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► **To cite this version:**

Mohammad Hossein Manshaei, Thierry Turetletti, Marwan Krunz. Media-Oriented Transmission Mode Selection in 802.11 Wireless LAN. RR-4958, INRIA. 2003. inria-00071621

HAL Id: inria-00071621

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N° 4958

October 2003

THÈME 1



*Rapport
de recherche*

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Thème 1 — Réseaux et systèmes
Projets Planete

Rapport de recherche n° 4958 — October 2003 — 17 pages

Abstract: In this report, 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.

Key-words: Wireless LAN, IEEE 802.11, multirate

Mécanisme de sélection de mode de transmission en fonction des besoins de l'application pour réseaux locaux sans fil 802.11

Résumé : Dans ce rapport, nous présentons un mécanisme de sélection automatique de mode de transmission pour des réseaux locaux sans fil 802.11 qui s'adapte à la fois aux conditions du canal et aux caractéristiques du média transmis. L'objectif de cette étude est d'améliorer la transmission de certaines applications multimédia qui sont robustes aux erreurs de bits. En particulier, si la couche MAC est consciente que l'application tolère un certain pourcentage d'erreurs de transmission binaire, elle peut sélectionner un mode qui supporte un débit de transmission plus élevé. Pour cela, l'application émettrice peut spécifier une demande de qualité de service (débit de transmission, tolérance d'erreurs de transmission binaire, etc) afin que le récepteur sélectionne le meilleur mode de transmission tout en prenant en compte les conditions dynamiques du canal. Nous présentons les modifications nécessaires dans les en-têtes de paquets de contrôle et de données pour implanter ce mécanisme dans un réseau sans fil 802.11. Nous avons comparé notre mécanisme avec d'autres algorithmes de sélection de débit par le biais du simulateur réseau NS-2. Les résultats montrent que le nouveau mécanisme permet d'augmenter jusqu'à 5 Mb/s le débit reçu et d'accroître de 20 mètres la portée de la transmission.

Mots-clés : Réseaux sans fil 802.11, multi-débits, interaction inter-couches de communication.

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 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: 802.11b, 802.11a, and 802.11g. The 802.11b standard is the most widely deployed in today's WLANs [2]. Since the end of 2001, higher data rate products based on the 802.11a standard [3] have appeared in the market.

More recently, the IEEE 802.11 working group has approved the 802.11g standard [4], which extends the data rate of the IEEE 802.11b to 54 Mbps¹. 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., Forward Error Correction(FEC), if any.

In this report, 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 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.

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 report, we assume that wireless stations use the Enhanced Distributed Coordination Function (EDCF), proposed in the IEEE 802.11e [5] to support different levels of QoS. 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. The mechanism we describe in this report is 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 remainder of this report is organized as follows.

The rest of this report is structured as follows. In Section 2 we briefly overview the salient features of the MAC and physical 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

The 802.11 standard defines two forms of medium access, Distributed Coordination Function (DCF) and Point Coordination Function (PCF), which work with different PHY layer standards.

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

DCF is mandatory and based on the CSMA/CA (carrier sense multiple access with collision avoidance) protocol. For supporting time-bounded delivery of data frames, the 802.11 standard defines the optional point coordination function (PCF) where the access point grants access to an individual station to the medium by polling the station during the contention free period. Stations can't transmit frames unless the access point polls them first. The period of time for PCF-based data traffic (if enabled) occurs alternately between contention (DCF) periods. PDF is an optional MAC layer and hasn't been implemented widely because the technology's transmission times are unpredictable. In this section we explain DCF in more detail.

The IEEE 802.11e working group effort to support QoS in wireless LANs that gives support to bandwidth-sensitive applications such as voice and video. The enhanced MAC layer to support QoS in WLAN is explained in this section. We address different PHY layers and current rate selection algorithms for 802.11 WLANs in this section.

2.1 802.11 Distributed Coordination Function

The 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.

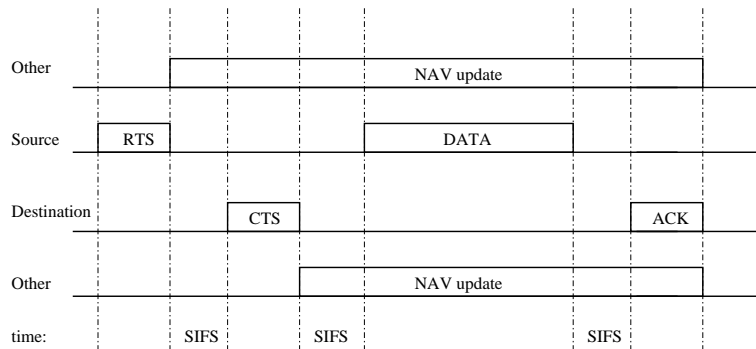


Figure 1: CSMA/CA protocol.

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). The basic mode also has the farthest transmission range.

2.2 IEEE 802.11e

The IEEE 802.11e draft proposes many features to support QoS in WLANs [5]. Figure 2 shows the EDCF, in a QoS-enhanced station (QSTA). Each QSTA has 4 queues to support up to 8 User Priorities (UP) [6]. User priorities are assigned by the application layer and are mapped to Access Categories (AC) based on a simple mapping table. Table 1 shows that one or more UPs can be mapped to the same AC queues. Based on IEEE 802.11e standard specification [5] this mapping (before there were 8 ACs) is done to reduce complexity.

There are two main methods to support service differentiation in EDCF: Using different Inter-frame Space (IFS) size or allocating different CW sizes for different AC. Further information about these mechanisms and a complete survey of QoS enhancements for 802.11 WLANs are available in [6].

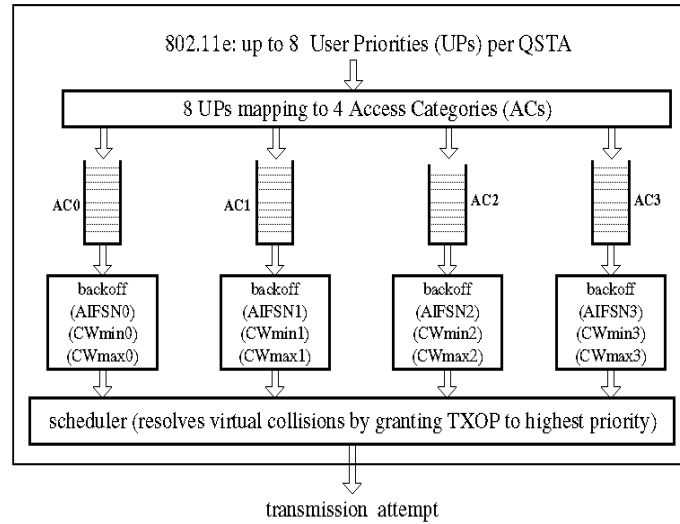


Figure 2: EDCF proposed by 802.11e

User Priority (UP)	802.11D Designation	802.11e AC (Access Category)	Service type
2	Not defined	0	Best Effort
1	Background (BK)	0	Best Effort
0	Best Effort (BE)	0	Best Effort
3	Excellent Effort (EE)	1	Video Probe
4	Controlled Load (CL)	2	Video
5	VI (Video < 100ms latency and jitter)	2	Video
6	VO (Video < 10ms latecnry and jitter)	3	Voice
7	Network Control (NC)	3	Voice

Table 1: Mapping between User Priority and Access Category

2.3 IEEE 802.11 Physical Layer

Table 2 shows the main characteristics of the IEEE 802.11a, 802.11 b and 802.11g physical layers. 802.11b radios transmit at 2.4 GHz and send data up to 11 Mbps using Direct Sequence Spread Spectrum (DSSS) modulation; whereas, 802.11a radios transmit at 5 GHz and send data up to 54 Mbps using Orthogonal Frequency Division Multiplexing (OFDM).

The IEEE 802.11g standard for WLANs [4], which extends the data rate of the IEEE 802.11b to 54 Mbps in an upgraded PHY layer named extended rate PHY layer (ERP) , has been approved by the IEEE 802.11 Working Group.

Table 2: Characteristics of the various physical layers in the IEEE 802.11 standard.

<i>Characteristic</i>	<i>802.11a</i>	<i>802.11b</i>	<i>802.11g</i>
<i>Frequency</i>	5 GHz	2.4 GHz	2.4 GHz
<i>Data Rates</i>	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
<i>Modulation</i>	BPSK, QPSK, 16 QAM, 64 QAM	BPSK, QPSK, CCK	BPSK, QPSK, 16 QAM, 64 QAM, CCK
<i>FEC Rate</i>	1/2, 2/3, 3/4	NA	1/2, 2/3, 3/4
<i>Basic Transmission Mode</i>	BPSK, 6 Mbps, FEC 1/2	BPSK, 1 Mbps	802.11a (6 Mbps) or 802.11b (1 Mbps) basic mode

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 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 2, this basic rate is 1 Mbps (with BPSK modulation and CRC 16 bits) for 802.11b and 6 Mbps (with BPSK and FEC rate equal to 1/2) for 802.11a. The higher rate used to transmit the physical-layer payload (which includes the MAC header) is indicated in the PLCP 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.

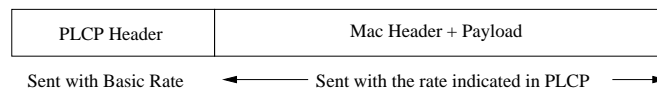


Figure 3: Data rates for packet transmission.

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) [7] and the Receiver-Based Auto Rate (RBAR) [8] 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 (such as the most recent rate

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

that was successful for transmission to the receiver), and then stores the rate and the size of the data packet into the RTS. Other stations, overhearing the RTS, calculate the duration of the requested reservation using the rate and packet size carried in the RTS. They update their NAV (Network Allocation Vector) to reflect the reservation. While receiving the RTS, the receiver uses the information concerning the channel conditions to compute an estimation of the conditions for pending data packet transmission. The receiver then selects the appropriate rate with a simple threshold mechanism, and transmits it along with the packet size in the CTS back to the sender. Other stations, overhearing the CTS, calculate the duration of the reservation similar to the procedure used by stations when they receive RTS and then update their NAV to reflect the reservation. Finally, the sender responds to the receipt of 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*. Implementation issues of RBAR and performance obtained for 802.11b are available in [8].

3 Media-Oriented Mode Selection

We now describe our media-oriented rate-adaptive algorithm that can be used on both ad hoc and infrastructure networks, then we explain how this scheme can be integrated into a wireless station supporting EDCF that uses RBAR for rate selection.

3.1 Algorithm Description

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 [9], 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).

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.

We have to modify the RBAR threshold mechanism to take into account both SNR and LT information, see Figure 4. Threshold calculation for different LT will be explained in the next section. 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, and sender uses this rate to send its data packets.

While the packet arrives to the receiver, 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.

Following we will explain implementation issues of our mechanisms in 802.11 WLANs.

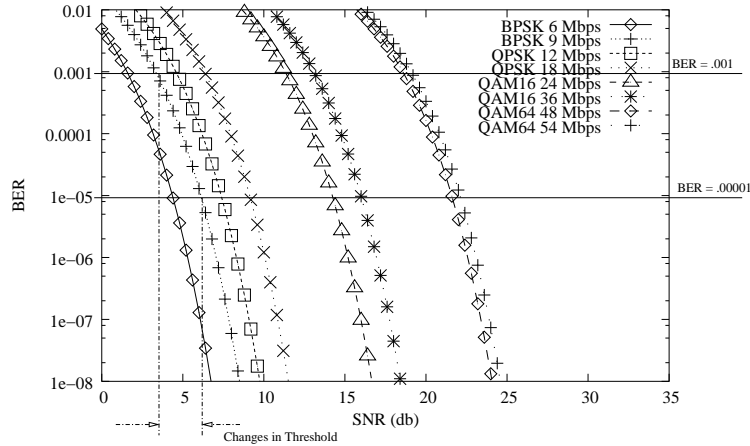


Figure 4: Bit error rate (BER) versus SNR for various transmission modes (802.11a).

3.2 Implementation

This mechanism can be implemented with the help of the EDCF protocol in the MAC layer. Figure 5 shows the QoS control field that is added to the MAC header in 802.11e specification [5]. Bits 6 and 7 of this header can be used to indicate the loss tolerance. Table 3 shows one possible use of these two bits.

Bit 0-3	Bit 4	Bit 5	Bit 6-7	Bit 8-15
<i>Traffic ID</i>	<i>Schedule Pending</i>	<i>Ack Policy</i>	<i>Reserved</i>	<i>TXOP duration</i>

Figure 5: QoS control field in the 802.11e.

As we explained LT information can be sent to the receiver by adding one byte to RTS packets as illustrated in Figure 6.

Table 3: 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)

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

Figure 6: Modifications to the RTS header.

Concerning 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 [8]. Figure 7 shows MAC and PLCP header formats in

802.11 [1, 2, 3]. It shows that the PLCP header encodes payload and header rates. The rate to transmit the header is always the basic rate to ensure receiving headers without error, while we will accept payload with some bit error. This causes an overhead over data packets, which will be investigated in the next section. 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.

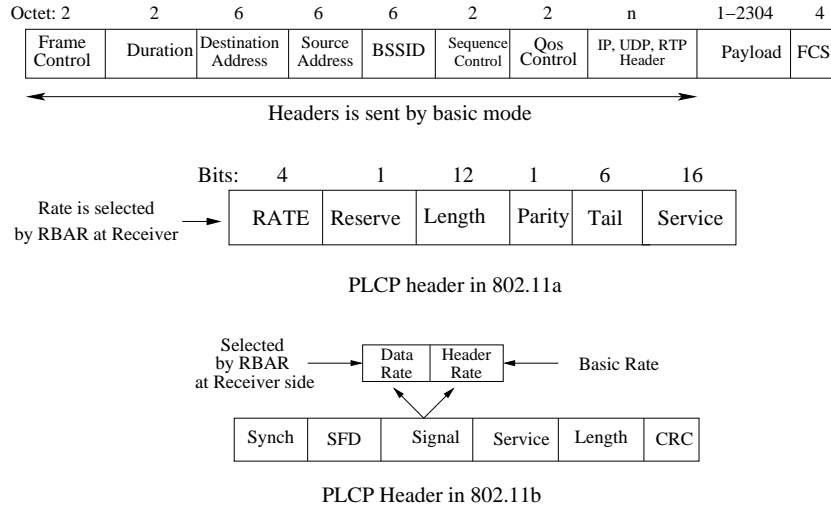


Figure 7: Proposed Frame format.

4 Simulation Results

Our simulations are based on the simulation environment described in [8] which uses the ns-2.1b3 network simulator, with extensions from the CMU Monarch project [11]. The Rice Monarch Project has made extensions to the ns-2 network simulator that enable it to simulate mobile nodes communicating by wireless network interfaces, including the ability to simulate multi-hop wireless ad hoc networks.

Holland [8] has modified this simulator in order to consider the effect of wireless physical layer in modeling mobile networks. Physical layer parameters like path loss, fading, interference and noise computation are usually not taken into account in WLAN simulations in spite of their important effects in simulation results [12]. In this simulator, Rayleigh fading channel and log-distance path loss model [8] are used for error model and estimation of received signal respectively. Moreover this simulation uses the Friis free space propagation model [8]. Further details on this simulation environment are available in [8] and the ns group is currently working to import these new functionalities to the next release of ns [10].

4.1 802.11a Simulation

For 802.11a/g, we assume that the receiver uses FEC Viterbi decoding. The upper bound probability of error provided in [15] is used under the assumption of binary convolutional coding and hard-decision Viterbi decoding. Specifically, for packet of length L this probability is:

$$P_e(L) \leq 1 - (1 - P_u)^{8L} \quad (1)$$

where the union bound P_u of the first-event error probability is given by

$$P_u = \sum_{d=d_{free}}^{\infty} a_d \cdot P_d \quad (2)$$

d_{free} is the free distance of the convolutional code, a_d is the total number of error events of weight d^3 and P_d is the probability that an incorrect path at distance d from the correct path is chosen by the Viterbi decoder. When hard decision decoding is applied, P_d is given by

$$P_d = \begin{cases} \sum_{k=(d+1)/2}^d \binom{d}{k} \cdot \rho^k \cdot (1-\rho)^{d-k} & \text{if } d \text{ is odd} \\ \frac{1}{2} \cdot \binom{d}{d/2} \cdot \rho^{d/2} \cdot (1-\rho)^{d/2} + \sum_{k=d/2+1}^d \binom{d}{k} \cdot \rho^k \cdot (1-\rho)^{d-k} & \text{if } d \text{ is even} \end{cases} \quad (3)$$

where ρ is the bit error probability for the modulation selected in PHY layer. In order to obtain more realistic results, Cisco Aironet 1200 Series parameters are used in our simulations [17]. Note that in the following simulations, CTS (Clear to Send) packets, RTS (Request to Send) packets and all data headers are sent with BPSK modulation with FEC rate equal 1/2 and 6 Mbps data rate. Note also that all throughput shown in the following figures exclude MAC and PHY headers.

Figure 8 shows the network topology used for the following simulations. Two wireless stations are communicating on a single channel. Station A is fixed and station B moves toward station A. Station B held fixed each 5 meters for a 60s transmission of data and we ensure that station B has always data to send to station A (with selecting proper rate) over a single CBR connection. 8000 CBR packets of size 2304 bytes (including FEC and payload) are sent in each step.

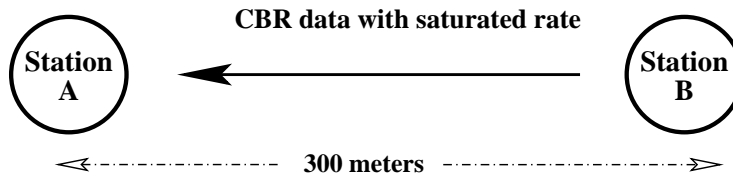


Figure 8: Network configuration.

Figure 9 shows the mean throughput of a single CBR connection for each mode according to the distance. One interesting point in this graph is the behaviour of mode 2. Similar to the analytical goodput evaluation presented in [13], PHY mode 3 achieves always better mean throughput (about 2 Mbps and with more coverage) than PHY mode 2. According to probability of bit error rate for QPSK and BPSK in [14], QPSK modulation has higher probability of bit error rate compared to BPSK, but the combination of rate 3/4 convolution code with BPSK achieves lower performance compared to rate 1/2 convolution code with QPSK. So mode 2 is not a good selection when mode 3 is available. However, using a suitable power control mechanism it can achieve better performance [13]. The differences between the theoretical maximum rate and the actually achieved data rate is due to MAC overhead and FEC overhead bits. Indeed, sending CTS/RTS before sending data decreases the mean throughput significantly.

Forward error correction is performed by adding bits to each transmitted character or code block, using a predetermined algorithm. Figure 10 shows the mean throughput once the redundancy data has been removed at the application level.

Referring to Figure 10, mode 5 has lower performance (about 2 Mbps) at application level comparing to mode 4. Thus, it is better not to use mode 5 when mode 4 is available.

4.2 Simulation of Media-Oriented Mechanism in 802.11a

In order to evaluate our media-oriented mechanism, from the 8 possible transmission modes of the 802.11a standard, we select four modes with four different modulations and two FEC code rates.

³We have used the a_d coefficients provided in [16].

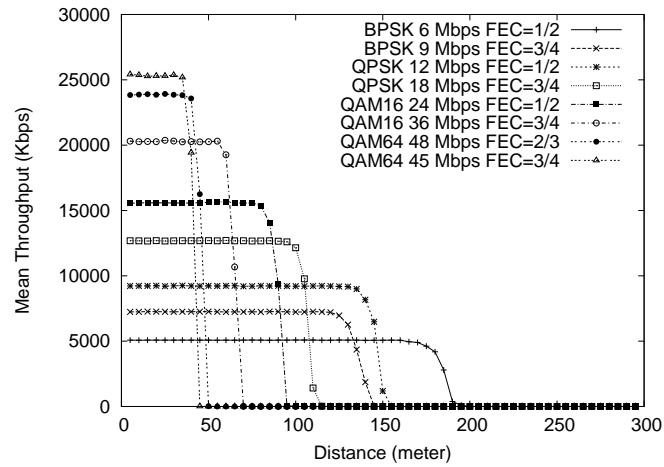


Figure 9: Mean throughput at PHY layer for single CBR connection

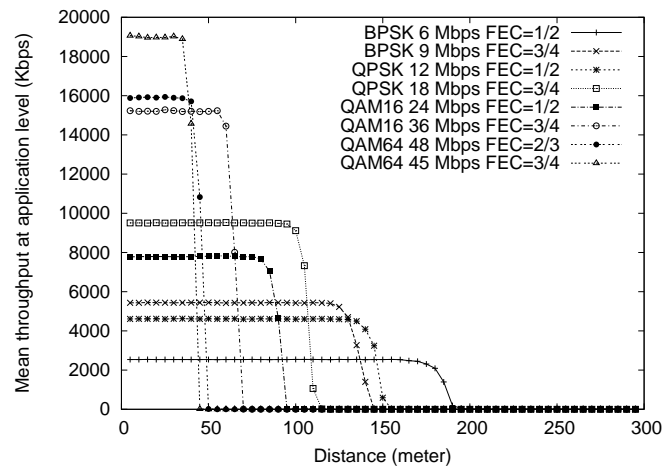


Figure 10: Mean goodput for single CBR connection

Figure 11 shows the mean throughput versus distance for these selected transmission modes. It should be considered that 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 [18, 8]. After several successful transmissions, there is no need to wait for the reservation sub-header in P-RBAR.

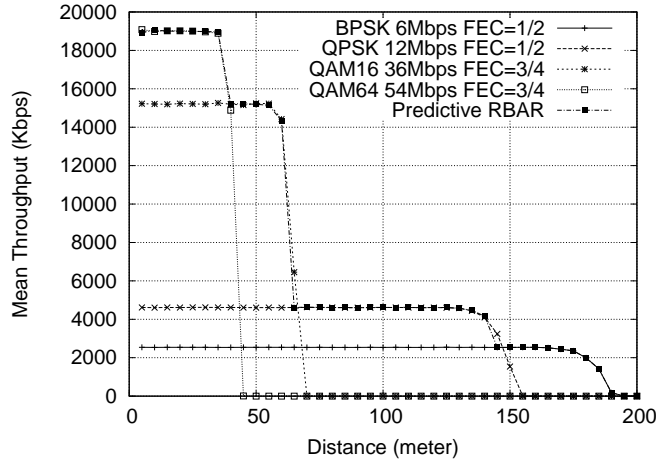


Figure 11: Mean throughput versus distance for four transmission modes and for P-RBAR.

It is necessary to calculate the threshold values to select the best rate based on SNR in order to use RBAR for 802.11a. Since Viterbi decoding algorithm is now used in the receiver, these thresholds are calculated using probability of error for all the modes discussed in Section 2. Table 4 shows these thresholds. In other words, with these SNR we have union bound of the first-event error probability less than $1e-5$, $1e-10$ or $1e-20$. In the following simulation we have used thresholds respecting to $P_u \leq 1E-5$.

Table 4: SNR(dB) threshold values for different union bounds of the first-event error probability with Viterbi decoding

	$P_u \leq 1E-5$	$P_u \leq 1E-10$	$P_u \leq 1E-20$
Mode 1	-2.31	0.79	3.69
Mode 2	1.74	4.89	8.06
Mode 3	0.68	3.80	6.97
Mode 4	4.75	7.90	11.07
Mode 5	7.08	10.49	13.81
Mode 6	11.39	14.72	17.97
Mode 7	15.95	19.82	23.40
Mode 8	17.29	20.79	24.13

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 12 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 the physical-layer FEC (which is applied to the whole payload of the

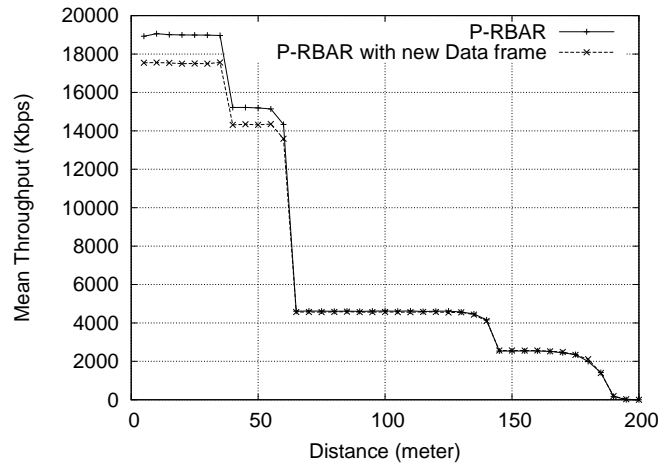


Figure 12: Modified frame format overhead.

physical-layer frame)⁴. The results are shown in Figure 13. 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.

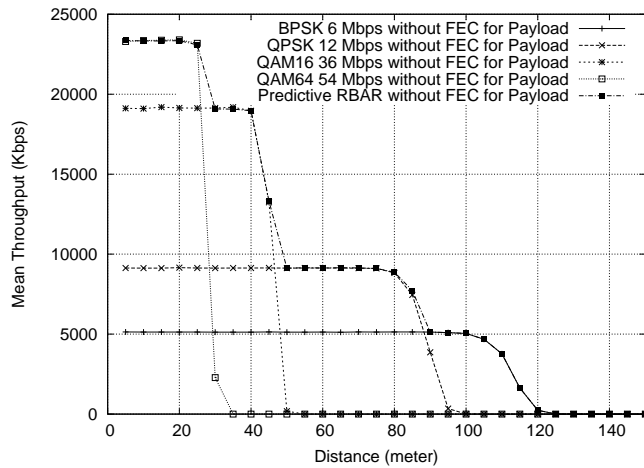


Figure 13: New transmission modes.

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:

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

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 payloads, $\epsilon = 1$

⁴The basic mode for sending headers, is still BPSK 6 Mbps with 1/2 FEC code rate.

and 10. Figure 14 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. Similar simulation has been done when we use QAM16 modulation with 36 Mbps data rate and FEC rate equal to 3/4, see Figure 15,

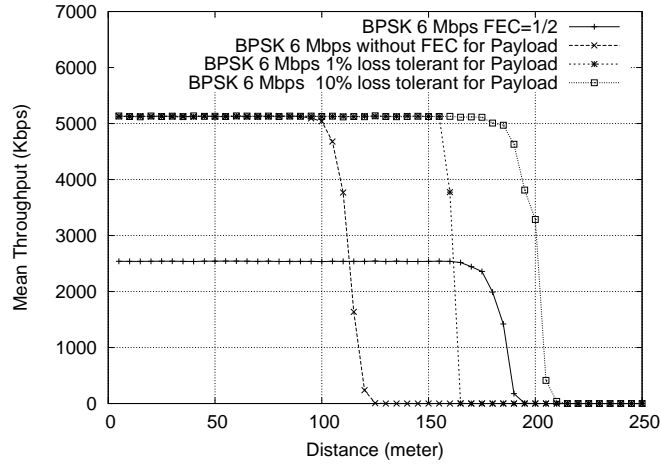


Figure 14: Mean throughput versus distance (basic mode) with and without loss tolerance.

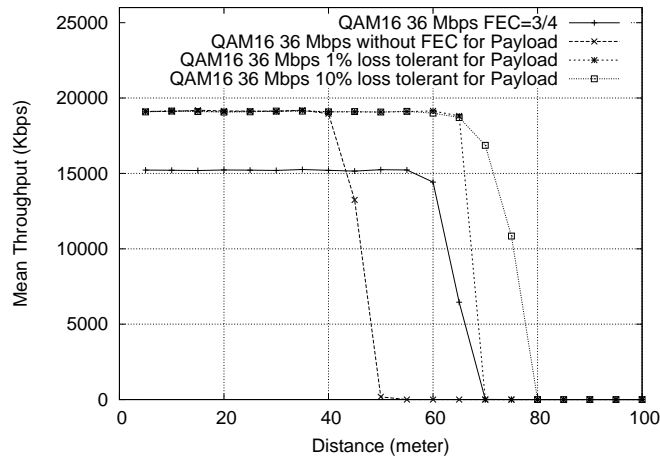


Figure 15: Mean throughput versus distance with and without loss tolerance for QAM16 36Mbps.

In order to implement the media-oriented mechanism for RBAR we propose to use precomputed thresholds in the receiver side based on different transmission modes and different LT information as we explained in Section 3.

In Section 3.1, we showed that thresholds are computed for different LT according to bit error rate. The corresponding values are shown in Table 5.

Figure 16, shows the performance with and without LT when using P-RBAR for automatic rate selection. New thresholds are used based on the LT in the payload, as we explained in Section 3.

In the following simulation we have used thresholds respecting to $P_u \leq 1E-5$ when FEC is used for payload (see Table 4). Two different thresholds are used respecting to $P_e \leq 1E-5$ and $P_e \leq 1E-2$

Table 5: SNR(dB) thresholds for different probability of bit error rate

	$P_e \leq 1E-6$	$P_e \leq 1E-5$	$P_e \leq 1E-2$
Mode 1	5.31	4.36	0.01
Mode 2	7.07	6.12	0.86
Mode 3	8.32	7.37	2.10
Mode 4	10.08	9.13	3.86
Mode 5	15.20	14.23	8.66
Mode 6	16.96	15.99	10.42
Mode 7	22.58	21.59	15.73
Mode 8	23.10	22.10	16.24

(see Table 5) when FEC is not used for payload and when the application tolerate some losses in the payload.

These results 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.

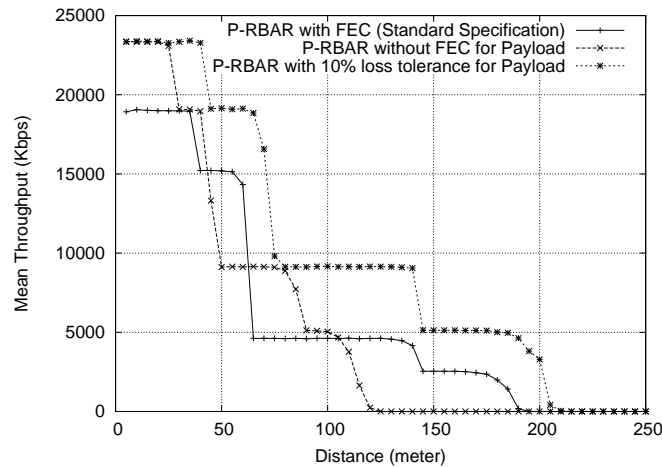


Figure 16: P-RBAR for LT applications.

5 Conclusions and Future Works

In this report we have introduced a media-oriented rate selection algorithm for 802.11 WLANs. Our mechanism uses both information from the physical channel and characteristics of loss tolerant applications to select the optimal PHY rate, modulation and FEC schemes. We have also proposed new transmission modes with less complexity that significantly increase application throughput and coverage range. Implementation issues are evaluated. The mechanism can be implemented with some minor changes, using 2 reserved bits in QoS control field and adding one byte to RTS packets. Also we have supposed that the MAC header should be sent with basic mode.

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.

Contents

1	Introduction	3
2	Background	3
2.1	802.11 Distributed Coordination Function	4
2.2	IEEE 802.11e	4
2.3	IEEE 802.11 Physical Layer	6
2.4	Rate Selection Algorithms	6
3	Media-Oriented Mode Selection	7
3.1	Algorithm Description	7
3.2	Implementation	8
4	Simulation Results	9
4.1	802.11a Simulation	9
4.2	Simulation of Media-Oriented Mechanism in 802.11a	10
5	Conclusions and Future Works	15

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Éditeur

INRIA - Domaine de Voluceau - Rocquencourt, BP 105 - 78153 Le Chesnay Cedex (France)

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ISSN 0249-6399