile station starts up and needs to send packets, it generally has the following options to select a transmission rate: the basic (lowest) data rate, last successful transmission rate, or an estimated data rate. By default, most current rate adaptation

schemes use the basic data rate for the first frame so that it can be captured by every station within the transmission range. Li *et al.* [12] observed with simulation that transmitting RTS/CTS at an estimated data rate, rather than the basic rate, can improve throughput. Therefore, similarly, a well-predicted initial rate at the sender may be challenging but benefits the network performance in environments with/without RTS/CTS. The estimation of the initial rate also benefits the transmission after a long inactivity. Suppose station A wants to send packets to station B. But station B is inactive and has not transmitted packets for a long time (e.g. several minutes), the sender station A is unaware of the link status to station B.

Since RTS/CTS are overhead to data frames, they are optional in IEEE 802.11 standard and thereof they are not used in most transmissions. Their absence in practice are shown by Rodrig *et al.* [15] with collected real time traffic in a conference. Without the RTS/CTS preceding, data frames are more likely to collide with each other. Frame losses due to collision can be addressed with collision suppression techniques without decreasing rate. But the frame loss due to channel degradation has to be recovered with lower rates. Therefore, a loss differentiation or diagnosis strategy is highly desirable in rate adaptation.

Also, from the collected real time traffic, Rodrig *et al.* [15] observed that retransmissions occupies 46% of the whole data transmission time and that most of the transmissions occur at 1 Mbps, the lowest rate. Such observation implies that current rate adaptation schemes implemented in commercial products are inefficient and an innovative scheme is highly desirable.

In the following, we first discuss some anomalies observed while implementing, testing, and debugging rate adaptation schemes on a Linux based testbed. Such observation also motivates and provides insights for the design of our final scheme *ERA*. Then, we highlight some design principles for an effective and efficient rate adaptation scheme.

1.3.1 Observations

While implementing and debugging rate adaptation schemes on a Linux based testbed, we observed some anomalous behaviors.

- *Inconsistent Performance*: The schemes that requires RTS/CTS at the basic rate to differentiate losses, (e.g. *RRAA* and *CARA*), do not perform as effectively in channel degradation environments as in collision dominated environments. The root cause is that they do not have proactive actions to *promptly* recover a frame loss due to channel fading. The success of the RTS/CTS at the basic rate does not improve the probability of the data frame successfully retransmitted still at the old rate. Consider a frame loss occurs due to fading and the RTS/CTS is sent (at the *basic* rate) to precede the retransmission. The RTS/CTS frames may succeed at the basic rate, but the retransmitted data frame definitely fails because the rate was not decreased to counter the fading. Therefore, although RTS/CTS control frames reduce the congestion frame loss, they can not help with channel fading.

- *Mistake in Loss Counting*: So far, all frame loss based schemes (either loss ratio based or "winning streak" based) mistakenly take the frame loss from collision into consideration in computing their loss statistics used as an indication of channel degradation only, even if some of them are able to diagnose the cause of a loss. The loss ratio or consecutive successful transmissions in literature are primarily used for detecting the channel degradation, not the collisions. Therefore, it is not reasonable to include collision losses in the measurement of loss ratio or consecutive successful transmissions. Namely, the counting of frame losses in existing schemes is too pessimistic in estimating channel degradation.

- *Clumsy Response*: Schemes that are based on frame loss statistic with a large cycle or window might get *numb* to channel fast variation unless some rate probing strategy is adopted in the middle of a transmission cycle. Although *RRAA* proposes an "optimization technique" to solve this problem, one failure case is still likely to occur. Such an issue is illustrated in Figure 1.2. Consider the transmission rate is 36 Mbps. From *RRAA* [9], the transmission window size for this rate is 40 frames. The loss ratio threshold P_{MTL} to increase the rate to 48 Mbps is 11.50% and the loss ratio threshold P_{ORI} to decrease the rate to 24 Mbps is 33.63% [9]. Suppose after a period of collision, the counter $RTS_{counter}$ reaches 32. Now, the channel fades after the transmission of the 34th frame. Then the six left transmissions in this window are all preceded by RTS/CTS, but they all will fail because the rate is not promptly decreased to respond to the fast channel variation.

- *Inefficient Transmission at Basic Rate*: It is not efficient to differentiate loss with the transmission at the *basic* rate like *LDRA* [34] or some schemes relying on RTS, even if the frame is very short. Generally the channel is rarely fading abruptly. The time for a transmission at the basic rate is long enough to allow several transmissions at the normal rate. This is particularly wasteful in case of a collision where a transmission at basic rate does not any help.

1.4 Design Principles

Based on the above observation and successful experience from other rate adaptation schemes, we summarize the design principles for an effective and efficient rate adaptation as follows.

- *Fast Recovery*: Despite that SNR is not a good indicator [11,18] of successful frame delivery, the SNR based *LDRA* [34] yields a significant throughput improvement when there are no
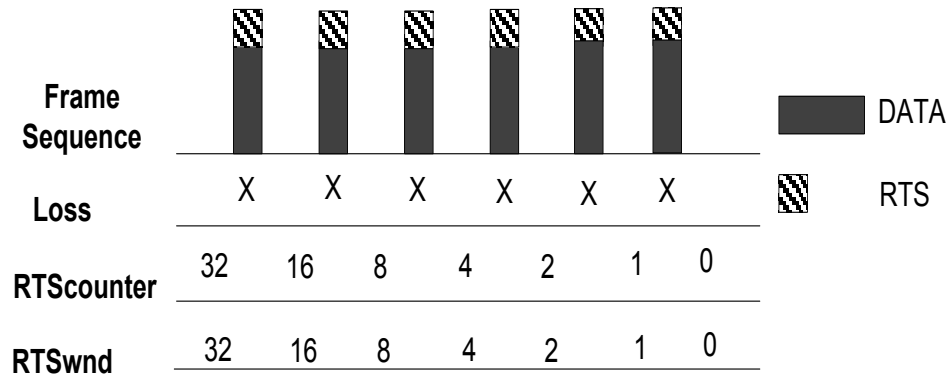
Figure 1.2: Continuous transmission failure anomaly

collision losses and no hidden terminal problem. The strength of *LDRA* stems from its re-transmission strategy that provides *prompt* recovery of lost frames. Therefore, a fast recovery strategy can improve network performance significantly.

- *Accurate Loss Differentiation*: Since different type of loss can not be addressed with the same solution, an accurate differentiation of loss causes is valuable for performance improvement. A common sense is that decreasing the rate for collisions is unnecessary and wasteful. *CARA* and *RRAA* illustrate that a rate adaptation scheme not only should not decrease its rate in response to collisions, but it should proactively diffuse congestion: both schemes use RTS/CTS frames to minimize congestion.

- *Consistent Performance in Different Environments*: a rate adaptation should not only be ef-fective and efficient in collision environment, it should also work outstandingly in collision free or channel degradation environments.

24

- *Fast Response to Channel Degradation*: After the cause of a frame loss is diagnosed from channel degradation, the network performance can be improved if the rate is adjusted promptly, rather than remains till the end of the cycle or window.

- *Separating Collision Loss*: When we compute frame loss statistics for channel degradation, the loss from collision in the middle of counting should not be taken into account.

So far, RTS/CTS frames are used [8, 9] to probe the channel and/or alleviate congestion to yield an efficient rate adaptation. Since RTS/CTS control frames may be a significant overhead and usually are not used, it is of utmost interest to find other ways than RTS/CTS to dissipate collisions.

This chapter presents our initial work on rate adaptation. This rate adaptation is named *Beacon Assisted Rate Adaptation* (**BARA**) because beacon frames are exploited to estimate instantaneous transmission rates In this chapter, we first discuss the design details of *BARA*. Then, its performance evaluation is presented.

2.1 Design of BARA

In this section, we discuss the details of the *BARA* scheme. We start from the estimation of the initial rate with *beacon* frame, then we illustrate the adaptive rate adaptation during an ongoing communication. Finally, we discuss the basic rate retransmission after a frame loss to reduce the number of retransmissions in loss recovery.

2.1.1 Rate Adaptation with Beacon

Since beacon frame is broadcast periodically, this leaves "room" to estimate data rate without introducing overhead. In whatever network mode, infrastructure or infrastructureless, beacon is mandatory. These periodic beacon frames received at a mobile station allow it to determine the statistics of the channel conditions (e.g. signal to noise ratio, signal strength, and loss rate). Based on such wireless channel information, the mobile station can calculate the best data rate to the source mobile station who initiates the beacon (in ad hoc networks, it is another mobile station; in WLAN it is the access point), and then record this data rate information into a table for later use. The table may be indexed by the destination mobile station address. Each tuple may include the modulation

level or data rate as content. When this mobile station needs to transmit data frames to another

mobile station, it looks up the rate table for a data rate that it can utilize to communicate with the

target mobile station. It instructs its physical layer to transmit the data frame by making use of the

associated modulation corresponding to that data rate. The main drawback of this basic strategy is

the estimated rate might not be exactly *real-time* because estimation is only adjusted every beacon

interval. But this mechanism is good for the initial rate estimation, because the estimated rate is still

more precise than a randomly selected data rate at the beginning of the transmission. It is also more

efficient than the basic data rate.

The following algorithm describes this strategy:

When a frame is received:

```
if (FrameType == Beacon)
{
        Index = GetFrameMacAddress();
        ChannelStatistics = GetChannelStatisticsFromPhy();


        if (ChannelStatistics > ThresholdRate_11)
                Rate = 11Mbps
        else if    (ChannelStatistics > ThresholdRate_5.5)
                Rate = 5.5Mbps
        else if (ChannelStatistics > ThresholdRate_2)                     10
                Rate = 2Mbps
        else{
                Rate = 0Mbps;
                Log(signal is too weak. No channel
                        to destination mobile station);
        }
```

27

```
                RecordRate2Table(Rate, Index);

        }
```

When a frame needs to transmit:

```
    Index = GetDestinationMacAddress();

    GetRateFromTable(Rate, Index);


    If (Rate == 0){

            Log(Transmission can not complete.

            No channel to destination mobile station);

            return 0;

    }

    TransmitDataPacket(Rate);
```

2.1.2 Adaptive Rate Adaptation During Ongoing Communication

This mechanism is applicable to instantaneous rate adaptation per frame for an ongoing stream of frames. As stated in the above subsection 2.1.1, dynamic data estimation with only beacon frame might not be real time, because beacon frame can only be sensed every beacon interval, not available at any time. If all frames flying during communication can be used to estimate data rate, the estimation would be more accurate. Thus a station in "capture" range can dynamically estimate the data rate of the link to those transmitting stations based on the channel condition statistics. Therefore, after the initial rate is estimated from the most recent beacon frame, an adaptive mechanism aims to predict data rates on both beacon frames and all other data frames that the station can receive. This is helpful especially in two scenarios.

1. The first case happens within a transmission sequence of data frames. In this case, a sender mobile station sends a stream of data frames to a mobile station with one data frame at a time. When the receiver captures the data frames, MAC protocol mandates the receiver to respond with an acknowledgement control frame. Therefore, the sender gets the channel condition information from this acknowledgement packet and adapts its data rate for the next frame transmission if necessary. Since a significant number of packets can be transmitted within one beacon interval, even if the channel condition varies within a duration shorter than a beacon interval, the practical data rate can still be promptly adapted by such a adaptive strategy.

2. The other case occurs when a mobile station is just eavesdropping. In wireless communication, a mobile station can sniff all wireless signal within its capture range through its antenna because the wireless medium is unguided in essence, not like point-to-point in wired network [35]. Also, IEEE802.11 requires that each mobile station must listen to any flying frame to determine whether this packet is addressed to itself or not, unless the station is in power save state. Therefore, when a mobile station is filtering the received packet, even though it is not the receiver station, it can still get the channel condition information between itself and the packet sender. Therefore, the data rate of wireless channel can be adapted. Each station in power save mode still needs to wake up for beacon frame periodically. Thus, even in this *Adaptive Rate Adaptation* technique, periodic beacon frame is still necessary and helpful to estimate the channel condition.

For instantaneous rate adaptation, if the rate is estimated per frame, the data rate might frequently fluctuate due to fast channel condition variation, if only the latest single received frame is

used to the predict channel data rate for the next transmission. This happens especially when a mobile station is experiencing the ping-pong effect in handoff where the data rate might iterate between two levels of rate. It results in retransmissions. Multiple retransmissions definitely hurt the network performance. Therefore, to solve the above rate fluctuation problem, the data rate should not only be adjusted per frame, but also adaptively adjusted based on multiple history frames. Namely, it is more important to predict the trend of the channel condition variation to adjust data rate before the channel changes than just the instantaneous rate.

The rate adaptation core procedure in *Adaptive Rate Adaptation* is still the same as in *Rate Adaptation with Beacon* strategy in Section 2.1.1. The only difference is an adaptive coefficient introduced to smooth the data rate on wireless channel with history rate information.

One low pass filter α is introduced in the channel statistic calculation:

$$ChanStat_t = (1 - \alpha) \times ChanStat_{t-1} + \alpha \times ChanelStatistics \qquad (2.1)$$

$(0 < \alpha < 1)$

It can also be expressed as:

$$ChanStat_t = ChanStat_{t-1} + \alpha \times (ChannelStatistics - ChanStat_{t-1}) \qquad (2.2)$$

Here, the *ChanStat* is the cumulative prediction of potential channel status. It is used to predict the channel variation trend. *ChannelStatistics* is the instantaneous channel status, which is calculated from the frame just received.

α is not a constant: it is dynamically updated as following:

$$\alpha = \frac{ChanStat - Threshold_{low}}{Threshold_{high} - Threshold_{low}} \tag{2.3}$$

Here $Threshold_{low}$ and $Threshold_{high}$ are the corresponding channel statistic thresholds for rate estimation at each data rate level determined by physical hardware. These thresholds vary for different data rates. For instance, Φ is the low threshold for 5.5 Mbps. if the 2 Mbps and 5 Mbps are two neighbor data rate levels in our data adaptation, Φ also is the high threshold for 2 Mbps.

From the Formula 2.3, as the cumulative channel signal strength *ChanStat* approaches to $Threshold_{high}$, α increases, and thus instantaneous channel signal strength contributes more to the adapted rate. This helps data rate adaptation quickly increase to higher data rate levels if more frames are transmitted successfully under good channel conditions. This cumulative prediction can also alleviate rate fluctuation in case of heavily fluctuating channel conditions, for example during a handover. Therefore, the channel data rate is smoothed to a certain degree.

Adaptive Rate Adaptation operates with all sensed frames, including both broadcast control frames like beacon and data transmission packets. Thus, the algorithm for *Adaptive Rate Adaptation* evolves from algorithm for *Rate Adaptation with Beacon* as following.

When a frame is received:

```
b = constant;

Index = GetFrameMacAddress();

ChannelStatistics = GetStatFromPhy();

Threshold_h = GetCurrentHighThreshold(Rate);

Threshold_l = GetCurrentLowThreshold(Rate);

a = b*(ChanStat−Threshold_l)/(Threshold_h−Threshold_l)

ChanStat = ChanStat+a*(ChannelStatistics−ChanStat);
```

```
if (ChanStat > ThresholdRate_11)

    Rate = 11Mbps                                                    10

else if (ChanStat > ThresholdRate_5.5)

    Rate = 5.5Mbps

else if (ChanStat > ThresholdRate_2)

    Rate = 2Mbps

else{

    Rate = 0Mbps;

    Log(signal is too weak. No channel

        to destination mobile station);

}
                                                                     20

RecordRateToTable(Rate, Index);
```

2.1.3 Basic Rate Retransmission

It is impossible for a rate adaptation scheme to accurately predict the actual data rate all the
time. Namely, there are definitely mismatches between the predicted data rate and the actual data
rate that the channel supports throughout an entire transmission. In such case, the receiver does
not successfully receive the data packet, therefore does not generate the acknowledgement. But the
sender is waiting for an acknowledgement to estimate the actual data rate for the next transmission.
A deadlock happens in this scenario. The only way to avoid such deadlock is to retry the same frame
at a lower data rate. But, minimizing the retransmissions after a transmission failure is not trivial.
Thus we propose *Basic Rate Retransmission* that can efficiently address the above problem. When
a sender fails to transmit a frame, the sender does not retry the transmission at the *same* data rate

used in the failing transmission, because the channel condition might have deteriorated. Instead, it retries at the basic data rate just as for RTS/CTS frames, at which the receiver can certainly receive the packet, if it is still in the communication range. The reason we retry directly at basic rate (and not at the next lower level rate), is to avoid more failures. For example, if a transmission fails at 11 Mbps and the channel can only support the 2 Mbps basic rate temporally, the transmission at 5.5 Mbps, will still fail. But, the frame at 2 Mbps data rate is still able to be decoded, even in case that the channel condition is robust enough for 5.5 Mbps. Then, when the sender catches the acknowledgement frame for the basic rate data frame, the sender is able to adapt its instantaneous data rate for next frame transmission based on the channel information of acknowledgement frame. With this strategy, the transmission is immediately recovered with a retransmission at the basic data rate. The number of retransmissions after a transmission failure can be reduced to only *one*. *Basic Rate Retransmission* improves the network performance especially when network load is heavy, because multiple retransmissions result in longer delay between two successful consecutive transmissions due to the binary exponential backoff in MAC protocol. It should be stated that although this variant can be combined with the previous two mechanisms that we propose, it also can work individually with any other rate adaptation strategy to minimize the retransmission in case of frame loss.

2.2   Performance Evaluation

2.2.1   Simulation configuration

The simulation is performed on *ns-2* [36] version 2.29 for both WLAN and ad-hoc modes. Only three levels of data rates are chosen in simulation: 11 Mbps, 5.5 Mbps and 2 Mbps, among which 2 Mbps is defined as the basic rate. These simulations use two-ray ground model [32] as
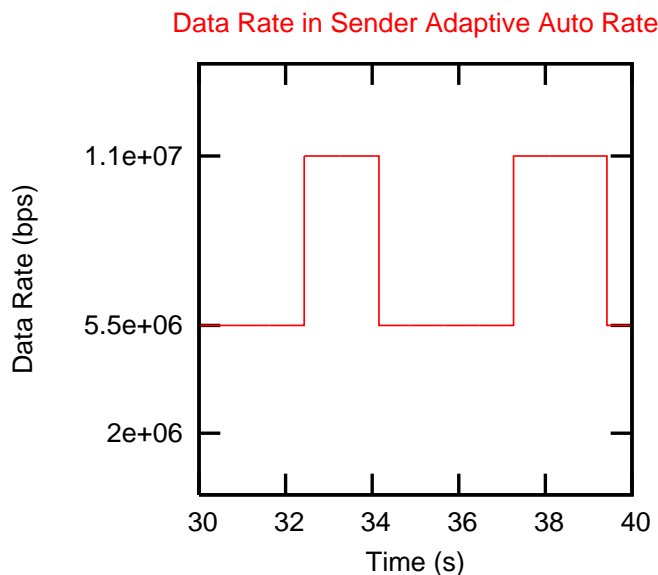
Figure 2.1: Data Rate Adaptation

channel fading. All motion scenarios are generated by CMU mobile scenario program "setdest" in *ns-2* package. And all traffic flows for ad-hoc network simulation are generated by CMU scripts "cbrgen.tcl" in *ns-2*. In the following result figures, we mainly compared our mechanism with *ARF* proposal unless there is special explanation.

2.2.2 Data Rate Smoothing

Compared to the rate fluctuation in *ARF*, the data rate does not vibrate widely with our proposed strategy. The result can be observed from Figure 2.1.

2.2.3 Throughput performance

We separately evaluate the performance of rate adaptation with beacon only technique and adaptive rate adaptation technique. The improvement for WLAN network is really dramatic, as illustrated in Figure 2.2 and in Figure 2.4. Figure 2.2 represents the improvement for multiple
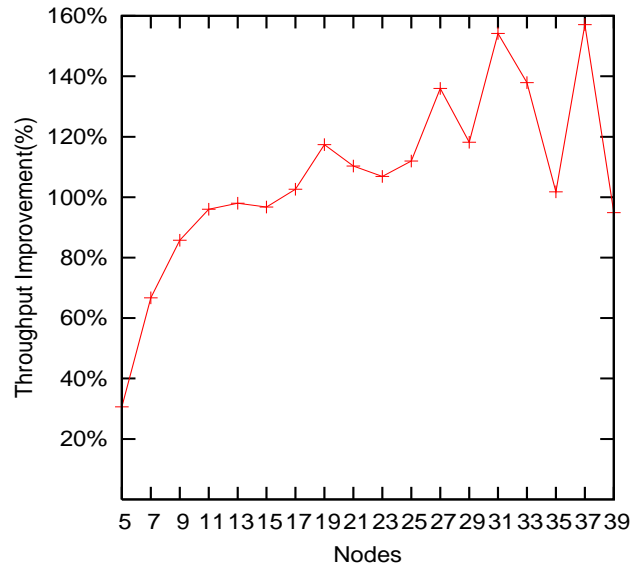
34

Figure 2.2: multi-node Rate Adaptation with only Beacon Frame

nodes operating in rate adaptation with only beacon frame. Figure 2.4 shows the improvement on a network with multiple nodes with different fixed adaptive coefficients. In both figures, the X-axis represents the number of nodes (including one access point) in an area of 100x100. The Y-axis is the throughput improvement by percentage(%). As network density increases, more than 100% improvement can be achieved. However, the Figure 2.3 shows that *BARA* with adaptive coefficient in formula 2.3 does not perform so well as with constant coefficients as illustrated in Figure 2.4.

Figure 2.5 shows the improvement from data rate adaptation with only beacon frame in Ad-Hoc network mode. The X-axis represents the number of nodes in the networks and the Y-axis represents the throughput improvement by percentage. There are 3 scenarios for different network sizes in this figure: 100x100, 150x150 and 200x200. As can be observed in Figure 2.5, the improvement increases as the network density increases. It is because *BARA* adapts its data rate with the proper signal strength (or communication distance) regardless of frame losses from collision. But
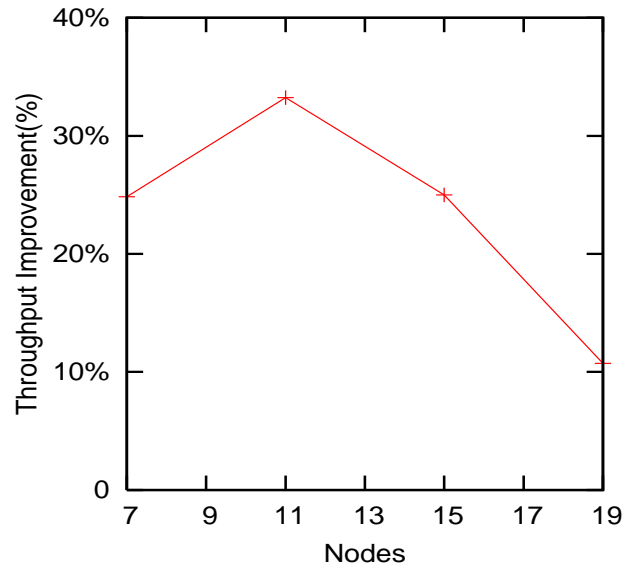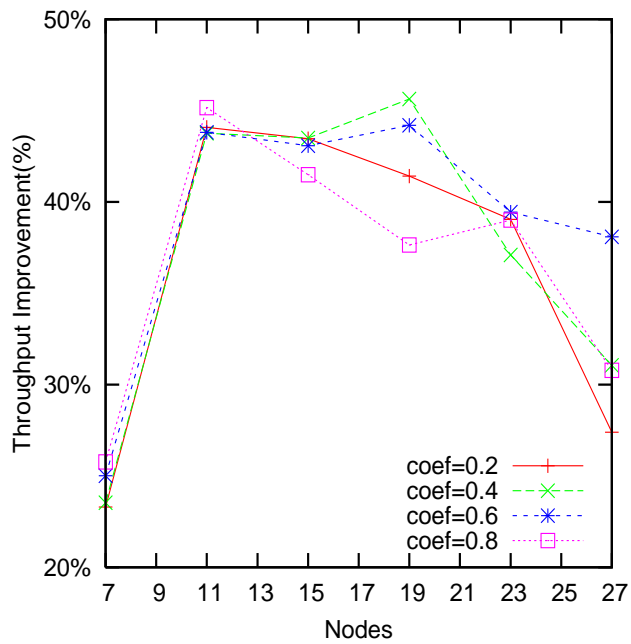
35

Figure 2.3: multi-node in BARA in WLAN

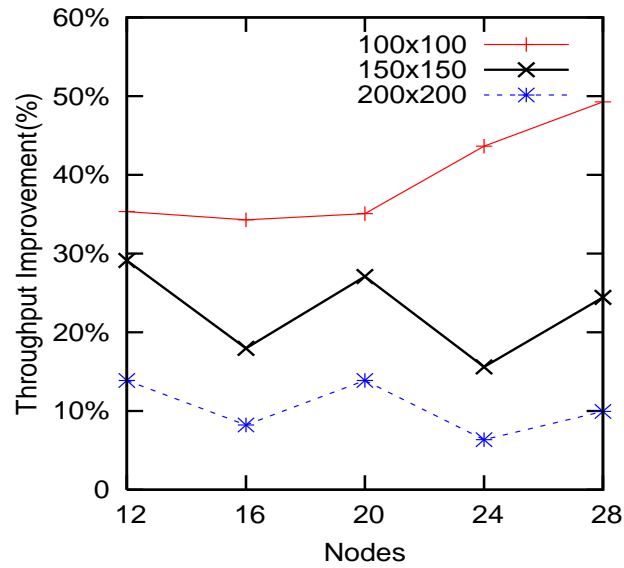Figure 2.4: multi-node in BARA with Coefficients

36

Figure 2.5: Rate Adaptation with only Beacon Frame in Ad-Hoc network

ARF strategy is impacted extensively by frame losses. When the network density increases, so does frame collision. Thus, *BARA* can achieve more improvement.

The simulation result for *BARA* with different constant coefficients in Ad Hoc network is illustrated in Figure 2.6. The X-axis denotes the number of nodes in the network area of 100x100 and the Y-axis represents the throughput improvement. The different results lines in Figure 2.6 show the result for different coefficients used in *BARA* adaptive rate adaptation mechanism. From Figure 2.6, we can observe that, unlike in WLAN mode, *BARA* with constant coefficient is outperformed by *BARA* with adaptive coefficient of formula 2.3. Also, we can observe that the different fixed coefficients do impact on the improvement, the result of coefficient of 0.3 is better than other four counterparts.
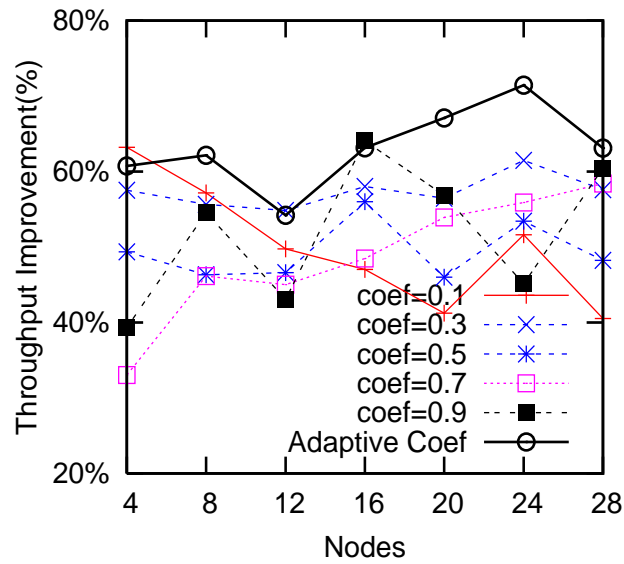
Figure 2.6: multi-node in BARA with Coefficients in Ad-Hoc

Delay Jitter Improvement:

The delay jitter improvement is shown in Figure 2.7, in which only one access point (AP) and one mobile station communicate with each other with continuous CBR traffic. In this figure, the X-axis represents the two mechanisms that are compared and the Y-axis is the percentage of improvement. It can be observed that: although the jitter improvement by *BARA* with only adaptive coefficient mechanism (SAAR in figure)is not noticeable, only a little more than 10%, there is still marginal improvement by *Basic Rate Retransmission* (BRR in figure). It can improve the delay jitter as much as 25% for station. This supports our statement on *Basic Rate Retransmission* for improvement on delay and jitter because it minimizes the delay to just once retransmission.
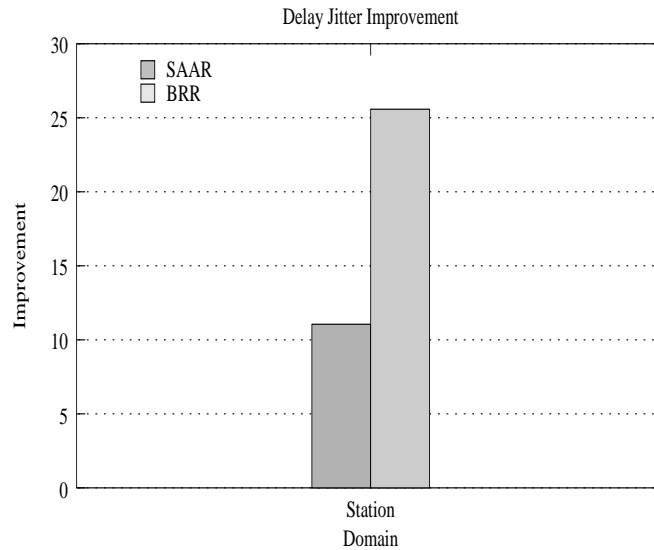
Figure 2.7: Delay Jitter Improvement by SAAR/BRR

2.3 Discussion

Two key strengths of *BARA* are: 1) the exploit of periodical and mandatory beacon frames in rate adaptation without introducing any extra control frames overhead and 2) the basic rate technique to reduce the number of retransmissions in loss recovery. However, *BARA* is based on the SNR that is studied not to be a good channel indicator in practice. Therefore, although the simulation shows its benefits, its implementation and application in the real world is limited. Also, *BARA* does not have any functionality to diagnose the frame loss cause in composite lossy environments.

CHAPTER 3

LDRA: LOSS DIFFERENTIATED RATE ADAPTATION

Although *BARA* estimates the transmission rate without introducing overhead, like its many peer schemes, it does not have any loss differentiation capability. To integrate such a loss diagnosis ability, we propose a rate adaptation scheme *Loss Differentiated Rate Adaptation* (*LDRA*). It differentiates the frame loss cause by retransmitting the lost frame at the basic rate directly. Note that *LDRA* works for IEEE 802.11 in infrastructure mode as well as in infrastructureless (ad hoc) mode.

In this chapter, we first discuss the details of the scheme *LDRA*. Then we justify the use of the basic rate retransmission technique used by *LDRA*. Finally, its performance evaluation is presented.

3.1 Design of LDRA

LDRA mainly consists of three components: (1) data rate estimation using Signal-to-Noise Ratio (SNR) from the beacon, (2) data rate selection for retransmissions after a frame loss, and (3) frame loss differentiation with appropriate actions for each type of frame loss. The following presents each in order.

3.1.1 Rate Estimation

The rate estimation in *LDRA* is similar to the mechanism we adopt in *BARA* in Chapter 2. In an IEEE 802.11 infrastructure network, a station that is associated and synchronized with its access point knows the beacon interval. Each station periodically listens for a beacon frame that can be used to measure the channel conditions through the signal-to-noise ratio (SNR), the received signal strength (RSS), the frame loss rate, or the error bit rate. Based on such collected information,

the station estimates the most appropriate data rate to communicate with the source of the beacon. The smaller is the beacon interval, the more accurate is the estimation as channel conditions would less drastically change. This is particularly true for mobile nodes at low speed or stationary nodes like mesh networks. Even if the beacon interval is comparably large, this estimated rate is still more appropriate than a randomly selected or "guessed" *initial* data rate at the beginning of the transmission.

A supplement to the beacon estimation is to take all communication frames into consideration for adaptive data rate adaptation after two stations start their communication. Such a strategy can provide more accurate estimation if there are multiple transmissions within a beacon interval.

When SNR is collected from the physical layer, *LDRA* adjusts the data rate R_n such that $R_n = \alpha * R_{n-1} + (1 - \alpha) * r$ where r is the instantaneous rate estimated from the SNR. When α is a constant, this equation is similar to Pavon and Choi [5]. We use an adaptive α such that $\alpha = \beta * \frac{R_n - R_{low}}{R_{high} - R_{low}}$ and β is a constant from 0 to 1. R_{low} is the SNR low threshold for a given data rate (e.g. 5.5 Mbps); and R_{high} is the SNR high threshold for the same data rate.

3.1.2   Frame Loss Differentiation:

Since no data rate adaptation scheme is perfect, the sender may transmit at an overoptimistic data rate, leading to frame losses. In practice, due to the sparse of RTS/CTS, a data frame can also be lost from transmission collision. A frame loss may hint to decrease the data rate. In *ARF* [2] and other frame loss based rate adaptation schemes, the number of lost frames or the frame loss rate determine the data rate. *ARF* decreases the data rate whenever two consecutive frames get lost, no matter what caused the frame loss. However, a lower rate would unduly hurt performance if the

frame loss was due to a collision. Thus, it is critical to determine the cause of a frame loss and take appropriate actions in a data rate adaptation scheme.

LDRA exploits the retransmission (after a loss) at the lowest rate technique to accurately diagnose the cause of a frame loss (collision or channel degradation). When a frame loss occurs for the very first time, the sender retransmits the lost frame at the lowest data rate, instead of the same rate. Such retransmission allows the discrimination based on the following cases:

- If the receiver is still within the radio range, then the receiver would most likely receive this retransmitted data frame in case of channel degradation because of the lowest data rate. Thus the sender can receive the acknowledgement and correctly infer that the loss is due to channel degradation.

- If the retransmitted frame is lost, *and* no beacon is received during the latest *N* beacon periods, the sender can correctly infer that the receiver is *out of range* (for the current channel quality).

- If the retransmission at the lowest data rate is lost, but a beacon frame has been received in the latest *N* beacon periods, then the sender can infer that the loss is rather due to collisions.

After diagnosing the loss, *LDRA* will take the appropriate actions for each type of loss as explained in the next section.

3.1.3 Reactions to Frame Loss:

After a frame is lost, three reactions are proposed to improve network performance upon different diagnosed frame loss causes.

Frame loss due to channel degradation: Since the loss is not due to a collision, then the sender should *not* double its contention window as stipulated in IEEE 802.11 [1]. Moreover, based

on the frame exchange retransmitted at the lowest rate, the sender is able to estimate the new appropriate data rate.

Frame loss due to out of range of the receiver: The sender will immediately pause its transmissions until it detects a new beacon frame. This results in two benefits: 1) it eliminates unnecessary network traffic and therefore reduces collisions (or hidden/exposed terminal problems), and 2) it saves power.

For the frame loss identified from a collision: As usual, the sender will invoke, for good reason, the binary exponential backoff mechanism, but will not decrease the data rate: a lower data rate does not remedy or alleviate collisions. On the contrary, a lower data rate increases signal coverage and thus may *worsen* interference and collision. Therefore, the sender should maintain the current data rate after a collision. This is a critical difference from traditional data rate adaptation algorithms without loss differentiation.

The core algorithm of *LDRA* is illustrated in the flow chart in Figure 3.1.

3.2 Justification for Retransmissions at the Basic Rate

In IEEE 802.11b [25], there are four modulation schemes: BPSK (1Mbps), QPSK (2Mbps), CCK5.5 (5.5Mbps), CCK11 (11Mbps). We analyze the expected time required to successfully transmit a frame for each modulation (each rate) in different signal-to-noise ratio (SNR) environments. In IEEE 802.11, the Frame Error Rate (**FER**) is associated with a frame length of 1024 bytes. If the bit error rate (BER) p is very small and losses are independent, then the corresponding FER can be approximated as $p * 1024 * 8$. The expected number of transmissions for a frame to be successfully delivered is $\frac{1}{1-FER}$. The expected transmission time at the rate bw is $\frac{1}{(1-FER)*bw}$. Based on the bit error rate data reported by Wu [37], Figure 3.2 illustrates the relationship between
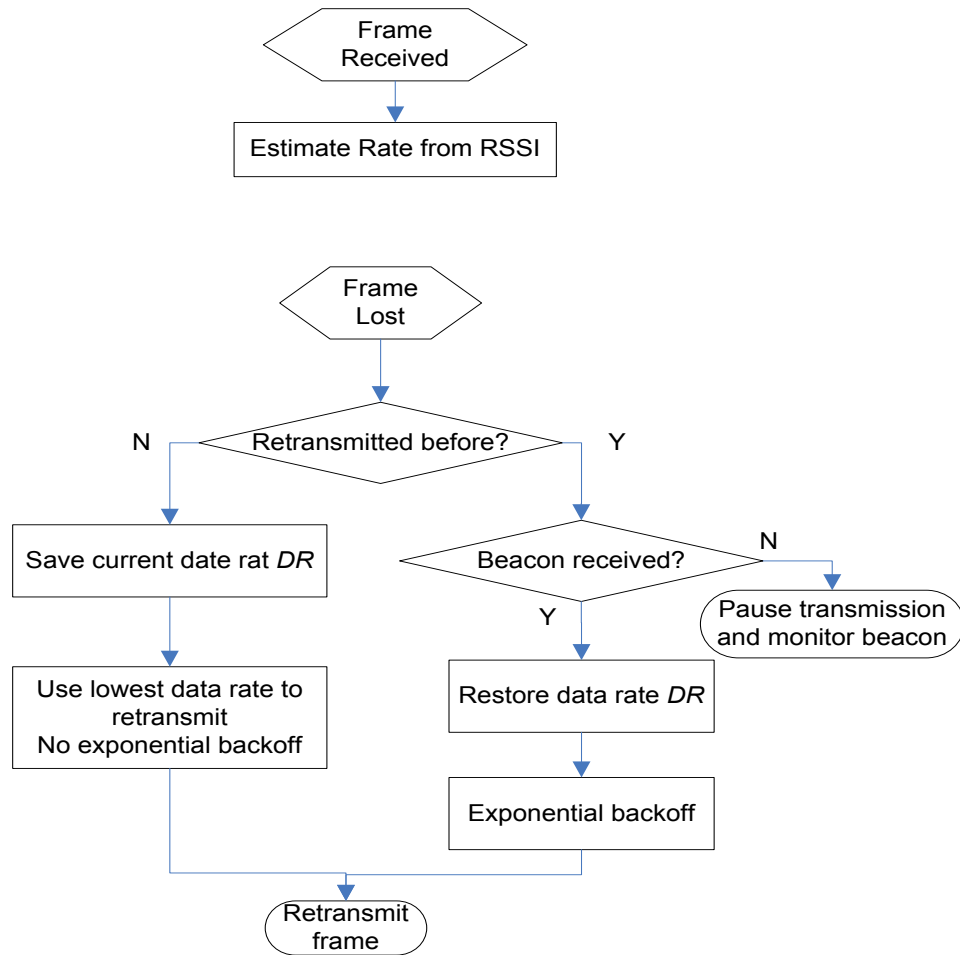
Figure 3.1: LDRA Algorithm Flow Chart

expected transmission time and SNR for different modulations. The x-axis represents the SNR, and the y-axis is the expected transmission time. Bianchi *et al.* [38] observed that the SNR in outdoor environment is usually less than 6 db for 802.11b/g. From Figure 3.2, it can be observed that, if a frame is transmitted at 5.5 Mbps or 11 Mbps in an environment with SNR less than 6 db, it has to be retransmitted so many times that transmissions are unlikely to succeed. But for the lowest rate of 1 Mpbs or 2 Mbps, a frame will successfully be transmitted almost every time in low SNR environment. Thus, in traditional retransmission schemes, even if a frame is lost due to channel fading with a very low SNR, CSMA/CA always assumes the loss to be due to collision. It backs off and retransmits the frame at the same high data rate again for several times. As illustrated by Figure 3.2, this retransmission strategy is wasteful and doomed for frame losses resulting from channel degradation. Thus, it is important to retransmit at the lowest data rate and differentiate the cause of a frame loss.

3.3   Performance Evaluation

In this section, we mainly evaluate the network performance improvement by *LDRA* over other existing algorithms with composite frame losses from wireless transmission collision and channel degradation.

3.3.1   Simulation Configuration

LDRA is simulated in IEEE 802.11b WLAN and ad hoc modes with ns-2 [36] (version 2.29). Three data rates are used: 11 Mbps, 5.5 Mbps and 2 Mbps. The proposed scheme is compared with *ARF* [2] and *Adaptive Auto Rate* [5] through simulations of a single flow and then multiple
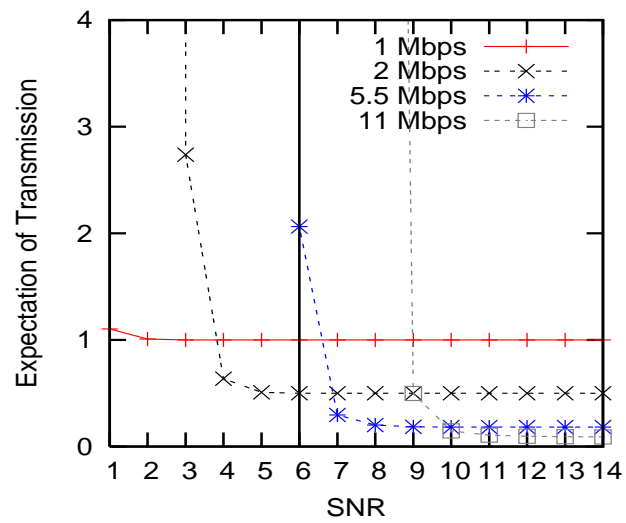
Figure 3.2: Expected Transmission

competing flows. All wireless stations are within a 500mx500m area that is covered by the signal of one access point located at the center.

We simulate the channel degradation with the two-ray ground fading model. The two-ray ground fading model [32] is a large scale fading model. According to this model, the signal received is composed of two components: the line-of-sight through the direct path and the wave reflected by ground. In general, the power of the received signal at a location is proportional to the exponent of the transmission distance, which follows:

$$P_r = P_t \Gamma \frac{1}{d^4} \tag{3.1}$$

where P_t is the power of the signal at the transmitter; Γ represents the antenna factors, such as the heights, the antenna gains; and d is the transmission distance. With the two-ray ground fading, SNR is constant at the receiver if both the receiver and the transmitter are static at some locations.

3.3.2   Data Rate Adaptation

At first, *LDRA* is compared with *ARF* to show *LDRA*'s ability to converge to a steady data rate. The first experiment consists of one mobile client node with one access point. The client node moves around an area such that 5.5 Mbps is the most appropriate data rate. Figure 3.3 plots the results: the x-axis represents the time and the y-axis is the data rate. Figure 3.3 shows, as expected, that *ARF* frequently changes the data rate because it blindly makes adjustments regardless of wireless channel conditions. Under the same conditions, the proposed scheme, *LDRA*, remains steady at the appropriate data rate of 5.5 Mbps.
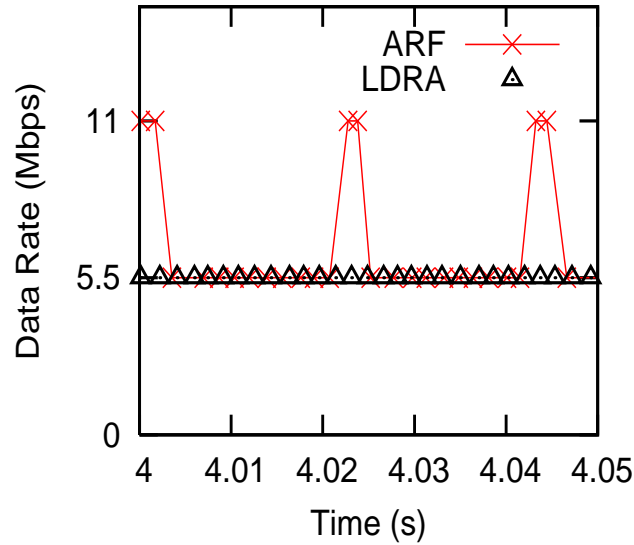
Figure 3.3: Rate Adaptation Comparison

3.3.3 Throughput Improvement

We define throughput improvement of *LDRA* over some scheme X as $\frac{Throughput_{LDRA} - Throughput_X}{Throughput_X}$. UDP CBR is used as the traffic in most experiments, except the last case testing TCP flows.

Figure 3.4 plots the throughput improvement over *ARF* in an experiment with varying the density of mobile nodes at different velocities in a WLAN network. The x-axis is for velocity and the y-axis represents the network throughput improvement with different node densities. As shown on the figure, although the velocity impacts throughput, it does not significantly impact the *improvement*. The improvement sharply increases with nodes density. This is due to *LDRA*'s ability to correctly distinguish a frame loss due to collision from that due to wireless channel fading. As collision increases in denser networks, *LDRA* exploits its ability to correctly diagnose collision losses and to maintain the original data rate. With *ARF*, the node unduly decreases the data rate. As collision increases, *LDRA* performs better than *ARF*, leading to a better throughput improvement.
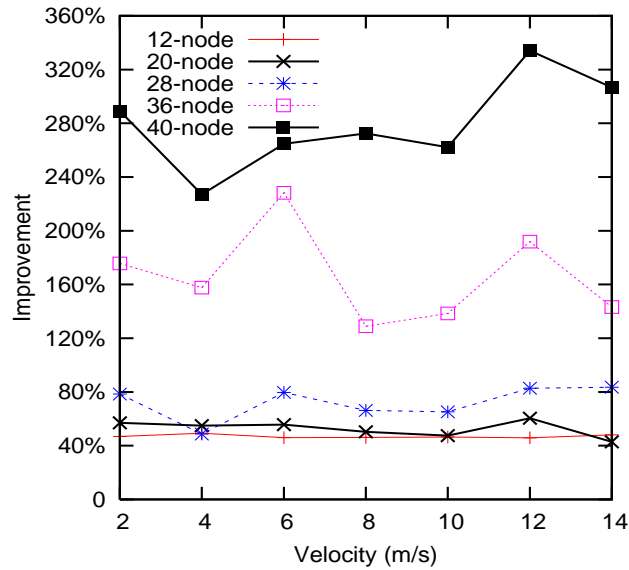
Figure 3.4: Throughput Improvement at Different Velocity

The relationship between throughput improvement and network density is even stronger from the simulation results depicted in Figure 3.5. In this scenario, the mobile nodes move at random velocity in an ad hoc network. The x-axis and y-axis respectively represent the number of mobile nodes in network and the throughput improvement.

We also compare the throughput from *LDRA* with that from work *AAR* by Pavon and Choi [5] in ad hoc network. Figure 3.6 illustrates the results. Note that the benefits from *LDRA* are more remarkable as the network density increases. In low network density, *AAR* scheme [5] performs better than *LDRA*, because *LDRA* spends time on the loss differentiation that is not so necessary with little composite loss. But in a high node density environment, *LDRA* outperforms it, due to the ability to differentiate losses.

Another set of experiments were carried with TCP flows. Different numbers of TCP flows are tested for *LDRA* and *ARF*. The total network throughput for all TCP flows is used to measure the improvement. The simulation results are plotted in Figure 3.7. The x-axis represents the number of
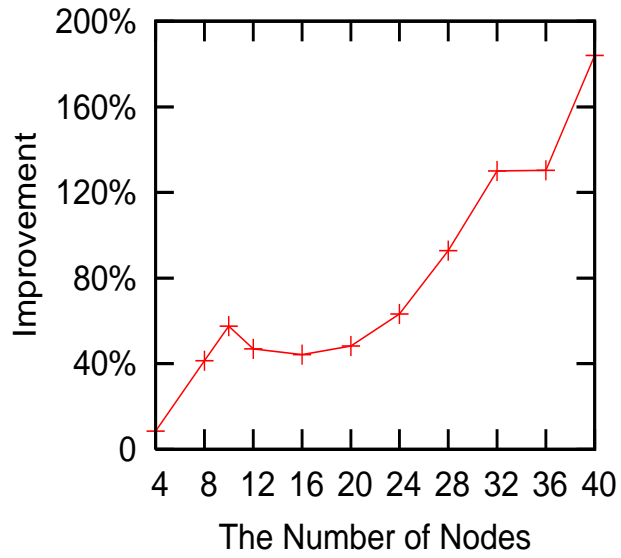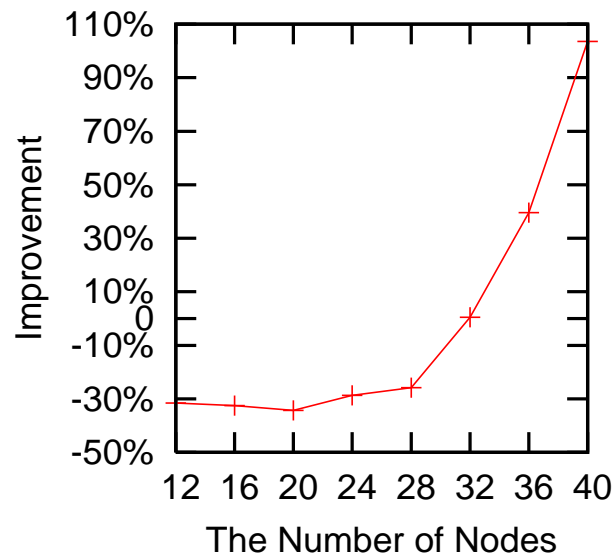
49

Figure 3.5: Improvement with Network Density

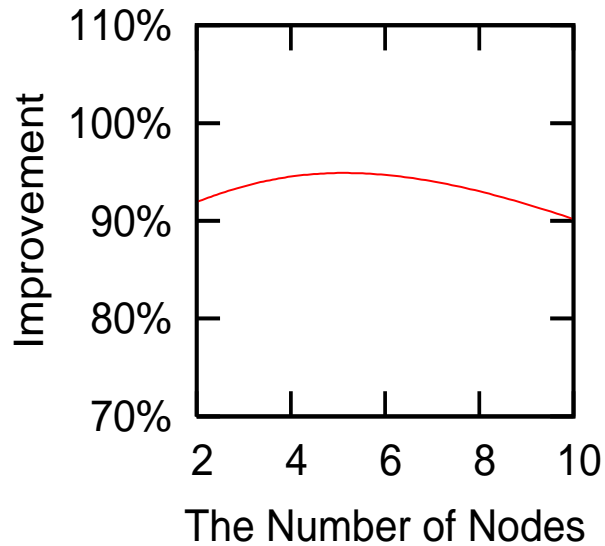Figure 3.6: Improvement with LDRA vs Adaptive

Figure 3.7: Throughput Improvement in TCP

TCP flows. The y-axis represents the overall network throughput improvement of *LDRA* over *ARF*.
This figure shows a dramatic improvement up to almost 100% for different scenarios.

3.3.4 Delay Jitter Improvement

Figure 3.8 shows a significant improvement of delay jitter in WLAN by *LDRA* over *ARF*. Delay
jitter is defined as the time difference in delay between two successive frames. We collect the maxi-
mum delay jitter for each scheme to compute the delay jitter improvement as $\frac{DelayJitter_{ARF}-DelayJitter_{LDRA}}{DelayJitter_{LDRA}}$.
The experiment involves one mobile client node and one access point with a UDP flow. The x-axis
and y-axis respectively represent the velocity and the delay jitter improvement. *Basic Rate Retrans-
mission* technique substantially contributes to this improvement because the number of retransmis-
sions is minimized by *LDRA*. *LDRA* induces less variability of delay after a frame loss and therefore
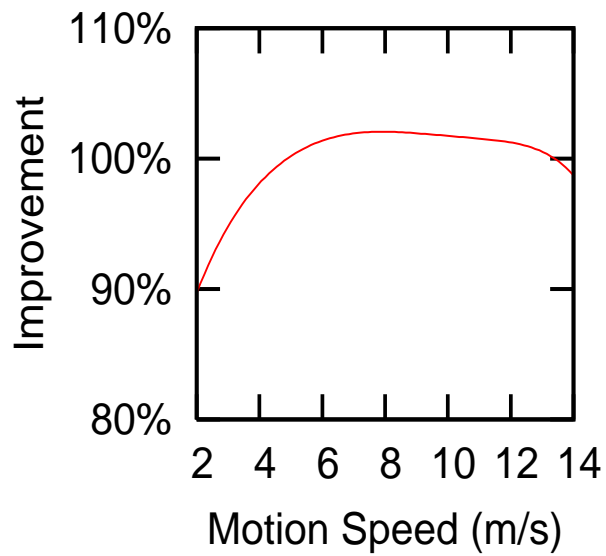yields a lower delay jitter than *ARF*.

Figure 3.8: Delay Jitter Improvement

3.4 Discussion

LDRA benefits from its loss diagnosis ability. It also recovers lost frames promptly with the basic rate retransmission technique. As *BARA* does, *LDRA* also relies on SNR for rate adaptation, which restricts its implementation to evaluate in practice. Therefore, an effective rate adaptation scheme based on frame loss is challenging but highly desirable to implement and evaluate in the real world for more sense in research.

CHAPTER 4

ERA: EFFECTIVE RATE ADAPTATION

To identify a reasonable, implementable, effective, and practical rate adaptation scheme with loss diagnosis, we investigate most recent rate adaptation schemes. As presented in Section 1.3.1, we identify their anomalies in the implementation. Based on these observations and design principles in Section 1.4, we design the scheme *Effective Rate Adaptation* (**ERA**).

In order to address the challenge of not using RTS/CTS frames in most scenarios, we take advantage of the fragmentation mechanism to assess the channel after a frame loss, to diagnose the cause of the loss, and to develop an efficient retransmission strategy for loss recovery. Also, to efficiently adapt rates in collision free environments, we introduce the binary backoff concept similar to that in *AARF*, but different in counting the number of successfully delivered data frames.

In this chapter, we first discuss the rationale to exploit fragmentation mechanism in *ERA*. Then we detail the design of this scheme. Afterwards, we partition the performance evaluation into two sections: the first section presents the results from simulation; the second section discusses the experiments on a Linux based tested including the implantation platform, the experimental environments and methodology, and the results from extensive experiments conducted on the testbed implemented with *ERA* and selected most recent rate adaptation schemes.

4.1   Rationale

This section details the rationale of using fragmentation technique to serve loss differentiation, fast recovery purpose in our proposal.
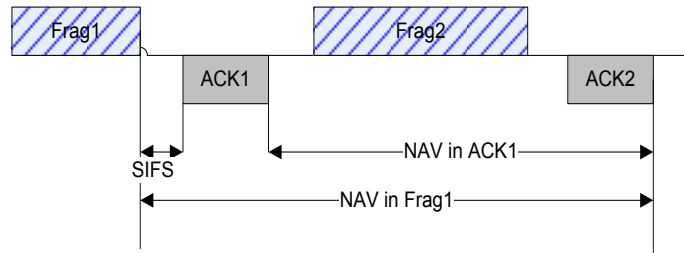
Figure 4.1: NAV of fragments in IEEE 802.11

4.1.1 Fragmentation in IEEE 802.11

Fragmentation is standardized in IEEE 802.11 that is *usually* used for packets too large to load in an MAC data frame or unfit for current channel conditions. In a collision dominated environment, a long data frame may yield poor performance, but fragmenting it into fragments of optimal size may dramatically improve performance, although the per-bit overhead is increased. Figure 4.1 illustrates the transmission of fragments stipulated in IEEE 802.11. Consider a sender station splits a frame F into two fragments $Frag_1$ and $Frag_2$. Fragment $Frag_1$ bears a network allocation vector (NAV) NAV_1 that reflects the time duration that the sender requires to complete the transmission of the second fragment $Frag_2$ (till the reception of ACK_2), therefore the full frame F. Due to the virtual carrier sense mechanism, all stations other than the receiver that hear the fragment $Frag_1$ will remain silent for duration NAV_1. The receiver acknowledges with ACK_1 that bears a new NAV also reserving the complete transmission of the second fragment. Such reservation completes till the completion of all fragments of the full frame F. Therefore, in an environment with collision, if the first fragment is successfully transmitted, the channel is cleared for all remaining fragments of that frame. Namely, the strength of fragmentation is that *the probability of collision for the whole frame is reduced to the probability of collision of its first fragment*.

4.1.2 Fragmentation and RTS/CTS

When a frame is fragmented into several pieces, each of them is needed to assemble with a MAC header. Therefore, fragmentation yields more per-bit overhead to data payload. At first glance, fragmentation appears to incur more overhead than RTS/CTS control frames because RTS/CTS are shorter than the MAC header. However, with detail investigation, this impression is not correct for following reasons. **First**, the lead fragment is transmitted at the same rate as a data frame, instead of the *basic* rate at which RTS/CTS are transmitted. The following example gives a more clear view. In IEEE 802.11g, RTS is sent at the basic rate 6 Mbps while the median rate is 24 Mbps. Therefore, the time to transmit a RTS (20 bytes) is equal to the time to transmit an 80-byte ($= 20 * 24 \div 6$) data frame at 24 Mbps. Deducting 34-byte of the MAC header and CRC, the time is still enough for a fragment carrying 56 bytes data payload. Another benefit of transmitting the short (lead) fragment at the data frame rate after a frame loss is discussed later in Section 4.2.2. **Second**, the transmission in IEEE 802.11 operates in time slots: namely, every transmission uses up a whole slot time even the transmission is over at the beginning of this time slot. Thus, when we calculate the transmission time, we should computer the number of time slots occupied, rather than the continuous nano-seconds (or even smaller time units). In 802.11g, a time slot lasts for 20 μs. A RTS plus PLCP header requires about 50 μs to be transmitted at the basic rate 6 Mbps in OFDM coding. The transmission time of a RTS is rounded up and equal to 3 slots, which is enough to transmit about 77 bytes MAC data payload plus the MAC layer header at the intermediate rate 24 Mbps. In conclusion, a short fragment transmitted at a normal rate often incurs less overhead than RTS/CTS.

4.1.3 Numerical Analysis of Fragmentation

To analytically demonstrate the strengths of fragmentation in a collision dominated environment, first we compute the probability for a frame to collide as a function of its length. Then, we will compute the ratio of the probability to collide for a whole frame over the probability for a fragmented frame. Consider an IEEE 802.11 network with M stations under saturated traffic: each station has a frame available to transmit all the time. Suppose all data frames are of equal length and the transmission of a frame occupies N time slots. Let p be the stationary probability that a station transmits a frame in a randomly chosen slot. p can be obtained using Bianchi's analysis [39]. Now, suppose a station is transmitting.

Let q denote the probability that no other station transmits in any generic slot (K) occupied by the frame being transmitted. q can be deducted as following:

$$q = (1 - p)^{M-1} \tag{4.1}$$

For the frame in transmission *not* to collide, all N slots occupied by the frame should not overlap with any transmission from other stations. Therefore, the probability that this frame is *not* in collision should be:

$$r = q^N = (1 - p)^{N(M-1)} \tag{4.2}$$

Then, the probability that this frame is collided can be calculated as:

$$P = 1 - r = 1 - (1 - p)^{N(M-1)} \tag{4.3}$$

When a frame is fragmented into two parts: one is very short, the other is the remainder. If the short fragment occupies S slots, the probability of this fragment to collide is $1 - (1-p)^{S(M-1)}$. As noted above, with fragmentation in IEEE 802.11, the probability of collision for the whole frame is reduced to the probability of collision of its first fragment. Therefore, the collision probability ratio for a whole frame over the same frame but fragmented is:

$$\frac{1 - (1-p)^{N(M-1)}}{1 - (1-p)^{S(M-1)}} \tag{4.4}$$

With above analysis, it is convenient to computer the numerical benefit of the fragmentation. Consider a WLAN of five contending stations ($M = 5$) and the data payload of a frame is one Kbyte. After fragmentation, the first (lead) short fragment carries 20 bytes. In IEEE 802.11g, a time slot is 20 μs. Taking into consideration the headers of PLCP, MAC, IP, and UDP, the data frame occupies 10, 20, and 74 slots respectively at transmission rates 54 Mbps, 24 Mbps, and 6 Mbps. Following the same calculation, the short fragment occupies 2, 3, and 7 slots, respectively. Based on these data and Equation 4.4, the collision probability *ratio* as a function of the transmission probability p is plotted in Figure 4.2. The y-axis represents the collision probability ratio and the x-axis is for the stationary probability p that a station transmits in a generic time slot. Bianchi [39] shows that the optimal throughput performance for a small M occurs at very small p. The binary exponential backoff mechanism in IEEE 802.11 does keep p small. As we can observe from Figure 4.2, when p is small (less than 0.15, shown in the inner magnified figure), the probability for a long frame to collide is several times that of a fragmented frame. We can conclude that the probability for a frame to collide in contention dominated environments is dramatically reduced by using fragmentation.
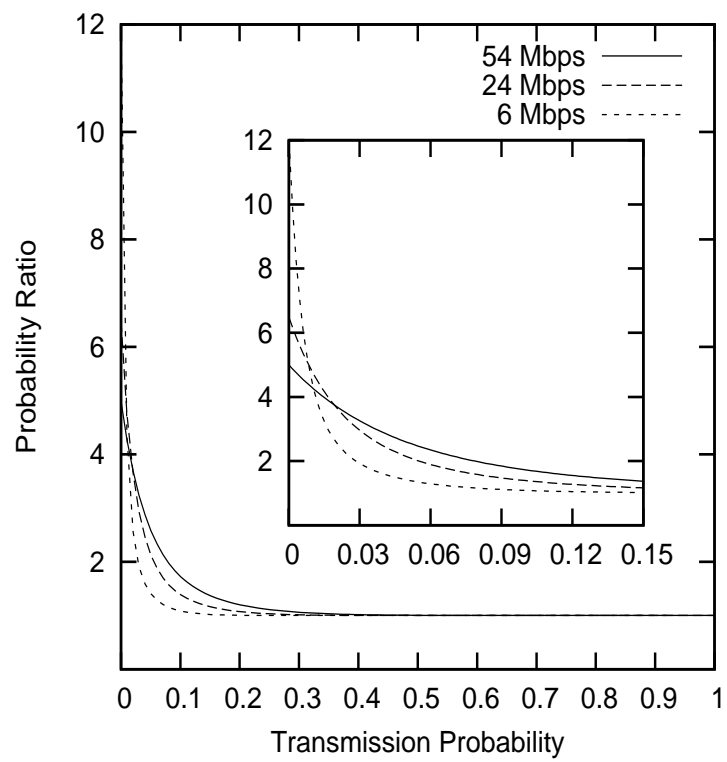
Figure 4.2: The Probability Ratio of Successful Frame Delivery

4.2 Design of ERA

ERA is purposely designed upon frame loss, since SNR and RSSI do not show strong correlation with channel conditions in complex environments. It is a "winning streak" scheme that counts the number of consecutive data frames transmitted successfully for rate increase to exploit potential better channel conditions. *ERA* accurately differentiates frame losses with fragmentation mechanism and promptly adjusts rate if a loss is diagnosed from channel degradation. In the following, we discuss each of these features.

4.2.1 Channel Assessment

One consideration deserving special effort for rate adaptation is the method to exploit potentially better channel conditions. Channel exploit for frame loss based schemes can only be achieved through rate increase. It is double-edged: 1) if the rate is increased more frequently than the channel condition upgrades, frame loss also increases as if it is from channel degradation; 2) if the rate is adjusted less frequently, the better channel condition is not fully exploited. It is impossible for a rate adjustment to keep the same pace as the channel condition varies. However, to achieve optimal network utilization, a rate adaptation should identify a delicate balance between these two "edges". *ERA* exploits the potential better channel condition by using the "winning streak" mechanism with an adaptive threshold. Before the detail of the rate increase in *ERA* is presented, a notion of *transmission counter* is introduced first for convenience in explanation.

The transmission counter Tr is used to assess the *signal stability* of the channel in communication. Tr used by the sender records the number of consecutive successfully transmitted *data* frames at some rate. Tr increments upon a successful transmission. It is reset to 0 for another rate under two scenarios: 1) a transmission failure is diagnosed due to channel fading; 2) the transmission
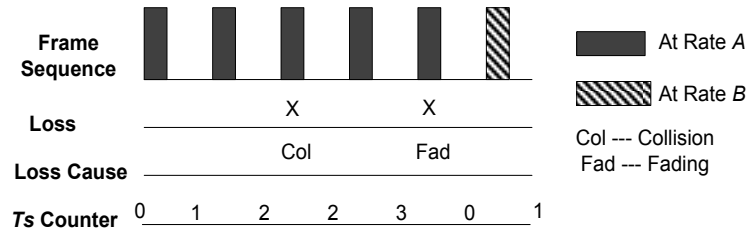
59

Figure 4.3: Consecutive Transmission Counting

succeeds after a rate increase with consideration that channel condition is good enough for higher rates. But Tr is not impacted by any diagnosed collision loss. The detail counting process of Tr is plotted in Figure 4.3. Let $Tr = 2$ after two consecutive successful transmissions at data rate R. Suppose that the third frame fails and the frame loss is diagnosed as a collision, then Tr remains 2. Furthermore, Tr is incremented after the third frame is recovered (as the forth) at rate R: the channel is still good enough at rate R! However, the fifth frame fails and the cause is diagnosed as from channel degradation, Tr is reset for the lower rate in a new counting cycle.

To balance the "double-edge" of rate increase, *ERA* introduces an adaptive rate increase threshold Ts, which is similar to that in *AARF*. In channel exploit, *ERA* increases the rate when the transmission counter Tr reaches the adaptive threshold Ts. Such an increase is based on the inference that, since there are Tr consecutive data frames transmitted successfully (excluding the collision) at current channel condition, the channel condition might be good enough to support higher rates. To achieve the balance, Ts should not be too high so that a sender can quickly reach the highest available rate and should not be too small to avoid rate oscillations and frequent losses due to overly optimistic increases. The initial value of Tr is set to 8 in our testbed. To reduce the frame loss in stable signal environment, the value of Tr is adaptively updated: if a transmission fails immediately after a rate increase, Ts is doubled, up to 32 maximum. At the same time, the rate falls back to the

prior rate with the assumption that the prior rate is already the highest that current channel condition can support.

4.2.2 Loss Diagnosis

In IEEE 802.11 wireless environments containing fast channel variation plus transmission collision, the frame loss is inevitable. But different type of frame loss can not be treated with the same solution. Therefor *ERA* achieves the loss diagnosis/differentiation with fragmentation mechanism. Suppose a frame loss occurs at rate r_{fail}. Let $r_{previous}$ be the rate of the last successful transmission before the current transmission failure. Two possible cases deserve consideration in loss differentiation.

1. $r_{fail} > r_{previous}$: This occurs only when the rate is increased from $r_{previous}$ to r_{fail} after Tr consecutive successful transmissions. The immediate transmission failure might indicate that the current channel condition can not support the new higher rate r_{fail}. Therefore, it is reasonable to conclude that the frame loss is likely due to overoptimistic rate increase to r_{fail}: the wireless channel condition can only support rate $r_{previous}$, but not rate r_{fail}.

2. $r_{fail} = r_{previous}$: In this case, previous transmissions succeeded at the same rate r_{fail}, but the current frame is lost. Therefore, the loss may be due to channel degradation as well as collision. Loss diagnosis requires more delicate strategy. We assume that the channel condition is fairly stable during the short loss *differentiation and recovery* process. This is reasonable because that time is shorter than channel variation.

   In order to further diagnose the loss, *ERA* splits the lost frame into two fragments by following the fragmentation technique in standard: a very short fragment and the other of the remainder. First, the short/lead frame is retransmitted at the *same* rate r_{fail}. The transmission

61

consequence directs the diagnosis. If the transmission is successful for this short fragment, then the loss was most likely due to a collision. This inference is reasonable for two reasons: 1) the probability of successful delivery is much higher for a smaller frame in a collision prone environment, 2) the probability of successful delivery is not significantly higher for a smaller frame in a degraded channel environment. Otherwise, if the transmission of the short fragment fails also, then the loss is likely due to channel degradation, but not certainly. Therefore, further differentiation is required. Then *ERA* halves the rate to retransmit the short fragment. For example, the rate is halved to 24 Mbps if r_{fail} was 48 Mbps. Such a retransmission with rate halving is repeated till the transmission succeeds, or the rate is decreased to the lowest rate. If a retransmission at any halved rate other than the lowest rate is successful, the loss is diagnosed from channel degradation. Otherwise, the loss is assumed due to severe collision. The complete diagnosis procedure is illustrated in Figure 4.4.

A key feature of *ERA* in diagnosing the cause of a loss consists of using the current rate r_{fail} to retransmit the short (lead) fragment rather than using the basic (lowest) rate immediately after the *original* failure [1]. This feature improves the accuracy and speed in the loss diagnosis in contrast with other schemes. Other schemes use short frames (e.g. RTS) sent at the *basic* rate immediately after the original failure. If the transmission at the basic rate is successful, it is equivocal if the transmission success is the result of the use of a *lower* (basic) rate (in channel degradation) or the use of a short frame (in collisions).

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[1]It refers to the first failure of a specific data frame, instead of the failure of any retransmission of this lost frame or its fragments.

4.2.3 Prompt Recovery

The prompt and efficient recovery of a frame loss in *ERA* benefits from is accurate frame loss diagnosis. The recovery has to consider three cases that result from loss differentiation.

- If the loss is diagnosed due to overoptimistic rate increase, namely the frame is lost immediately after rate increase, the rate falls back to the prior rate immediately and the rate increase threshold Ts is doubled at the same time.

- If the loss is diagnosed due to collision, the rate remains. But the short (lead) fragment from fragmentation combined with the binary backoff in the 802.11 standard should efficiently address the problem.

- If the loss is diagnosed due to channel degradation, the halving-rate technique can identify the appropriate new rate quickly for the degraded channel condition and recover the loss.

Unlike other rate adaptation schemes, *ERA* diagnoses the cause per loss: it differentiates the loss upon each frame loss. Once it diagnoses the loss is due to channel degradation, it adjusts the rate immediately, rather than waiting till some time later. This per loss diagnosis speeds up the response to frame loss, recovers lost frames quickly and therefore improves the network utilization. The complete and precise scheme of *ERA* is described in the flowchart in Figure 4.4.

4.3 Performance Evaluation on Simulation

We evaluate *EAR* on both network simulator *ns-2* and a implemented Linux testbed. This section only presents the result gathered from simulations.

Figure 4.4: *ERA* flowchart

4.3.1 Simulation Configuration

All experiments are restricted to WLAN infrastructure networks. Experiments were carried out with IEEE 802.11g MAC on simulator *ns-2* [36]. We refer the physical layer parameters to the Cisco Aironet 802.11a/b/g cardBus wireless LAN adapter [40]. Schemes *ARF, CARA* and our *ERA* were chosen and implemented in *ns-2* for comparison: *ARF* from the category without loss differentiation; *CARA* from the category with loss differentiation but requiring RTS/CTS; and *ERA* from the category with loss differentiation but without requiring RTS/CTS. The performance of these schemes is evaluated in two cases: a static mesh network environment and a mobile network. A set of experiments evaluated the impact of congestion level while another set evaluated the performance in a collision free environment. Most simulations were conducted with multiple wireless nodes. The *static* network topology of 16 client stations and one access point is shown in Figure 4.5. These stations are within the radio coverage of the access point, but they do not necessarily *hear* each other. Therefore the existence of hidden terminal depends on the network area size. To simulate a realistic environment, wireless channel fading is simulated with Ricean fading model [32] that takes into account both distance fading and time varying channel fluctuations. Consequently, such a simulation environment contains mixed losses from both channel fading and collisions. Another simulation scenario was carried out with one client and two-ray channel fading model.

The performances of *ARF*, *CARA*, and *ERA* are respectively evaluated and compared under different scenarios: strictly congested network (no losses due to channel degradation), strict channel degradation (collision free), and the more usual case where mixed losses may occur due to channel degradation as well as collisions.

Performance evaluation is conducted through extensive simulations using *ns-2* network simulator. IEEE 802.11g [26] is used as the MAC layer because it has more predefined rates. For
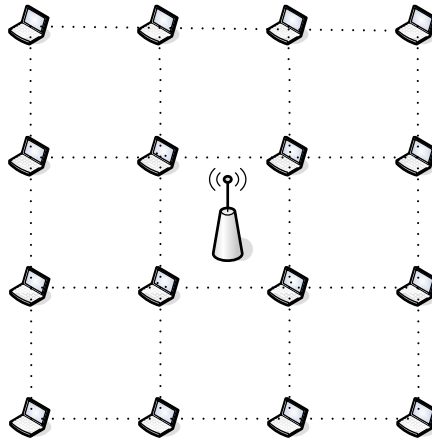
Figure 4.5: Topology of 17 static stations

Simulator	*ns-2* v.30
MAC	CSMA/CA DCF basic access
MAC data rates	6, 9, 12, 18, 24, 36, 48, 54 Mbps
Channel fading	Two-ray ground or Ricean
Ricean fading factor (K)	4
Traffic	UDP Constant Bit Rates (CBR)
Frame length	100, 500, 1000, 1500 Bytes
The short fragment length	Same as RTS

Table 4.1: Simulation parameters

Topology	Number of Stations	Motion Model
♯1	2	static
♯2	2	random movement
♯3	17	static
♯4	17	random movement

Table 4.2: Simulation topologies

the physical layer, we use the parameters of Cisco Aironet 802.11a/b/g cardBus wireless LAN adapter [40] and assume symmetric links. Table 4.1 summarizes the network configurations and parameters used in these simulations; and Table 4.2 describes the main four network topologies and the number of stations (including the access point) in each one. The traffic is always from the stations to the access point.

Each result figure plots the data averaged from multiple runs with random staring time in different evaluation environments. Each run of simulation is of 2-minute saturated UDP traffic. We evaluate *ERA* in three environments: channel degradation dominated, collision dominated, and composite (frame losses due to collisions as well as channel degradation). First, we present the experiments made in a composite environment.

4.3.2   Composite Environments: Collision and Channel Degradation

First, we evaluate the presumed strength of *ERA*, i.e., its effectiveness to promptly recover from losses. Then we measure the throughput (by Kframes/s) for the static network on Figure 4.5 and for a network of mobile stations.

ERA Retransmission/Recovery Effectiveness

We measure the effectiveness by collecting the fraction of frames losses recovered after only ONE retransmission. For example, if there are 100 losses and forty of them were recovered after

only one retransmission, then the fraction is 0.40. The experiments are conducted over a static network with 16 stations and one access point located at the center as shown on Figure 4.5. We use topology #3 (See Table 4.2). We adopt the Ricean fading model to induce dynamic channel conditions. By varying the size of the network layout area, we get a variable of loss ratio due to collision: the collision ratio is the number of frame losses due to collision over the total number of frame losses. If the collision ratio is null, this means that the environment is collision free (channel degradation dominated), which is emulated with only one client station. A strictly collision dominated environment has a collision ratio of one, which happens when all the stations are so close to the access point that there is no transmission failure due to channel fading even with Ricean model. Figure 4.6 plots the fraction of recovered losses after only one retransmission on the y-axis with the collision ratio on the x-axis. Under most conditions, *ERA* succeeds in recovering 80% of the losses with only one retransmission of the whole lost frame and outperforms *CARA* under all circumstances. In a channel degradation dominated environment (close to null collision ratio), *CARA* does not recover quickly because it often addresses the loss as a collision.

For the same experiment, we also compute the mean of the number of retransmissions plotted on the y-axis of Figure 4.7. The strength of *ERA* stems from its refined mechanism to diagnose the cause of a frame loss and adopt the best strategy to recover. But for *CARA*, the success of RTS/CTS at the *basic* rate can not indicate channel fading or collision.

Throughput for Static Networks

To evaluate *ARF*, *CARA*, and *ERA* under realistic channel conditions with both channel degradation and collision, we use the Ricean fading model on a static network of 16 stations (see Figure 4.5) and vary the area size to adjust the congestion level. We use topology #3 (See Table 4.2).
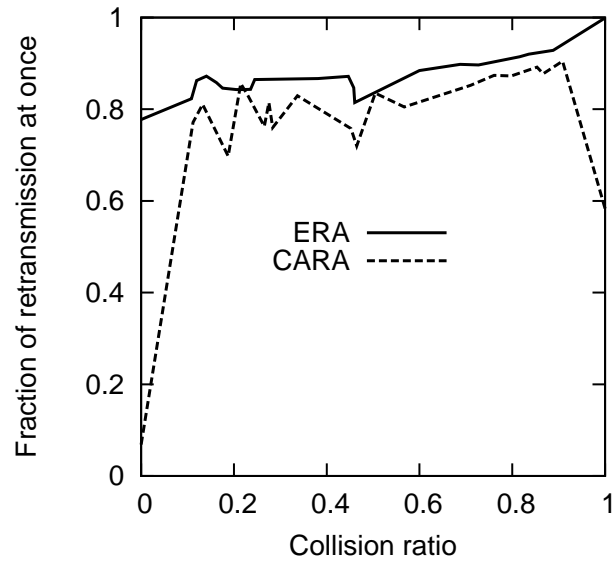
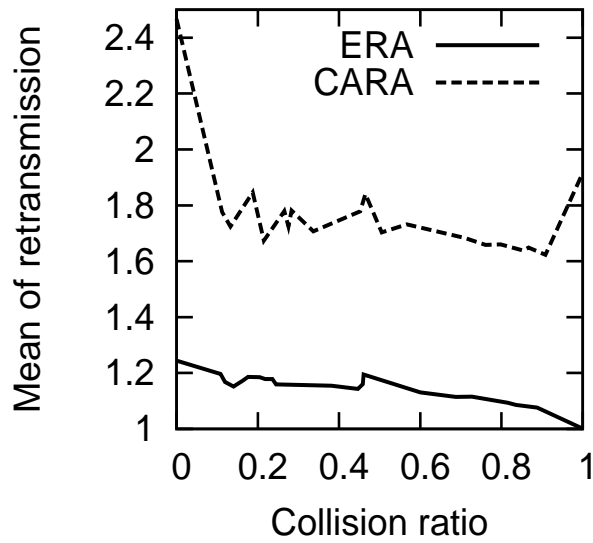Figure 4.6: Fraction of recovered losses with ONE retransmission

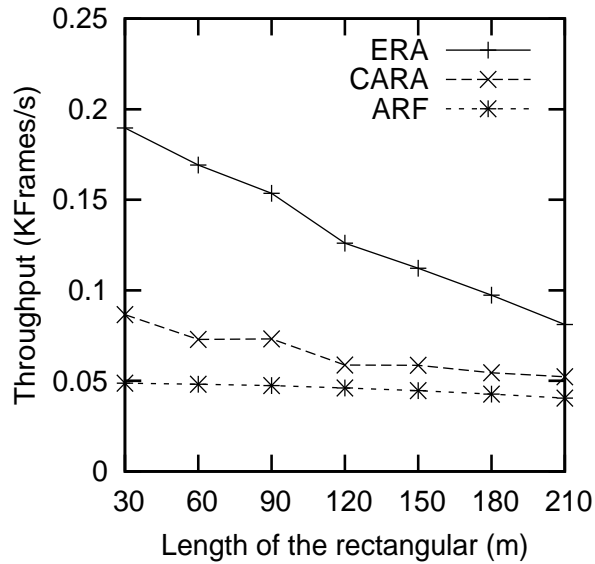Figure 4.7: The mean of retransmission

69

Figure 4.8: Throughput under both channel degradation and collision

CBR traffic with 1000 bytes packets is sent from the 16 stations to the access point. Figure 4.8 plots on the y-axis the throughput achieved by *ARF*, *CARA*, and *ERA*, respectively. As we observe, *ERA* performs better in heavily congested environments (in smaller areas). This is because *ERA* implicitly assumes the loss is due to collision at first (by transmitting the short fragment at the current rate). Consequently, it favors the loss differentiation in collision dominated environments.

Throughput for Mobile Nodes

The throughputs of *ARF*, *CARA*, and *ERA* are respectively evaluated on a mobile network of 17 stations: all 16 client stations are *mobile* at random velocities to random destinations with short pause between two consecutive movements in a 350mX350m area. The access point, located at the center of the network , is static. We use topology #4 (See Table 4.2). The Ricean fading model is used to simulate channel degradation. Table 4.3 shows the parameters used in these simulations. All stations keep transmitting to the access point UDP CBR traffic.

parameter	value
Network Area	350mX350m
Location of Access Point	(175m, 175m)
Fading	Ricean
Velocity	2 m/s to 30 m/s
Pause Time	1 second
Client Stations	16

Table 4.3: Network parameters for mobile network simulations
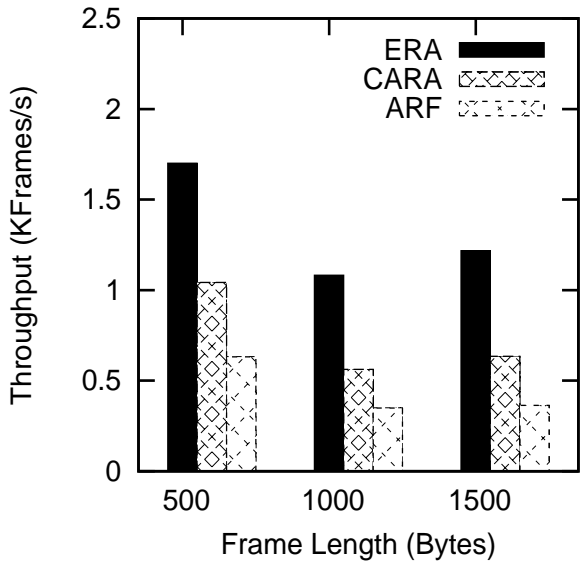

Figure 4.9: Throughput of mobile stations

The first experiment evaluates the throughput for all schemes with a CBR traffic with different packet lengths: 500 bytes, 1000 bytes and 1500 bytes. Figure 4.9 plots the throughput of *ARF*, *CARA*, and *ERA* for different packet lengths. The horizontal axis represents the packet length used. With longer frames, there are more collisions, therefore *ERA* has more opportunities to deploy its strength.

We also collect the adaptation transient dynamics for *ARF*, *CARA*, and *ERA* for the mobile stations. Figure 4.10 displays three plots for *ARF*, *CARA*, and *ERA*. Each plot corresponds to a
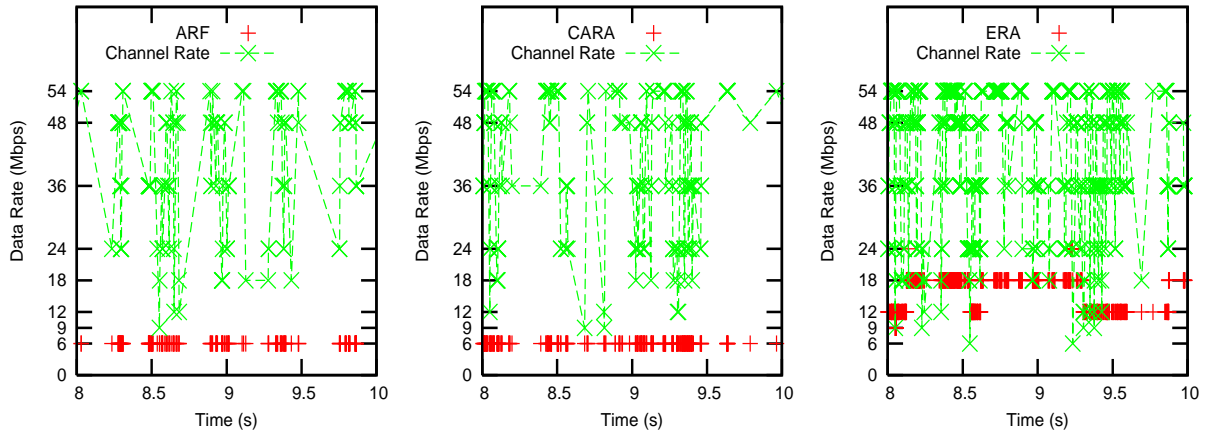
Figure 4.10: Adaptation dynamics

two-second period of dynamics for a randomly chosen mobile client station. The x-axis is the time. The y-axis shows the data rate in Mbps. Each plot presents two curves: the first curve labeled "Channel Rate" (symbol $\times$) is the rate that the channel can support at that instant (assuming perfect knowledge) and the second curve (symbol $+$) is the rate selected by *ARF*, *CARA*, or *ERA*. We observe that in such conditions (mixed environment with both collision and channel degradation), 1) the volatile nature of the channel makes it difficult for any rate adaptation scheme to select the optimal supported rate 2) the plot for *ERA* has more points (which means more frame transmissions) than the two others (*ARF* and *CARA*) because *ERA* results in less congestion loss, and 3) *ERA* selects a rate that is closer to the optimal achievable rate. *CARA* outperforms *ARF* mainly due to its ability to reduce collision with RTS frames.

With above extensive simulations on different network topologies and configurations, *ERA* robustly exhibits dramatic performance improvement. These simulation results illustrate *ERA*'s effectiveness in composite environments (channel degradation and collisions). In order to understand better *ERA*, we evaluated it separately in a strictly channel degradation dominated environment and then in a strictly collisions dominated environment.

4.3.3 Channel Degradation Dominated Environment

For these experiments, we use topology #1 (See Table 4.2) with one **static** station at a constant distance from an access point. First, we evaluate the throughput performance throughout the experiments of *ARF*, *CARA*, and *ERA* in stable channel environments. Second, we measure the throughput for these three schemes in an environment with dynamic channel conditions simulated with the more general Ricean fading model. Finally, we measure the impact of packet length on the throughput of those three schemes for a station at 300m from an access point. Due to the space limitation, we only present the result gathered in the stable environment and leave the result of the other two scenarios in our technical report [41].

Stable Channel Conditions

In general, a user sits at a desk and has a long work session where a given constant rate will be most often supported. We somewhat create such environment with only one fixed station and one access point. The two-ray ground fading model is used to ensure that a given constant rate is supported: the supported rate under a two-ray ground fading model depends only on the communication distance. Figure 4.11 plots on the y-axis the throughput achieved by *ARF*, *CARA*, and *ERA* respectively when the client is at a constant distance (x-axis) from the access point. *ERA* outperforms the other schemes due to 1) the effectiveness of its recovery strategy and 2) the mechanism inherited from *AARF*: adaptively adjusting the threshold number ($Threshold$) of consecutive successful transmissions required to increase the rate.
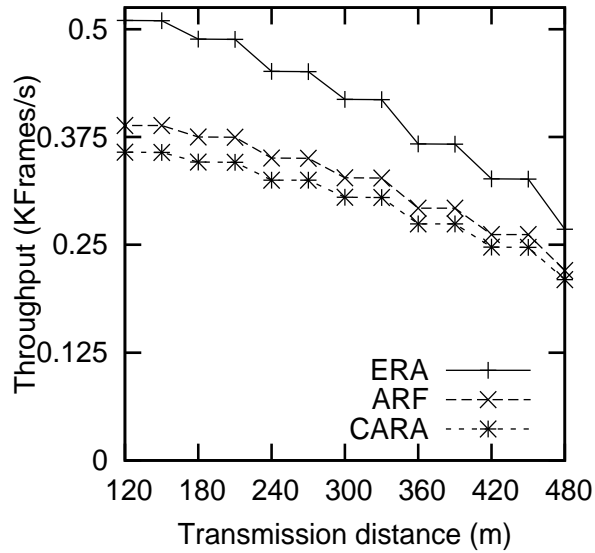
Figure 4.11: Throughput under stable channel conditions

4.3.4 Collisions Dominated Environment

For this environment, all simulations are carried out in a network with 17 *static* stations corresponding to topology #3 (See Table 4.2): one access point and 16 client stations. The objective is to evaluate the performance of *ERA* and compare it to *ARF* and *CARA* performances in a collision dominated environment. We use the two-ray ground fading model such that the best supported rate is constant (stable channel conditions): with the two-ray ground fading model, the quality of the channel is determined only by the distance. All 16 client stations are uniformly placed in a rectangular area as shown in Figure 4.5. The congestion level in the network is varied by adjusting the size of the network layout rectangle. All stations keep sending 10 Mbps CBR traffic to the access point at the center of the rectangular area. We measure the throughput and evaluate the impact of the packet length on the performance.
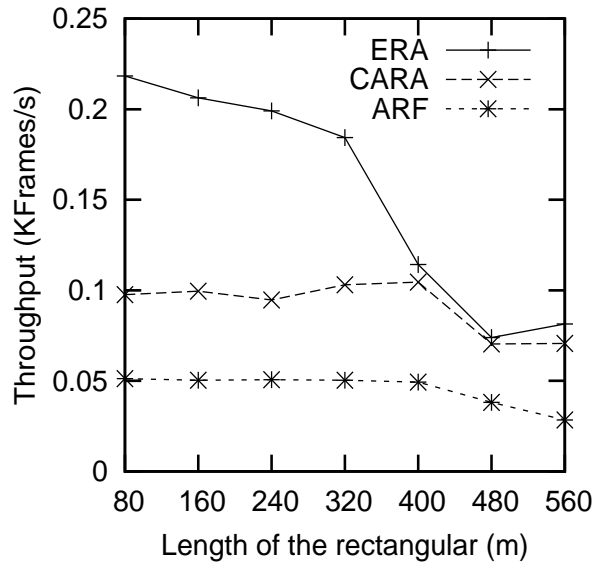
Figure 4.12: Throughput under different levels of collision

Impact of the Congestion Level

Figure 4.12 plots on the y-axis the throughputs of *ARF*, *CARA*, and *ERA*, respectively. The x-axis represents the length of the rectangle. The packet size is set to 1000 bytes.

ERA performs best again, followed by *CARA*, and then *ARF*. Especially, as the collision increases (smaller rectangular area), more improvement is achieved by *ERA* due to a more accurate loss differentiation and to the judicious recovery strategy.

Impact of Packet Length on *ERA*

In each simulation, each station sends CBR traffic with packets of 100 bytes, 500 bytes, 1000 bytes, or 1500 bytes. All stations send packets of the same size in each run. Figure 4.13 plots the percentage throughput improvement of *ERA* over *CARA* ($100 * \frac{ERA - CARA}{CARA}$). When the frame size increases, the improvement increases because collisions are more likely with longer frames. The
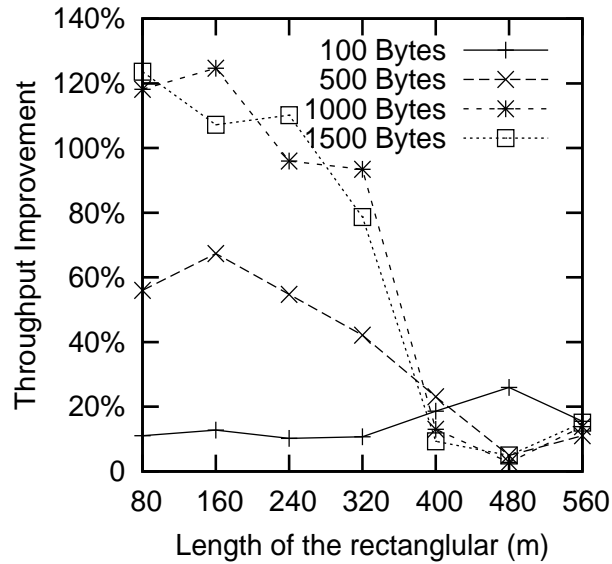
Figure 4.13: Impact of Packet length on throughput improvement

improvement is more significant for smaller rectangles because there are more collisions and *ERA* has more opportunity to take advantage of its superiority.

4.4 Performance Evaluation on Linux based Testbed

In addition to above simulation experiments, we also implement and evaluate *ERA* and selected presentative schemes on a Linux based testbed. Because SNR and RSSI have been studied to be poor channel conditions indicators, SNR/RSSI based rate adaptation scheme are not considered for implementation in our testbed. Moreover, some of these schemes are not IEEE 802.11 compliant and just cannot be implemented on off-the-shelf IEEE 802.11 network interfaces. Therefore, we only implement for performance evaluation the representative schemes that use frame loss as channel condition indicator. After investigation, *AARF, CARA, RRAA* and *SampleRate* are selected to respectively represent *Non Loss Differentiation-Winning Streak*, *Loss Differentiation-Winning Streak*, *Loss Differentiation-Statistics*, *Non Loss Differentiation-Statistics*, which are highlighted in

76

Table 1.1. These representative schemes together with our proposed scheme *ERA* are implemented in the Linux based testbed.

This section presents the implementation details of the Linux based testbed. Then experimental environments and methodology are discussed. Finally the results from this testbed are presented.

4.4.1   Implementation Platform and Architecture

Our testbed consists of one wireless router and six laptop clients. Considering the open architecture, we implement rate adaptation schemes in clients. Each laptop client is equipped with a mini-PCI 802.11 adaptor with Atheros R5212 chipset. This adaptor is chosen for some of its attractive features. First, Atheros R5212 chipset supports 802.11 a/b/g modes. It is convenient for us to tune to any 802.11 mode in our experiments without extra hardware support. Second, also most important, there is an open source driver comprehensively supporting adaptors with Atheros chipset in Linux, which is downloadable from MadWifi [42]. This driver is delicately designed and layered in architecture to take in new development. Third, the MadWifi driver is dissected into different modules: rate adaptation control is developed as an individual module. This architecture speeds up the implementation of a new rate adaptation scheme. In addition, the MadWifi driver provides an excellent mechanism to access the communication statistics through a delicate interface to the firmware on the chipset. The communication statistics (e.g. SNR, RTS/CTS rate, transmission result and retries) are accessible and controllable with minimum delay. The transmission queues in driver can also be manipulated for per-frame operation. The rough architecture of this open source driver is depicted in Figure 4.14. As shown in the figure, the MadWifi driver consists of four components. Among them, *net802.11* provides generic IEEE 802.11 services such as frame assembly/disassembly, user authentication and data encryption. *HAL* (Hardware Abstraction
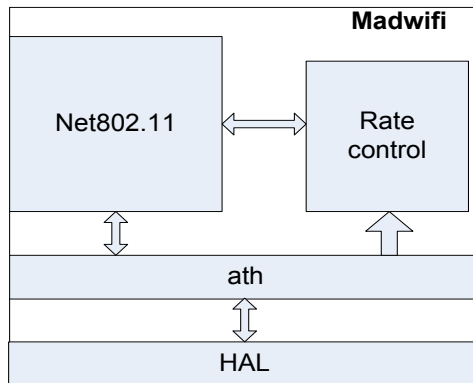
Figure 4.14: MadWifi Driver Architecture

Layer) provides access for the other components to the hardware firmware. *HAL* acts as an abstract application interface because it is a closed source package with binary only format maintained by Atheros Inc. The third component, *ath*, consists of specific callbacks for *net80211* and *rate control* through the interfaces supported by by *HAL*. Therefore, *ath* behaves more like an encapsulation of *HAL* to provide firmware access. The component *Rate control* is responsible for executing rate adaptation algorithm to select the transmission rate for each data frame. We implemented rate adaptation schemes in *Rate control* as individual modules. Our implementation is carried out on Linux kernel version 2.6.18 and MadWifi driver version 0.9.3. *AARF*, *CARA*, *RRAA*, and *ERA* are implemented as separate rate control modules. *SampleRate* [11] already exists in the MadWifi driver, but by default it deliveries the control of retransmission to firmware, it has a clear advantage over the other schemes. Therefore, for fairness in comparison, we turned off this capability for *SampleRate*: *SampleRate* is modified to control the retransmissions in the driver as the other schemes do.

4.4.2   Implementation of ERA in MadWifi

In our testbed, to be compatible with the architecture of MadWifi, *ERA* is implemented as an individual rate control module. *ERA* module consists of three primary components: rate adjustment,
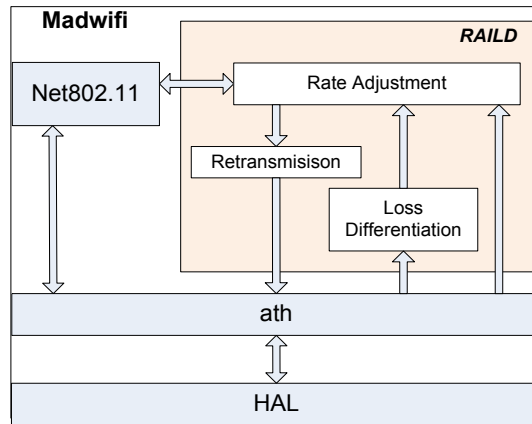
Figure 4.15: Architecture of *ERA* in MadWifi

retransmission and loss differentiation. The dependency and interaction of these components are illustrated in Figure 4.15. For each transmission, *ERA* detects the consequence (failure or success) of the transmission from the callbacks in module *ath*. If the transmission succeeds, the rate adjustment component updates the transmission counter Ts and increases the rate if the condition is satisfied. In case of failure, the loss differentiation component takes over the control. It is charge of fragmenting the lost frame, handing over the fragments to retransmit and diagnosing the cause of a loss. The result of loss differentiation is output to the rate adjustment component to decrease the rate in case of channel degradation. The retransmission component is responsible for queueing corresponding fragments of the failed frame into the transmission queue and assigning the short fragment a rate by consulting the rate adjustment. Since we do not have control over the transmission buffer in the firmware, we turn off the retransmission functionality for each data frame in firmware provided by *ath*. Instead, we retransmit in the driver through the transmission queue.

4.4.3 Experimental Environment and Methodology

We conduct experiments in both controlled tests and field public site tests. The controlled experiments are carried out in a building called Shop Building whose floor plan is depicted in Figure 4.16. The experimental building consists of classrooms, offices and labs. But all office furniture and lab equipment have been moved out as the department relocated. To minimize the external surrounding interference, we conduct all experiments in the evening after all classes dismissed when nobody was walking around during the experiments.

Our testbed consists of one access point and six laptops. The access point is a Belkin wireless G router with compatibility to 802.11b. All client laptops are IBM Thinkpad T60 whose configuration is CPU Intel Core Duo 1.8Ghz, RAM 1G, HD 80G and mini-PCI WIFI adaptor with Atheros R5212 chipset. All rate adaptation schemes are implemented in the laptops so that any scheme can be selected at any time for performance evaluation. The access point is closed to hack and runs "AS IS" with its factory settings. All data traffic is generated from the laptops to the access point. The access point marked with *AP* in Figure 4.16 is located in the upper rightmost classroom. Those laptops *generating* traffic for trace collection are placed at locations (*L-1, L-2, L-3* and *L-4*). *Interfering* laptops are placed at locations (*I-1, I-2, I-3* and *I-4*).

Both laptops and the access point run in IEEE 802.11b/g compatible mode. Therefore, the available rates are 1, 2, 5.5, 11, 6, 9, 12, 18, 24, 36, 48, and 54 Mbps in experiments.The access point supports 11 channels (US standard). There are three non-overlapping channels (1, 6, 11) available in IEEE 802.11 networks. To reduce co-channel interference, these channels are normally deployed to neighboring access points in campus networks. In the experimental building, before the experiments, we sniffed the channels that are in use by other surrounding networks. Channel 6 and 11 are heavily used by more than ten surrounding access points. The signal from these networks is
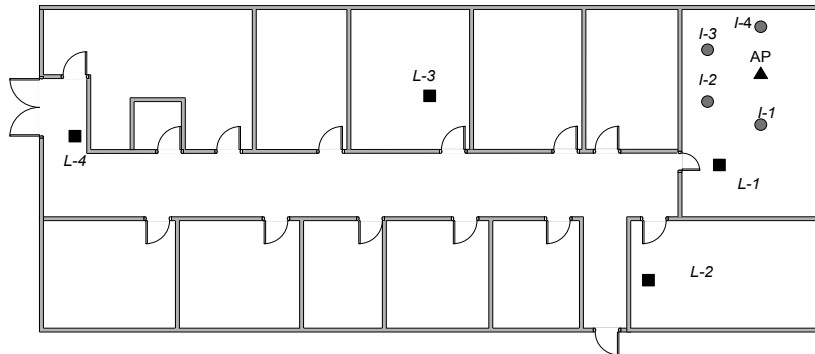
Figure 4.16: Floor plan in experiments

strong and has a large impact on the result if we tune our testbed to these channels. Comparably, channel 1 is lightly used by only two networks. Therefore, to minimize the interference from other networks, we tuned the testbed to channel 3 because it is not used by any other close-by network. Note however that Channel 3 is slightly impacted by networks running on channel 1 and channel 6.

Besides the controlled experiments in Shop Building, we also carried out field tests in public areas. Laptops with implemented schemes *AARF, CARA, RRAA, SampleRate* and *ERA* are connected to some public access point of the campus network. The traffic generated by client laptops are transmitted to a wired receiver. These transmissions contend for channel access against other anonymous wireless stations connected to the same access point.

Each experiment lasts for 2 minutes for two considerations. First, the time is long enough to cover time variations of 802.11 wireless signal. Second, it is short so that the channel background environment does not change wildly for interleaved measurement for each scheme in each run. For each experiment scenario, five runs are repeated. And for each run, tests for all schemes are interleaved such that they are evaluated under the *same* conditions (as much as we can). Then, in gathered trace, the maximum and the minimum values of the five runs are ignored to avoid wild variation. The remaining three values are averaged as the final result. In all our experiments, the

RTS threshold is set to the maximum to turn off RTS completely for all sizes of frame. UDP and TCP traffic are used as data load. Two types of UDP traffic are generated with *Mgen* [43] at 16 Mbps (1000 packets of 2 KBytes per second) or 48 Mbps (2000 packets of 3 KBytes per second). TCP traffic is generated with *Iperf* [44]. The receiving TCP window size is set to the default value (16 Kbytes) (which is not allowed to change in the Linux we used).

To comprehensively compare *ERA* and other selected rate adaptation schemes, we mainly measure their performance in the following scenarios:

- **Channel degradation dominated**. In this scenario, only one client runs in an 802.11 cell. This scenario emulates a wireless home network. Experiments are conducted for both stationary and mobile scenarios. For the static case, the performance is evaluated at each location *L-i*. Even static transmissions suffer from frame loss due to time-variant channel fluctuations [32]. For the mobile case, one client laptop with traffic to the access point is carried by a person walking at an almost constant speed from *L-1* to *L-4* through all four locations. Mobility introduces more complexity to channel degradation [32].

- **Composite lossy environment**. For this scenario, the communication experiences frame loss from both channel fading and collision. We place two contending clients at interfering locations *I-1* and *I-2*. These clients continuously generate data traffic to the access point. Such a configuration attempts to reproduce an office or campus network. Again, both static and mobile experiments are conducted as we did for the channel degradation dominated environment.

- **Heterogenous rate adaptation schemes**. In this scenario, each of four laptops runs one of the rate adaptation schemes *CARA, ERA, RRAA* and *SampleRate*, respectively. The objective

is to observe how these schemes contend with each other. *AARF* is not taken into evaluation in this scenario because it performs poorly in congestion dominated environments.

- **Increasing contention level**. The contention level is gradually increased by running one more laptop each time (incrementally) at interfering locations *I-1, I-2, I-3* and *I-4*. The rate adaptation schemes experience different levels of collision generated by two, three, or four laptops.

4.4.4 Experiment Results from Testbed

With the testbed of implemented rate adaptation schemes: *AARF, CARA, RRAA, SampleRate* and our *ERA*, extensive experiments are carried out to evaluate their performance. Our experimental setting is as presented in Section 4.4.3. Performance is evaluated under a combination of scenarios described in Section 4.4.3. Both static and mobile scenarios are considered. We conduct both controlled experiments and public field tests in the real world where all conditions are out of our control. The testbed in controlled testes is set as only one WLAN: all stations are within the radio coverage of the same access point. But note that these client stations are not necessarily within the radio coverage of each other, leading to possible hidden terminal collisions. The traffic is always from the clients to the access point. Unless particularly specified, the y-axis in all the following figures represents the goodput: data successfully collected at the receiver station per second. As we may observe from the following experiment results, in most cases, *ERA* outperforms the other schemes. The results are presented in groups of experimental scenario.
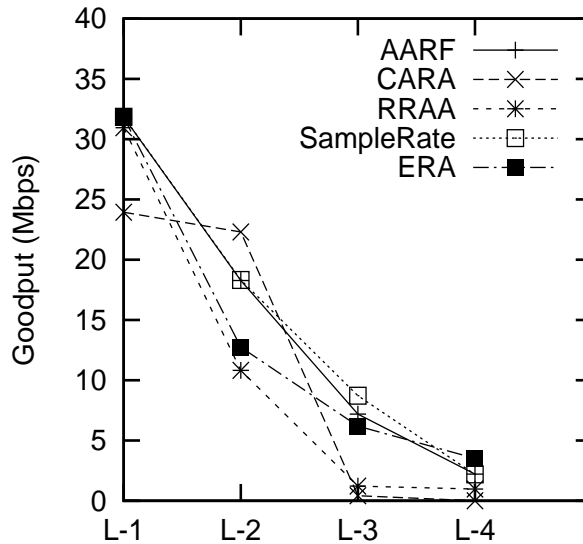
Figure 4.17: UDP goodput in static mode in channel degradation

Channel Degradation Dominated

We first evaluate the performance of these rate adaptation schemes in channel degradation dominated environments. Such environments are widely visible in residential wireless home networks where only one wireless client at a time is accessing the Internet through the access point. The client may move around, but more often be stationary. Therefore, in channel degradation environments, we conduct experiments for both static and mobile modes.

In the static mode, a client laptop is successively placed at each evaluation location (*L-1, L-2, L-3, L-4* in Figure 4.16). Experimental trace is collected at one location at a time. UDP traffic generated by the client is transmitted towards a wired receiver through the access point. Figure 4.17 depicts the goodput of each scheme in the static mode. The x-axis represents locations (*L-1* through *L-4*). We observe that the performance of each scheme may vary widely at each location. *ERA* performs best at *L-1* and *L-4*. Particularly at *L-4*, *ERA* improves goodput by about 70% over the runner

up *SampleRate*. At these two spots, the signal strength is either best or worst, therefore supporting very stable rates. *L-1* can support highest rate all the time and *L-1* can only support the lowest rate. Without collision, in locations with stable rates, *ERA* outstands with benefits from its adaptive threshold for rate increase. It incurs fewer frame losses, thereby less overhead from fragmentation and less resource consumption in retransmission. But, *ERA* does not perform best at *L-2, L-3*. Since all losses are due to channel variation in this environment, after a frame loss, the first transmission of the lead (short) fragment in *ERA* at the normal rate most likely fails, therefore causing an overhead to each retransmission. *SampleRate* samples other rates at every tenth transmission, constantly wasting the resource roughly 10%. Observe that *RRAA* and *CARA* perform poorly at most locations because they do not explicitly take prompt actions for degradation even they detects it. *CARA* almost achieves a null goodput at *L-4* (weakest signal strength). It is also not surprising that traditional schemes *SampleRate* and *AARF* have better performance because they primarily target channel degradation environments without explicit consideration of collision.

In mobile experiment mode, a client laptop is carried by a person moving at an almost constant pace through over the path *L-1 → L-2 → L-3 → L-4* shown in Figure 4.16. The traffic pattern is the same as in above static mode. The experimental results are plotted in Figure 4.18. We can observe that *ERA* achieves the best goodput in this scenario. Again, *RRAA* and *CARA* are outperformed because they do not address well the channel variation explicitly.

Composite Lossy Environments

Then we evaluate the performance in environments with mixed frame losses due to channel variations and collisions. Such mixed environments are similar to campus or corporate environments where each access point is associated with multiple clients and more than one wireless client
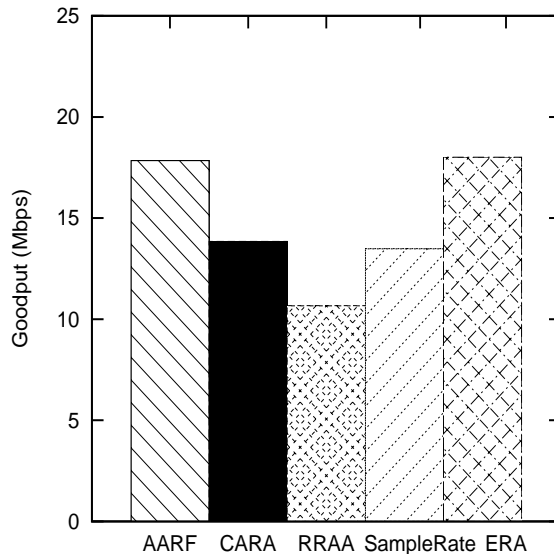
Figure 4.18: UDP goodput in mobile mode in channel degradation

may well transmit to the access point in a generic time slot, especially when they are hidden terminals to each other. To introduce the collision into the environment, we place two interfering wireless stations statically at location *I-1* and *I-2* in Figure 4.16. These two stations continually transmit traffic at a constant pattern also to the wired receiving node through access point over all the experiment time. Both static and mobile scenarios are evaluated.

The mobile scenario still adopts the route *L-1* → *L-2* → *L-3* → *L-4*. Only UDP traffic is evaluated in this case. But, the traffic has to contend with the interfering traffic from those stations located at *I-i*. The results are presented in Figure 4.19. Although without using RTS/CTS, *ERA* improve the goodput about 25% over *RRAA*, the second best scheme in this scenario. The strength of *ERA* is that it not only diffuses collisions, but also it can react to channel degradation. Its per frame loss diagnose also prompts the loss recovery. Because of their ability to suppress collisions with RTS/CTS after loss, *RRAA* and *CARA* outperform traditional schemes in the combinational loss environments.
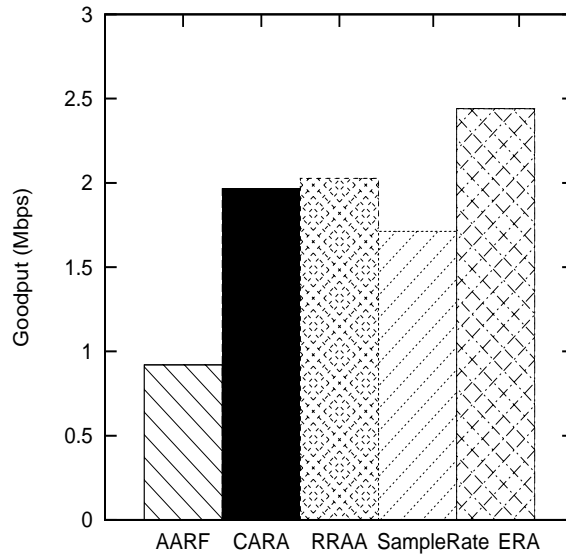
86

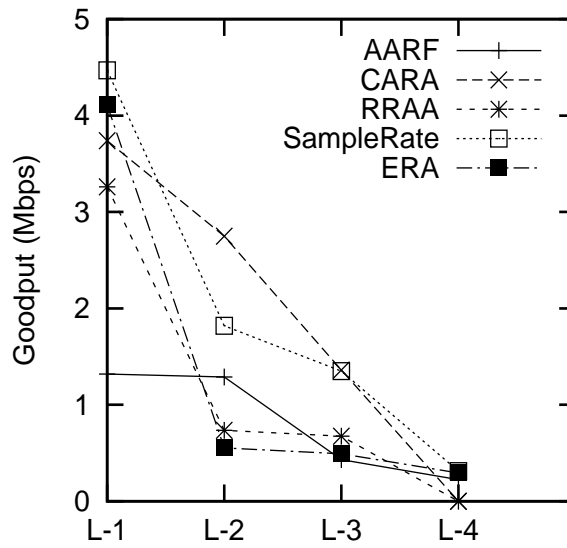Figure 4.19: UDP goodput in mobile mode in combinational loss

Figure 4.20: TCP goodput in static mode in combinational loss

87

In the static test, a client laptop running with implemented schemes is successively placed at each location (*L-1, L-2, L-3, L-4* in Figure 4.16. Measurements are made at one location at a time. We measure the TCP traffic in this scenario. The collected goodput results are plotted in Figure 4.20. The x-axis represents locations *L-1* through *L-4*. From the figure, the performance for each scheme fluctuates wildly at different locations in such mixed loss setting. But, one surprising observation is that *SampleRate* does not perform poorly as reported in the literature. After investigation, we find that the *SampleRate* implemented in the MadWifi driver probes (samples) all rates except the current one. This behavior is different from the behavior described in the paper [11] that only probes the rate "that may do better than the current one". *SampleRate* implemented in the code probes also all the rates lower than the current one at every tenth transmission. In collision environments, the probing at lower rates likely increase the transmission failure due to more collision with bigger effective transmission range. Therefore, *SampleRate* is aware that lower rates can not have better performance than the current rate in use in the transmission window. It remains its current rate or even increases rate for better throughput depending on the sampling result. Such operation is most appropriate in a collision environment. Implicitly, it somehow has the ability not to decrease rate in collision prone case while it does not have any explicit differentiation measure.

Multimedia Parameters

As multimedia services become popular in mobile applications, the capability of a WLAN network supporting the multimedia payload transportation is critical. We conduct experiments to evaluate metrics of interest to multimedia applications: jitter and the percentage of out-of-order packets. The experimental setting is as in a static loss environment. In these experiments, we use *Iperf* [44] to generate VoIP UDP traffic and gather the jitter and the number of out-of-order
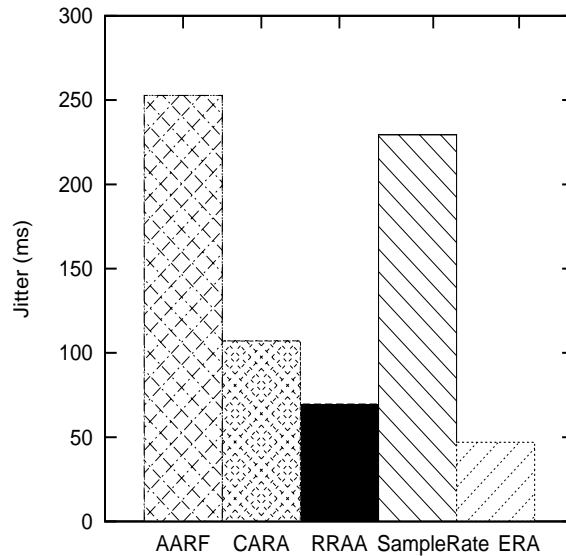
Figure 4.21: Jitter

packets. The report is generated at intervals of 500 ms. Then the largest jitter is recorded as the final result. Figure 4.21 plots the largest jitter for each rate adaptation scheme where the y-axis is the jitter in ms. *ERA* performs best (with smallest jitter) in this scenario. This is because *ERA* deploys a prompt loss recovery strategy upon each loss in both collision and channel fading cases. Except for *ERA* and *RRAA*, all other schemes deliver jitters larger than 100 ms, the time constraint required by real-time multimedia applications such as VoIP. The percentage of out-of-order packets is shown in Figure 4.22. The y-axis depicts the percentage of out-of-order packets. With *ERA*, the percentage is reduced to about 10%. A larger number of out-of-order packets requires more buffer at the application client to play back multimedia.

Heterogeneous Clients Contention

To evaluate the contention among these rate adaptation schemes, four laptops clients transmit together, each of which runs one of schemes *CRA, RRAA, SampleRate* and *ERA*. *AARF* is not taken
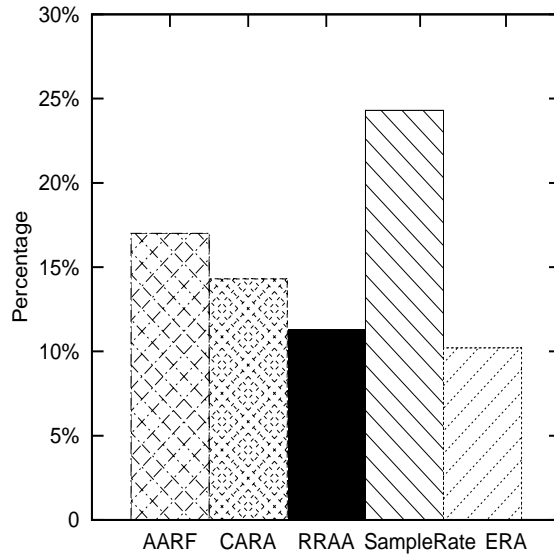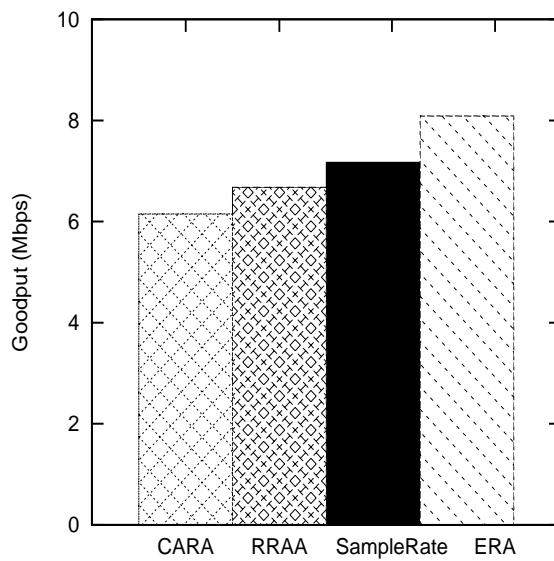
Figure 4.22: Percentage of packets out-of-order

Figure 4.23: UDP goodput in heterogeneous competition

90

into experiments because it performs poorly in collision or mixed environments. For fairness to each scheme, the four laptops are evenly distributed on a circle centering at the access point. We checked that they all nearly receive the same signal strength. The measurement is illustrated in Figure 4.23. The y-axis represents the goodput of each scheme. From the result, *ERA* wins the contention. It achieves almost 35% improvement over *CARA*. Its outstanding performance benefits from its accuracy in loss diagnosis and prompt recovery strategy in the heavy contention environment.

Increasing contention level

We also measured the performance of each rate adaptation scheme undergoing different collision levels. In this scenario, one client running one rate adaptation scheme each time is statically placed at location *L-2*. The collision level is gradually increased by adding one more interfering stations at locations *I-1, I-2, I-3,* and *I-4*. The experiments start with two interfering stations and end up with four. All interfering stations generate the same UDP traffic to the receiver through the access point. Figure 4.24 plots the results. To keep consistent, we start the interference from 2 stations (we take 2 interfering stations as minimum interference in all our experiments). Although *ERA* does not perform best with 2 interfering stations, it outperforms all others as the collision level increases and its gain is higher in heavier collision: 40% more than the runner up. When collision is light, the overhead introduced by fragmenting in *ERA* is not offset by its benefits. But, as the collision becomes severe, the benefits of fragmenting in collision environments result in the outstanding performance of *ERA*.

Field Tests

To evaluate the performance of these schemes in practice, we also conduct two field tests by connecting client(s) running with rate adaptation schemes to an access point in the campus network.
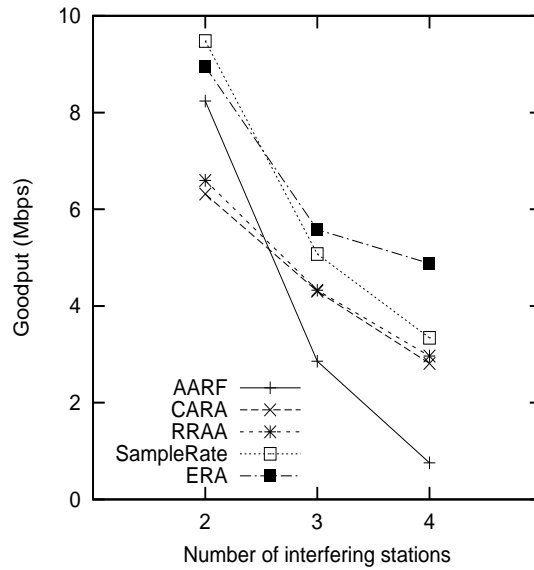
Figure 4.24: UDP goodput under collision levels

One scenario is measured with UDP traffic while the other with TCP. All clients are stationarily placed. For the scenario with UDP traffic, the four laptops are distributed with almost the same signal strength to the campus access point. The clients are competing with other anonymous clients transmitting to the same access point. Also, they contend with each other. Consistent to the results obtained in Figure 4.23, *ERA* performs best, as shown on Figure 4.25.

In the TCP experiment, we run one client only. One scheme is measured each time. The experiment results are plotted in Figure 4.26. As we observe, *ERA* marginally outperforms *RRAA*. But it can outperform *CARA* and *AARF* by more than 30%.
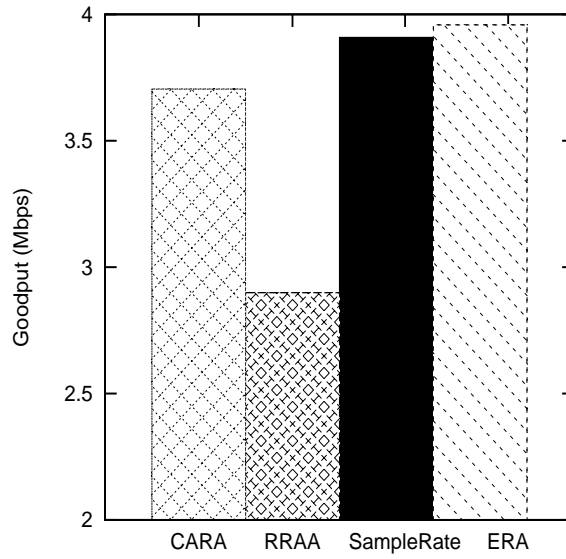
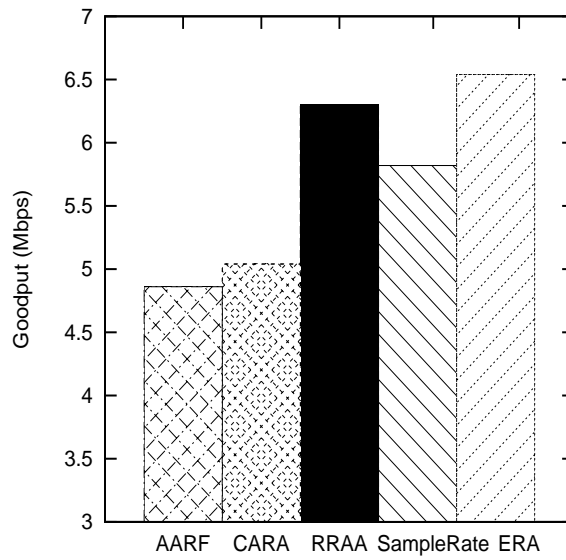Figure 4.25: UDP goodput in field test

Figure 4.26: TCP goodput in field test

CHAPTER 5

CONCLUSIONS AND FUTURE WORK

Multi-rate support in IEEE 802.11 networks requires efficient and effective rate adaptation solutions to fully exploit scarce wireless resource. In this dissertation, we review rate adaptation schemes in literature for IEEE 802.11 networks. Although rate adaptation on IEEE 802.11 networks has been deemed for years, most of these schemes are based on SNR/RSSI that are studied not closely correlated to the successful frame delivery. Different approaches are also proposed to adapt rates based on frame loss. However, most of them do not consider the causes of different losses, assuming that the collision in IEEE 802.11 networks are avoided by the use of RTS/CTS. The real time trace from practice shows the presence of RTS/CTS frames is rare in the traffic in IEEE 802.11 networks because they are optional due to the overhead. Such overhead is particularly severe in multimedia communication that is surging in daily life. The absence of RTS/CTS frames results in complexity to rate adaptation because the frame losses might be caused by channel degradation and also by transmission collision. Different frame loss can not be addressed by the same strategy.

In our effort to identify an effective rate adaptation scheme, we successively propose three rate adaptation strategies. With our investigation of proposed rate adaptation schemes in the literature, we found that there is no solution taking into account the estimation of initial data rate to start the transmission of a stream of data frames. But a proper initial rate can improve the utilization of network resource or reduce frame losses in the beginning. Therefore, we propose a scheme called *Beacon Assisted Rate Adaptation* **BARA** for the initial rate and ongoing rates with the periodically broadcast beacon frames which are mandatory in IEEE 802.11 standard. This technique benefits the

94

network performance without introducing any extra transmission overhead to data frames in networks with or without RTS/CTS control frames. To differentiate frame losses, we propose a second scheme *Loss Differentiation Rate Adaptation* **LDRA** that diagnoses the cause of losses and quickly recovers lost frames with the basic rate retransmission technique. Also, in this dissertation, we extensively investigate the rate adaptation schemes proposed in recent years with explicit consideration of different types of frame loss. Implementing these schemes on a Linux based testbed, we observe some anomalies in their adaptation dynamics in running time. Based on these observations, we design an effective rate adaptation scheme *Effective Rate Adaptation* **ERA**. It is a "winning streak" scheme increasing rate upon the number of consecutive successfully transmitted data frames. But, unlike other frame loss based schemes, *ERA* does not reset the count of this number in confronting any loss from collision. Namely, the collision does not interrupt the counting of frame losses to indicate channel degradation. To dissipate frame losses in stable channel environments, *ERA* adaptively extends the rate increase cycle to exploit potential better channel conditions. Upon a data frame loss, *ERA* judiciously exploits the IEEE 802.11 fragmentation mechanism in full compliance with the standard to diagnose the cause. In this work, we analytically and numerically show the benefits of fragmentation: 1) *ERA* diagnoses loss accurately; 2) the loss can be recovered quickly; 3) it even incurs less overhead at the intermediate rate than RTS/CTS control frames. For a diagnosed collision, *ERA* diffuses the congestion with a short (lead) fragment and retains the rate. Moreover, it halves the rate for a diagnosed channel degradation to quickly recover loss.

We evaluate the benefits of these three schemes with extensive simulation on a network simulator. Furthermore, to evaluate the performance of rate adaptation schemes in practice, we implement on a Linux based testbed *ERA* and four other representative adaptation schemes: *AARF, CARA,*

RRAA, and *SampleRate*. Extensive both controlled and public field experiments on the testbed suggest that *ERA* often performs best in the most channel fading or collision dominated environments. Also, we observe that each of these schemes has undeniable strengths and some weaknesses.

Part of our future work is to understand the IEEE 802.11 channel dynamics. As reviewed in Section 1.3, there are limited measurements of IEEE 802.11 channels under realistic scenarios, though some work has been conducted in literature. Most current measurements are limited in the following aspects: 1) the measurement experimental settings are limited to specific testbeds or controlled tests. These environments are not generic or representatives of public hot spots. 2) most of the experiments were conducted on IEEE 802.11b channels. However, most of current networks operate on 802.11g or even the 802.11n draft recently. It is shown that the different encoding schemes lead to a considerable difference in performance among these IEEE 802.11 variants. 3) the observations are primarily from overall network performance. Investigation of detailed per frame channel dynamics is scarce. Therefore, the next step is to analyze IEEE 802.11 channel dynamics with extensive experiments in both controlled tests and public sites. This analysis is needed to better understand traffic and channel models.

Our future agenda also includes the design of a hybrid rate adaptation scheme. As we observe with the extensive experiments in practice on our implemented testbed, although *ERA* outperforms its peers in most scenarios, every implemented rate adaptation scheme demonstrates its own strength. It is challenging but beneficial to design a rate adaptation scheme that synergies the strengths and mitigates the weaknesses of the above five schemes on the testbed. This design will be based on the above understanding of channel dynamics, and on strategies considering both short-term and long-term network performance. As multimedia applications surge in daily life, a new rate

adaptation scheme should not only perform efficiently for traditional TCP based services, e.g. FTP, HTTP, but also for these new multimedia services, e.g. online video, radio, VoIP.

Another future work is to integrate rate adaptation into routing metrics. Recent years, research on routing protocols in multiple hops wireless networks, e.g. mesh networks, ad-hoc networks, has gained extensive attention. However, most of these innovative protocols are based on very flat routing metrics. Since different links along a routing path operate on rates adapted frequently by rate adaptation, introduction of rate adaptation into routing metrics should significantly impact route selection and improve network performance.

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