A Media-Oriented Transmission Mode Selection in 802.11 Wireless LANs

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Abstract

In this paper, we present a media-oriented mechanism for selecting the appropriate transmission mode in 802.11-based wireless LANs (WLANs). The main goal of this mechanism is to improve the effective throughput for transporting loss-tolerant multimedia traffic over a WLAN by taking into account both the application characteristics and the physical channel conditions. In particular, the proposed cross-layer mechanism exploits the robustness of multimedia coding by allowing packets with corrupted payloads reach the receiving application. The sending application specifies its quality of service requirements (data rate, BER tolerance, etc.), and the receiver selects the best transmission mode (transmission rate, modulation scheme, FEC scheme) while taking into account the time-varying channel conditions. We discuss the modifications needed for the control and data-packet headers to implement our approach in the framework of the IEEE 802.11 standards. We use ns2 simulations to contrast our scheme with an existing 802.11 rate selection algorithm. The results indicate that the proposed cross-layer approach achieves up to 5 Mbps increase in throughput and 20-meter increase in the coverage range. Furthermore, by disabling FEC from some of the standard transmission modes, we show that the goodput of loss-tolerant applications can be improved significantly.

Keywords : Wireless LAN, IEEE 802.11, Cross-Layer interactions, multirate.

1. Introduction

In recent years, high-speed WLANs have become widely popular in various sectors, including health care, manufacturing and academic settings. These sectors benefited from the productivity gains of using hand-held terminals and notebook computers to transmit real-time information within physically distributed environments. Currently, the IEEE 802.11 is the de facto standard for WLANs [1]. It specifies both the medium access control (MAC) and the physical (PHY) layers for WLANs. According to this standard, the MAC layer operates on top of one of several physical layers. Medium access is performed using carrier sense multiple access with collision avoidance (CSMA/CA) [1]. The increasing number of wireless users and the demand for high-bandwidth multimedia applications over WLANs led the IEEE working groups to provide powerful physical layers and to extend the MAC layer to provide QoS support.

Concerning the physical layer, three IEEE 802.11 standards are currently available: a, b, and g [2]. The 802.11b standard is the most widely deployed in today's WLANs. Since the end of 2001, higher data rate products based on the 802.11a standard have appeared in the market. More recently, the IEEE 802.11 working group has approved the 802.11g standard, which extends the data rate of the IEEE 802.11b to 54 Mbps[1]. As described in the next section, each of these three physical-layer standards support a multitude of *transmission modes*. A transmission mode is specified by a data rate, a modulation scheme, and an error control scheme (e.g., FEC), if any.

In this paper, we propose a simple and efficient media-oriented mechanism for dynamically adjusting the transmission mode in 802.11-based WLANs. This mechanism is aimed at loss-tolerant (LT) applications (e.g., video and audio), which does not require 100% reliability (i.e., the loss of few packets can be ignored or concealed at the receiver). Our proposed mechanism takes into account both the intrinsic characteristics of the application and the channel conditions. It selects the highest available transmission rate (mode) while guaranteeing a specific bit error rate (BER). The selected mode varies in time depending on the loss sensitivity of the packet and on the observed signal-to-noise ratio (SNR) at the receiver. For instance, in the case of MPEG video, packets are obtained from video frames with dif-

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[1] Multi-standard (802.11a/b/g) cards are already available.

ferent levels of "importance" (e.g., intra-coded frames are more significant than predicted frames), which maps into different degrees of loss tolerance. By adaptively changing the transmission mode depending on the loss sensitivity and the channel state, a marked improvement in the application-layer throughput can be achieved. Throughout this paper, we assume that wireless stations use the Enhanced Distributed Coordination Function (EDCF), proposed in the IEEE 802.11e [3] to support different levels of QoS. As shown in Section 3, our scheme can be incorporated in the existing standard after some minor modifications.

In the past few years, significant works have been done in wireless network to increase throughput, but most of them have concentrated on a single layer in the protocol stack. In this paper, we describe a cross layer mechanism that uses application's information in the MAC layer to decide about physical data rate. This is a simple example of cross layer optimization in WLANs that shows the potential profits of such a design. We have modified the ns simulation tool to evaluate the total system efficiency considering interaction between layers in the protocol stack. Cross layer optimization mechanisms can be investigated with this tool.

The rest of this paper is structured as follows. In Section 2 we overview the salient features of the MAC and PHY layers in the 802.11 schemes. We also review some of the automatic rate selection algorithms that were proposed in the literature. Our mechanism and a possible implementation of it within a 802.11-compliant device are described in Section 3. Simulation results are presented in Section 4, followed by conclusions and open issues in Section 5.

2. Background

2.1. 802.11 Distributed Coordination Function

The Distributed Coordination Function (DCF) of the IEEE 802.11 standard defines how the medium is shared among stations on a WLAN. Medium access is achieved using a CSMA/CA mechanism, as explained in Figure 1.

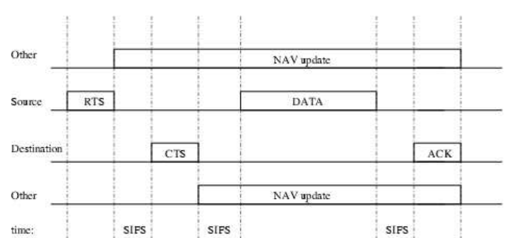

Figure 1. CSMA/CA protocol.

If the channel is busy, a backoff time (measured in time slots) is chosen randomly in the interval [0,CW), where CW is called the *contention window*. Once the medium has been detected idle for at least a DIFS time interval, the backoff timer is decremented by one for each time slot the medium remains idle. When the backoff timer reaches zero, the source transmits a request-to-send (RTS) packet, containing the duration of ensuing data packet, the clear-to-send (CTS) packet, and the ACK. RTS and CTS packets are sent using the *basic transmission mode*, defined as the lowest rate supported by the underlying physical layer (all 802.11-compliant nodes *must* support the basic mode).

Nodes in the sender's range that hear the RTS packet update their Network Allocation Vectors (NAVs) and defer their transmissions for the duration specified by the RTS. After receiving the RTS, if the receiver wishes to receive the packet, it responds with a CTS that contains the duration of the upcoming transmission. Nodes that overhear the CTS packet update their NAVs and refrain from transmitting.

2.2. IEEE 802.11e

The IEEE 802.11e draft proposes many features to support QoS in WLANs [3]. Figure 2 shows the EDCF scheduler at a QoS-enhanced station (QSTA) in the 802.11e. Each QSTA has 4 queues to support up to 8 User Priorities (UPs). UPs are assigned by the application layer and are mapped to *access categories* based on a simple mapping table [3]. For example, best-effort traffic, video probes, voice, and video data can use four different access categories.

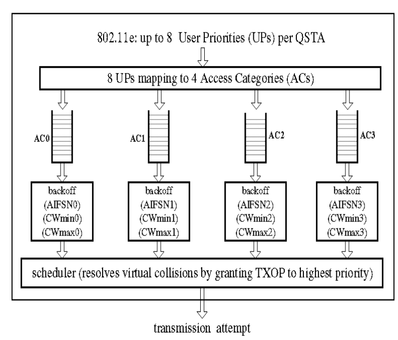

Figure 2. QoS support in 802.11e EDCF.

2.3. IEEE 802.11 Physical Layers

Table 1 shows the main characteristics of the IEEE 802.11 a, b and g physical layers. In each physical layer, the basic mode has the maximum coverage range among all transmission modes. This maximum range is obtained by using BPSK modulation[2] and the minimum data rate. As

[2]Compared to other modulations schemes, BPSK has the minimum probability of bit error for a given SNR.

Table 1. Characteristics of the various physical layers in the IEEE 802.11 standard.

Characteristic	802.11a	802.11b	802.11g
Frequency	5 GHz	2.4 GHz	2.4 GHz
Data Rates	6, 9, 12, 18, 24, 36, 48, 54 Mbps	1, 2, 5.5, 11 Mbps	1, 2, 5.5, 6, 9, 11, 12, 18, 22, 24, 33, 36, 48, 54 Mbps
Modulation	BPSK, QPSK, 16 QAM, 64 QAM	BPSK, QPSK,CCK	BPSK, QPSK, 16 QAM, 64 QAM, CCK
FEC Rate	1/2, 2/3, 3/4	NA	1/2, 2/3, 3/4
Basic Transmission Mode	BPSK, 6 Mbps, FEC 1/2	BPSK, 1 Mbps	802.11a (6 Mbps) or 802.11b (1 Mbps) basic mode

shown in Figure 3, each packet may be sent using two different rates; the PLCP (Physical Layer Convergence Protocol) header is sent at the basic rate while the rest of the packet might be sent at a higher rate. As shown in Table 1, this basic rate is 1 Mbps for 802.11b and 6 Mbps for 802.11a. The higher rate used to transmit the physical-layer payload (which includes the MAC header) is indicated in the PCLP header. The receiver verifies that the PLCP header is correct (using CRC or Viterbi decoding with parity), and uses the transmission mode specified in the PLCP header to decode the MAC header and payload. As explained earlier, the mode with the lowest rate is used to transmit the PLCP header. So, sometimes the PLCP header is accepted but the rest of the packet may be corrupted.

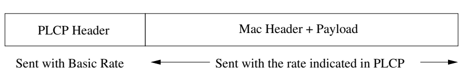

Figure 3. Data rates for packet transmission.

2.4. Rate Selection Algorithms

At each station, transmission mode selection can be performed either manually or automatically. In principle, if channel conditions are suitable, a station can increase its sending rate by selecting a new mode. A number of rate selection mechanisms have been proposed in the literature. They include the Auto Rate Fallback (ARF) [4] and the Receiver-Based Auto Rate (RBAR) [5] schemes. RBAR aims at selecting the best available mode based on the received SNR, while ARF uses a simple ACK-based mechanism to select the rate. Our proposed mechanism is based on RBAR. In RBAR, the sender chooses a data rate based on some heuristic (e.g., the most recent rate that was used to successfully transmit a packet), and then stores the rate and the size of the data packet into the RTS. Stations that receive the RTS calculate the duration of the requested reservation using the rate and packet size carried in this RTS. They update their NAVs to reflect the reservation. While receiving the RTS, the receiver uses the current channel state as an estimate of the channel state at the time the upcoming packet is supposed to be transmitted. The receiver then selects the appropriate rate using a simple threshold-based mechanism and includes this rate (along with the packet size) in its CTS. Stations that overhear the CTS calculate the duration of the reservation and update their NAVs accordingly. Finally, the sender responds to the CTS by transmitting the data packet at the rate selected by the receiver. In RBAR, nodes that cannot hear the CTS update their NAVs when they overhear the actual data packet by decoding a part of the MAC header called the *reservation subheader*. It should be noted that RBAR has not been standardized by IEEE 802.11 WG yet. Further information concerning RBAR, including implementation and performance issues in 802.11b is available in [5].

3. Media-Oriented Mode Selection

We now describe our media-oriented rate-adaptive scheme. This scheme can be used on both ad hoc and infrastructure networks. We also describe how this scheme can be integrated into a wireless station that uses RBAR for rate selection and that supports an EDCF MAC layer.

Multimedia applications, in general, are characterized by their ability to tolerate certain amounts of packet loss. These losses can be totally ignored (since they are barely noticeable by human beings) or can be compensated for at the receiver using various error concealment techniques. For instance, in some applications the information is updated every few seconds (e.g., status reports for stock market screens and airline information screens). For these applications, if one message is lost, a more updated message is transmitted a few seconds later. Furthermore, some multimedia applications have their own error control mechanisms [6], making it inefficient to provide 100% reliability at the link layer.

In our scheme, the sender has to specify the Loss Tolerance (LT) of the transported traffic in order that the receiver uses both this information and the current channel conditions to select the appropriate transmission mode (rate, modulation, and FEC). Basically, the LT information is included in each RTS packet (in order that the receiver select the best mode) and in the header of each corresponding data packet (to let the receiver decide to accept or not a packet according to its LT). This mechanism can be implemented with the help of the EDCF protocol in the MAC layer. Figure 4 shows the QoS control field that is added to the MAC header in 802.11e specification [3]. Bits 6 and 7 of this header can be used to indicate the loss tolerance. Table 2 shows one possible use of these two bits.

Bit 0-3	Bit 4	Bit 5	Bit 6-7	Bit 8-15
Traffic ID	*Schedule Pending*	*Ack Policy*	*Reserved*	*TXOP duration*

Figure 4. QoS control field in the 802.11e.

LT information can be sent to receiver by adding one byte to RTS packets as illustrated in Figure 5. While receiving the RTS, the receiver uses the information concerning the channel conditions along with the information related to LT to select the best data rate for the corresponding packet.

Table 2. Loss Tolerance classification.

Bit 6-7	Application Sensitivity
00	No tolerance (FTP)
01	1% loss tolerance (Voice)
10	5% loss tolerance (Video I-frames)
11	10% loss tolerance (Video P- and B-frames)

We have modified the RBAR threshold mechanism to take into account both SNR and LT information, see Figure 6. The receiver will use arrays of thresholds precomputed for different LT. The selected rate will then be transmitted along with the packet size in the CTS back to the sender.

OCTETS: 2	2	6	6	1	4
Frame Control	Rate & Length	Dest Address	Source Address	Tolerance Information	FCS

Figure 5. Modifications to the RTS header.

For the headers of the various layers (MAC, IP, UDP, and RTP), we propose sending them at the basic rate, which is the most robust rate against bit errors. This is somewhat similar to the reservation sub-header used in [5]. Figure 7 shows that the PLCP header encodes payload. In order to let a packet with corrupted payload reach the receiver application, the MAC CRC should cover only the MAC/IP/UDP/RTP headers. Moreover, the optional UDP checksum should be disabled. If the receiver is able to decode the headers, it can identify the BER tolerance for the encoded payload. If the packet is loss tolerant, it will be accepted even if its payload contains errors. As will be shown later, our mechanism makes it possible to define new transmission modes that do not use FEC but that exhibit comparable performance (in coverage range and throughput) to the ones with FEC.

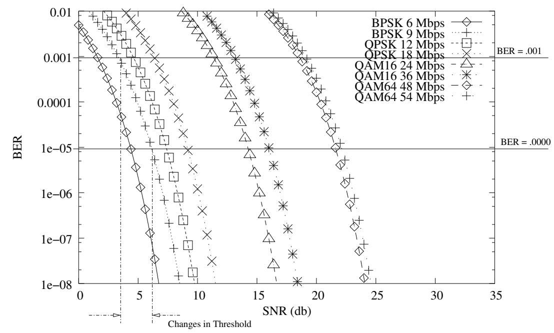

Figure 6. Bit error rate (BER) versus SNR for various transmission modes (802.11a).

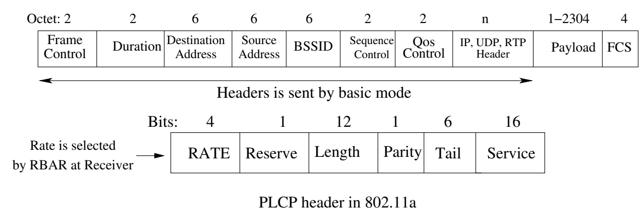

Figure 7. Proposed Frame format.

4. Simulation Results

To evaluate our mechanism, we use a simple topology with two wireless nodes shown in Figure 8. Station A is fixed, while Station B moves toward station A. After moving a distance of 5 meters, Station B pauses for 60 seconds during which it transmits 8000 packets, each of size 2304 bytes (including FEC for payload).

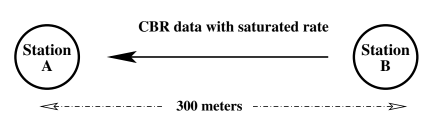

Figure 8. Network configuration.

Our simulations are based on the environment described in [5], which uses the ns-2.1b3 network simulator with extensions from the CMU Monarch project [7]. In order to obtain realistic results, Cisco Aironet 1200 Series parameters [8] are used in our simulations.

From the 8 possible transmission modes of the 802.11a standard, we select four modes with four different modulations and two FEC code rates. We also use the upper bound probability of error that is given in [9] for error model, under the assumption of binary convolutional coding and hard-decision Viterbi decoding. An evaluation of all transmission modes of the 802.11a and further information about the simulation environment are available in [10].

Figure 9 shows the mean throughput versus distance for different transmission modes. The basic mode (BPSK, 6 Mbps, 1/2 FEC code rate) gives the maximum coverage range. The figure also shows the performance of Predictive-RBAR (P-RBAR), which is a scheme that uses a cache to save the most recent rates as they are discovered [10, 5]. After several successful transmissions, there is no need to wait for the reservation sub-header in P-RBAR. The differences between the theoretical maximum rate and the actually achieved data rate is due to MAC overhead and FEC redundancy bits. Indeed, sending CTS/RTS before sending data decreases the mean throughput significantly.

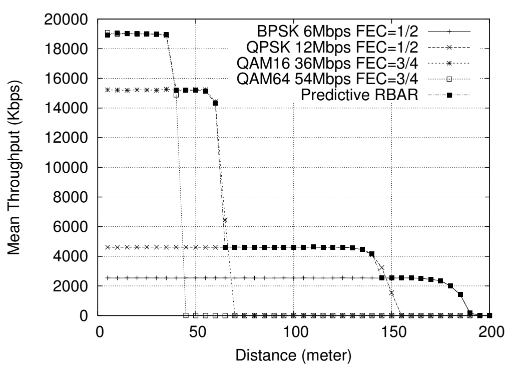

Figure 9. Mean throughput versus distance for four transmission modes and for P-RBAR.

We evaluated the extra bandwidth overhead of the modified frame format. This overhead is caused by having to send the MAC header at the basic mode and by the additional byte in the RTS packet. Figure 10 compares the mean throughput for the traditional P-RBAR and for P-RBAR with the modified frame format. The worst-case overhead at the maximum rate is about 1.5 Mbps, but the coverage range does not change much compared to the standard specification.

In order to evaluate the performance of FEC in WLANs, we simulated the same network configuration but without

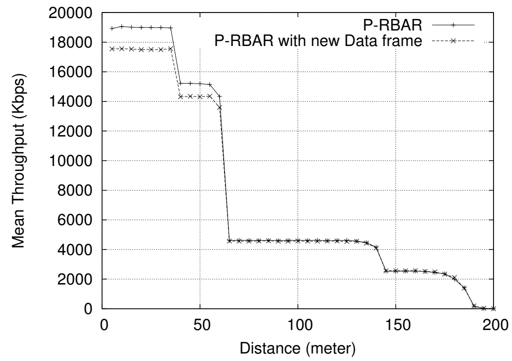

Figure 10. Modified frame format overhead.

the physical-layer FEC (which is applied to the whole payload of the physical-layer frame)[3]. The results are shown in Figure 11. Clearly, the mean throughput is increased significantly compared to the case with FEC. However, the transmission range has decreased. For example, the transmission range is 110 meters without FEC and 190 meters with FEC.

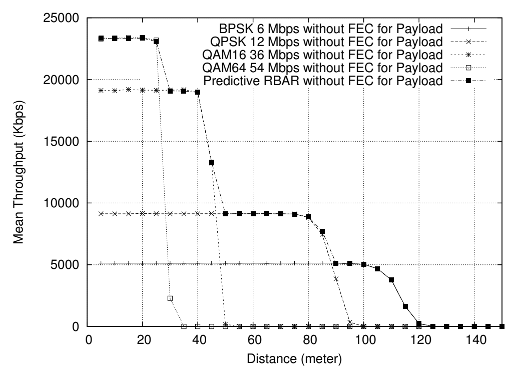

Figure 11. New transmission modes.

For the next simulations, we assume that the application can tolerate some losses and that bit errors in a packet are distributed according to a binomial distribution. If n represents the number of bit errors in a packet of N bits and p is the probability of bit error, then the probability of having less than L bit errors can be calculated by:

$$P(n \le L) = \sum_{i=0}^{L} \binom{N}{i} \cdot p^i \cdot (1-p)^{N-i}$$

We still use the basic mode to send PLCP and MAC headers and we do not accept packet with error in the header. But we accept packets with less than $\epsilon\%$ bit errors in their

[3] The basic mode is still BPSK 6 Mbps with 1/2 FEC code rate.

payloads, $\epsilon = 1$ and 10. Figure 12 shows the mean throughput versus distance when using BPSK modulation with 6 Mbps data rate. Compared to the standard specification of the 802.11a, the mean throughput and coverage range are both increased. Note that the complexity of Viterbi decoding for payload is removed in this case. This has the added advantage of reducing power consumption, which is a critical resource for wireless users.

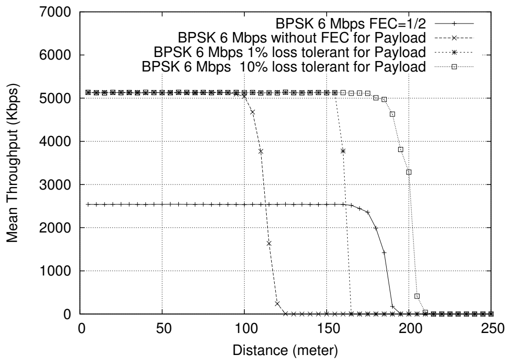

Figure 12. Mean throughput versus distance (basic mode) with and without loss tolerance.

Figure 13 shows the performance with and without LT when using P-RBAR for automatic rate selection. New thresholds are calculated based on the LT in the payload, as we explained in Section 3. The figure shows that under the media-oriented rate selection mechanism, there is about 5 Mbps improvement in throughput at the highest-rate mode and an increase in the coverage range by about 20 meters.

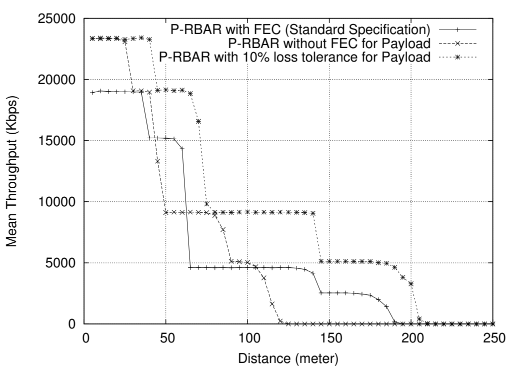

Figure 13. P-RBAR for LT applications.

5. Conclusion and Future Works

In this paper we have introduced a media-oriented rate selection algorithm for 802.11 WLANs. Our mechanism uses information from the physical channel and characteristics of loss tolerant applications to select the optimal PHY rate, modulation and FEC schemes. We have proposed new transmission modes with less complexity that significantly increase application throughput and coverage range. The mechanism can be implemented with some minor changes and achieves up to 5 Mbps increase in throughput and 20-meter increase in the coverage range.

Our future work includes a more thorough evaluation of the gain obtained from the application point of view. For example, the quality of corrupted audio flows could be assessed using the E-Model.

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