

IEEE 802.11 Wireless LANs: Performance Analysis and Protocol Refinement

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The IEEE 802.11 protocol is emerging as a widely used standard and has become the most mature technology for wireless local area networks (WLANs). In this paper, we focus on the tuning of the IEEE 802.11 protocol parameters taking into consideration, in addition to throughput efficiency, performance metrics such as the average packet delay, the probability of a packet being discarded when it reaches the maximum retransmission limit, the average time to drop a packet, and the packet interarrival time. We present an analysis, which has been validated by simulation that is based on a Markov chain model commonly used in the literature. We further study the improvement on these performance metrics by employing suitable protocol parameters according to the specific communication needs of the IEEE 802.11 protocol for both basic access and RTS/CTS access schemes. We show that the use of a higher initial contention window size does not considerably degrade performance in small networks and performs significantly better in any other scenario. Moreover, we conclude that the combination of a lower maximum contention window size and a higher retry limit considerably improves performance. Results indicate that the appropriate adjustment of the protocol parameters enhances performance and improves the services that the IEEE 802.11 protocol provides to various communication applications.

Keywords and phrases: IEEE 802.11, wireless LANs, DCF, packet delay, protocol tuning.

1. INTRODUCTION

During the past few years, the field of wireless local area networks (WLANs) has witnessed a massive development and has become one of the fastest growing areas in telecommunications and networking [1]. Continuing advances in wireless technology and mobile communications have equipped portable devices with wireless capabilities that allow networked communication even while a user is mobile. WLANs have found widespread use and have become an essential tool in many people's professional and personal life. To satisfy the

growing needs of wireless data networking, the IEEE working group proposed the 802.11 protocol family [2].

The IEEE 802.11 protocols have become the dominant standard for WLANs and can offer high data rates of 11 Mbit/s [3] and 54 Mbit/s [4]. The IEEE 802.11 standard specifies two different medium access control (MAC) mechanisms for WLANs; the contention-based distributed coordination function (DCF) and the polling-based point coordination function (PCF). The mandatory DCF supports asynchronous data transfer and best suits delay insensitive data whereas the optional PCF provides time bounded services (TBS). DCF employs a carrier sense multiple access with collision avoidance (CSMA/CA) access scheme using binary exponential backoff. Under DCF, data packets are transmitted through two access mechanisms, the basic access

and the request-to-send/clear-to-send (RTS/CTS) reservation scheme.

Many research efforts have been conducted on modeling the performance of DCF since the standardization of IEEE 802.11 MAC. Bianchi in [5] and Wu et al. in [6] use Markov chain models to analyze the throughput of 802.11 protocol. In particular, Bianchi assumes that packet retransmissions are unlimited and that a packet is being transmitted continuously until its successful reception. Wu in [6] extends Bianchi's analysis to include the finite packet retry limits as defined in the IEEE 802.11 standard [2]. In [7], we provide a new performance analysis of the 802.11 protocol, which is based on the extensively-used-in-the-literature Markov chain model of [6] and allows the calculation of the packet delay, the packet drop probability, and the packet drop time. Ziouva in [8] develops a Markov chain model that introduces an additional transition state to the models of [5, 6, 7] and actually allows stations to transmit consecutive packets without activating the backoff procedure.¹ This feature, which is not specified in any IEEE 802.11 standard, causes an unfair use of the medium since stations are not treated in the same way after a successful transmission. The proposed model in [8] lacks of any validation using simulation results and the calculation of average packet delay utilizes a very complicated approach since it calculates the average number of the collisions of a packet before its successful reception and the average time a station's backoff timer remains stopped.

Several other papers in the literature [9, 10, 11] have attempted to improve IEEE 802.11 performance by either modifying the backoff mechanism or by fine-tuning certain protocol parameters. Carvalho and Garcia-Luna-Aceves in [9] considered the impact of the minimum contention window (CW) size and the corresponding capacity improvement that is achieved when CW increases but not combined with packet retry limits and other protocol parameters. Cali et al. in [10] proposes a method of estimating the number of active stations via the number of empty slots and exploits the estimated value to tune the CW parameter based on a p -persistent version of the IEEE 802.11 protocol. Aad and Castelluccia in [11] suggests three different ways to enhance 802.11 performance; by scaling the CW based on the priority factor of each station or by giving each priority level with a different value of DIFS or different maximum packet length.

In this paper, we concentrate on the performance enhancement of IEEE 802.11 DCF by simply modifying specific protocol parameter values. In order to adjust the protocol parameters, the mathematical description of the system turns out to be extremely helpful in observing the effect on the considered performance metrics of any parameter changes made. Our work reports and explores several performance metrics such as the average packet delay, the packet drop probability, the average time to drop a packet, the packet in-

terarrival time, and the throughput efficiency. OPNET simulation results validate the accuracy of our performance analysis. Moreover, a performance comparison of (a) the proposed delay analysis in [8], (b) our validated delay analysis, and (c) simulation results, demonstrates that the analysis based on Wu's model, which takes into account packet retry limits, predicts very accurately DCF packet delay performance. We then propose a simple-to-implement appropriate tuning of the backoff algorithm for the basic access scheme (the conclusions are also applicable to the RTS/CTS scheme) depending on the specific communication requirements. The proposed fine-tuning does not depend on the employed access scheme or the packet size and aims to improve the services that the protocol provides to higher layers of the communication protocol stack.

2. DISTRIBUTED COORDINATION FUNCTION

In DCF basic access mode, a station with a packet to transmit monitors the medium activity. If the medium is idle, the station transmits the data packet. If the medium is sensed busy, the station waits until the medium becomes idle for more than a distributed interframe space (DIFS) time interval. The station then defers transmission for a randomly selected interval in order to minimize collisions and transmits the data packet. A station that receives a data packet replies by a positive acknowledgement packet (ACK) after a short interframe space (SIFS) interval. If the source station does not receive an ACK, the data packet is assumed to have been lost and a retransmission is scheduled. Each station maintains a station short retry count (SSRC) that has an initial value of zero for every new packet. The short retry count indicates the maximum number of retransmission attempts of a data packet when the basic access scheme is utilized.

In IEEE 802.11, a station waits a random backoff interval before initiating a packet transmission. The backoff timer value for each station is uniformly chosen in the interval $[0, W_i - 1]$ where W_i is the current CW size and i is the backoff stage. The backoff timer is decremented when the medium is idle, is frozen when the medium is sensed busy, and resumes only after the medium has been idle for longer than DIFS. A station initiates a packet transmission when the backoff timer reaches zero. The value of W_i depends on the number of failed transmissions of a packet; at the first transmission attempt, $W_0 = CW_{\min} = W$. After each retransmission due to a packet collision, W_i is doubled up to a maximum value, $W_{m'} = CW_{\max} = W2^{m'}$, where m' is the number of backoff stages. Once W_i reaches $CW_{\max}$, it will remain at this value until it is reset to $CW_{\min}$ in the following cases: (a) after the successful transmission of a data packet or (b) when SSRC reaches the short retry limit. When the short retry limit is reached, retry attempts will cease and the packet will be discarded. The SSRC is reset to 0 whenever an ACK is received in response to a data packet.

3. MATHEMATICAL MODELING

In this paper, we assume that the network consists of n contending stations and each station always has a packet

¹According to the authors of [8], this takes place when a station detects that its previous transmitted packet was successfully received and the channel is idle.

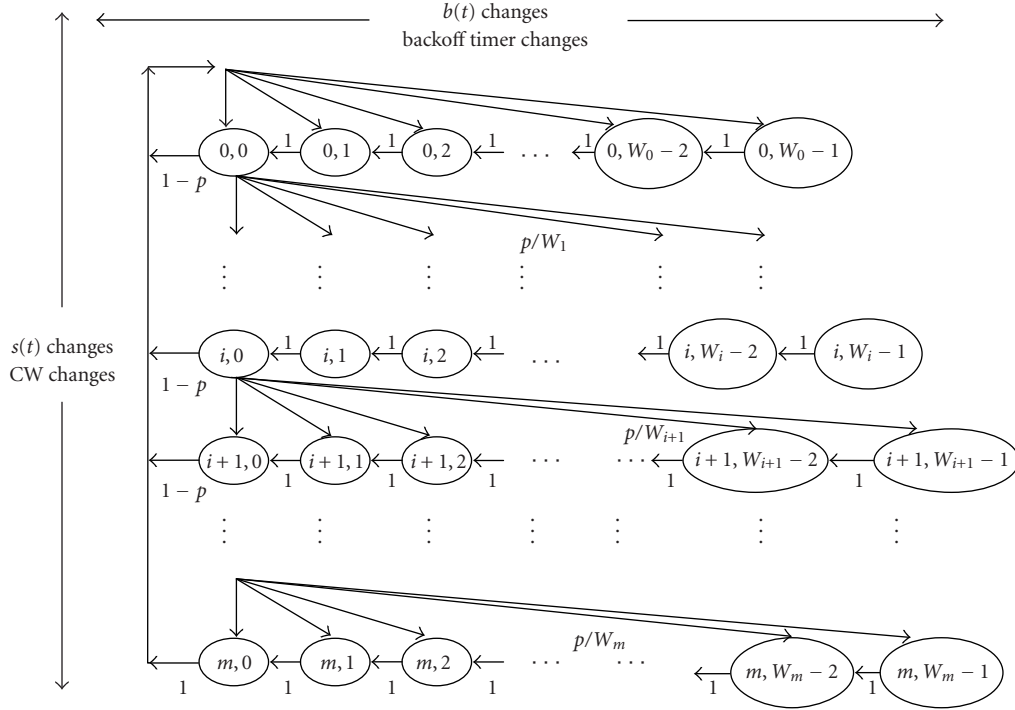


FIGURE 1: Markov chain model.

available for transmission. The main assumption of our model is that the collision probability of a data packet transmission is constant and independent of the number of collisions the packet has suffered in the past.

Let $b(t)$ and $s(t)$ be the stochastic processes representing the backoff timer and the backoff stage, respectively, for a given station at slot time t . The discrete-time Markov chain illustrated in Figure 1 is employed to model the bi-dimensional process $\{b(t), s(t)\}$. Let $b_{i,k} = \lim_{t \rightarrow \infty} P\{s(t) = i, b(t) = k\}$ be the stationary distribution of the Markov chain denoting the probability of a station to be in state (i, k) ,

where $i \in [0, m]$, $k \in [0, W_i - 1]$, and m is the station retry limit. By considering that $b_{i,0} = pb_{i-1,0}$, $i \in (0, m]$, we have the following relation for $b_{i,0}$:

$$b_{i,0} = p^i b_{0,0}, \quad 0 < i \leq m. \quad (1)$$

Following the same reasoning with [6, 7] and by means of the above Markov chain model, the probability τ that a station transmits a packet in a randomly chosen slot time is presented by (we consider the case of $m > m'$, which is usually the case)

$$\tau = \frac{2(1-2p)(1-p^{m+1})}{W(1-(2p)^{m'+1})(1-p) + (1-2p)[W2^{m'}p^{m'+1}(1-p^{m-m'}) + 1 - p^{m+1}]}. \quad (2)$$

The probability p that a transmitted packet encounters a collision is the probability that at least one of the $n - 1$ remaining stations transmits in the same slot time. If all stations transmit with probability τ , the conditional collision probability p is given by

$$p = 1 - (1 - \tau)^{n-1}. \quad (3)$$

Equations (2) and (3) form a nonlinear system with two unknowns τ and p . This nonlinear system can be solved

utilizing numerical methods and has a unique solution.²

4. PERFORMANCE ANALYSIS

Our performance analysis, as already shown in the previous section, includes the effect of packet retry limits and

²The full proof as well as additional details for the derived analysis can be found in the appendix.

considers the following metrics, which are good indicators for the performance of IEEE 802.11 WLANs. We consider throughput efficiency, average packet delay, probability of a packet being discarded when it reaches the maximum retransmission limit, the average time to drop a packet, and packet interarrival time.

4.1. Saturation throughput

Let P_{tr} be the probability that at least one station transmits a packet in a randomly selected slot time and P_s the probability that an occurring packet transmission is successful. For a wireless LAN of n contending stations, the probabilities P_{tr} and P_s are given by

$$\begin{aligned} P_{tr} &= 1 - (1 - \tau)^n, \\ P_s &= \frac{n\tau(1 - \tau)^{n-1}}{1 - (1 - \tau)^n}. \end{aligned} \quad (4)$$

Considering that a random slot is empty with probability $(1 - P_{tr})$ contains a successful transmission with probability $P_{tr}P_s$ and a collision with probability $P_{tr}(1 - P_s)$, the saturation throughput S is given by

$$S = \frac{P_{tr}P_s l}{E[\text{slot}]} = \frac{P_{tr}P_s l}{(1 - P_{tr})\sigma + P_{tr}P_s T_s + P_{tr}(1 - P_s)T_c}, \quad (5)$$

where $E[\text{slot}]$ is the average length of a slot time, l is the length of the transmitted packet, σ is the duration of an empty slot, T_s and T_c are the average durations the medium is sensed busy due to a successful transmission and a collision, respectively. We have

$$T_s = \text{DIFS} + T_{\text{header}} + T_{\text{DATA}} + \delta + \text{SIFS} + T_{\text{ACK}} + \delta. \quad (6)$$

In order to explicitly specify the value of the time interval T_c , we have to categorize stations in two groups: the listening (noncolliding) and the colliding stations. In the case of the “listening” stations, a packet collision will result in an error reported by the PHY (by utilizing the PHY-RXEND.indication) and the time interval T_c for those stations is equal to an extended interframe space (EIFS) after the packet transmission. For the “colliding” stations the time interval T_c is equal to an ACK_Timeout following the packet transmission. As it is specified in the IEEE 802.11 standard [2], the ACK_Timeout is equal to EIFS (almost equal since the latter is shorter by a slot time). Thus, the values of T_s and T_c , which both depend on the medium access mechanism, in the case of basic access are

$$T_s = T_c = \text{DIFS} + T_{\text{header}} + T_{\text{DATA}} + \delta + \text{SIFS} + T_{\text{ACK}} + \delta, \quad (7)$$

where T_{header} is the time required to transmit the MAC and the physical packet header, $T_{\text{DATA}} = l/C$ is the time required to transmit the packet data payload of l bits, when C is the data rate, $T_{\text{ACK}} = l_{\text{ACK}}/C_{\text{control}}$ is the time required to transmit the ACK packet of l_{ACK} bits, C_{control} is the control (base) rate at which the ACK packet is sent and δ is the propagation delay.

4.2. Packet drop probability

The packet drop probability is defined as the probability that a packet is dropped when the retry limit is reached. A packet is found in the last backoff stage m if it encounters m collisions in the previous stages and it will be discarded if it experiences another collision. Therefore, packet drop probability can be expressed as a function of the last backoff stage (by means of (1)) and the collision probability p as³

$$p_{\text{drop}} = \frac{b_{m,0}}{b_{0,0}} p = p^m p = p^{m+1}. \quad (8)$$

4.3. Average packet delay

The delay D for a successfully transmitted packet is defined to be the time interval from the time the packet is at the head of its MAC queue ready for transmission, until an acknowledgement for this packet is received. If a packet is dropped because it has reached the specified retry limit, the time delay for this packet will not be included in the calculation of the average packet delay since this packet is not successfully received.

The average packet delay $E[D]$ is given by

$$E[D] = E[X]E[\text{slot}], \quad (9)$$

where $E[X]$ is the average number of slot times required for a successful packet transmission and can be found by multiplying the number of slot times d_i the packet is delayed in each backoff stage by the probability q_i for the packet to utilize this backoff stage:

$$E[X] = \sum_{i=0}^m d_i q_i. \quad (10)$$

The average number of slot times d_i a station utilizes in the i stage (including the transmission slot) is given by

$$d_i = \frac{W_i + 1}{2}, \quad i \in [0, m]. \quad (11)$$

The probability q_i that a packet reaches the i backoff stage, provided that this packet is not discarded, is given by

$$q_i = \frac{(p^i - p^{m+1})}{1 - p^{m+1}}, \quad i \in [0, m] \quad (12)$$

since packets that are not dropped (with probability $1 - p^{m+1}$) arrive at the i stage with probability $(p^i - p^{m+1})$ (we have to deduct the probability p^{m+1} of dropped packets from the probability p^i of the total number of packets arriving at the i stage).

Combining (10), (11), and (12), $E[X]$ is given by

$$E[X] = \sum_{i=0}^m \left[\frac{(p^i - p^{m+1})((W_i + 1)/2)}{1 - p^{m+1}} \right]. \quad (13)$$

³Note that the packet drop probability is independent of the employed access scheme (basic access or RTS/CTS).

4.4. Average time to drop a packet

A packet is dropped when it reaches the last backoff stage and experiences another collision. The average time to drop a packet is equal to

$$E[D_{\text{drop}}] = E[X_{\text{drop}}]E[\text{slot}], \quad (14)$$

where $E[X_{\text{drop}}]$ is the average number of slot times required for a packet to experience $m + 1$ collisions in the $(0, 1, \dots, m)$ stages. Given that the average number of slot times a station defers in the i stage is $(W_i + 1)/2$, then $E[X_{\text{drop}}]$ is given by

$$\begin{aligned} E[X_{\text{drop}}] &= \sum_{i=0}^m \frac{W_i + 1}{2} \\ &= \frac{W(2^{m'+1} - 1) + W2^{m'}(m - m') + (m+1)}{2}. \end{aligned} \quad (15)$$

4.5. Packet interarrival time

The packet interarrival time is defined as the time interval between two successful packet receptions at the receiver and can be simply obtained from throughput:

$$E[D_{\text{inter}}] = \frac{1}{S/n}. \quad (16)$$

Using the same reasoning with (9), the packet interarrival time $E[D_{\text{inter}}]$ is also given by

$$E[D_{\text{inter}}] = \left(\sum_{j=0}^{\infty} p^{j(m+1)} \sum_{i=0}^m p^i \frac{W_i + 1}{2} \right) E[\text{slot}], \quad (17)$$

which after some algebra reaches (16).

Intuitively, the average packet delay, interarrival time, and drop time are related by

$$E[D] = E[D_{\text{inter}}] - \frac{p_{\text{drop}}}{1 - p_{\text{drop}}} E[D_{\text{drop}}], \quad (18)$$

where $E[D_{\text{inter}}]$ is given by (16) or (17), p_{drop} is given by (8), and $E[D_{\text{drop}}]$ is given by (14). The expression $p_{\text{drop}}/(1 - p_{\text{drop}}) = p^{m+1}/(1 - p^{m+1})$ represents the average number of dropped packets needed for a successful transmission. The expression in (18) is of key importance since it gives insights of the delay characteristics of the IEEE 802.11 backoff mechanism and relates the average packet delay with the packet interarrival time, the packet drop probability, and the average time to drop a packet.

5. MODEL VALIDATION

The mathematical analysis presented in this paper is validated by comparing analytical with simulation results obtained using our IEEE 802.11 simulator. This IEEE 802.11 simulator is developed using the OPNET modeler communication networks modeling and simulation software package. OPNET modeler is an event-driven simulator and provides a powerful graphical tool to display simulation statistics.

In fact, our OPNET 802.11 simulator emulates the real operation of a wireless station as closely as possible, by implementing the collision avoidance procedures and all parameters such as packet transmission times, propagation delays, turnaround times, and so forth. The simulator closely follows all timer values and packet element transmission times defined by IEEE 802.11 specifications. Furthermore, we have suitably modified the model of the IEEE 802.11 wireless station provided in the standard library of OPNET in order to employ saturation conditions, that is, all stations always have a packet ready for transmission.

The Markov chain analysis presented in the previous sections is independent of physical layer parameters and can be applied to all IEEE 802.11 PHY standards. The parameters used in both the analytical model and our simulations follow the parameters in [6, 7] and are summarized in Table 1. The system parameter values are those specified for the direct spread sequence spectrum (DSSS) physical layer utilized in IEEE 802.11b [3].

Figures 2 and 3 confirm the accuracy of the considered assumptions in the mathematical analysis.⁴ The figures provide performance results (throughput efficiency, packet delay, packet drop time, and packet drop probability) versus the number of contending stations. Figure 2 depicts an almost exact match observed between analytical results (lines) and simulation outcome (symbols) illustrating that the analytical model that considers retry limits predicts very accurately DCF throughput performance, a conclusion not clearly drawn in [6] which added packet retry limits in the analytical model in [5]. Figure 2 also displays packet delay calculated using our delay analysis as well as Ziouva's model [8] against OPNET simulation results. The performance comparison shows that our packet delay analysis gives results in high agreement with OPNET simulations. We can observe that the model in [8], which is less conformant to the IEEE 802.11 standard than our model, causes a high overestimation of packet delay due to the adoption of the additional transition state and the absence of packet retry limits. Figure 3 also validates our analysis for the other two considered performance metrics: packet drop time and drop probability.

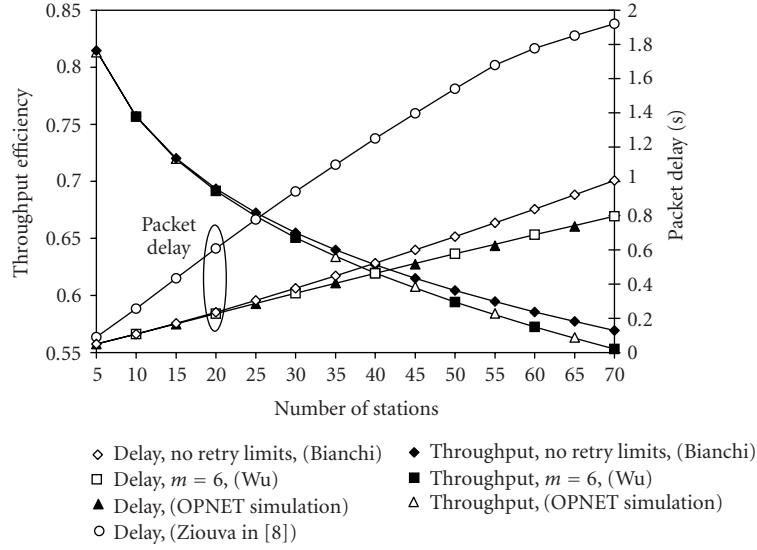
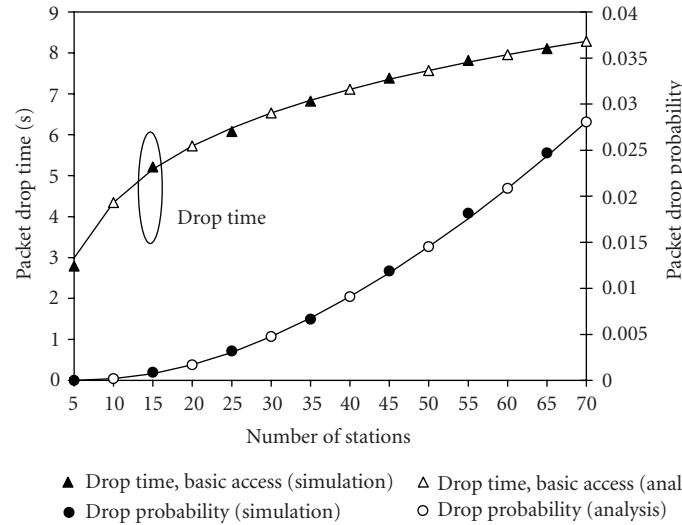
6. TUNING OF PROTOCOL PARAMETERS AND PERFORMANCE RESULTS

There are a variety of performance requirements according to the various communication needs or application desires. For example, time bounded applications that exchange query-like messages, require low packet loss and low delivery delay. Conversely, applications that provide delay insensitive services (i.e., email, ftp) are not concerned much with packet timely deliverance and maximising throughput performance is of prime importance in this case. Additionally, there are many applications that lie somewhere in the middle and may

⁴Note that simulation results are acquired with a 95% confidence interval lower than 0.002

TABLE 1: DSSS system parameters in IEEE 802.11b.

Parameter	Value	Parameter	Value
Slot time, σ	$20 \mu\text{s}$	Packet payload, l	1023 or 1500 bytes
MAC header, l_{MAC}	272 bits	DIFS	$50 \mu\text{s}$
PHY header, l_{PHY}	192 bits	SIFS	$10 \mu\text{s}$
Data header time, T_{header}	$(l_{\text{PHY}} + l_{\text{MAC}})/C_{\text{control}}$	Minimum CW, W_0	32
ACK packet, l_{ACK}	$112 \text{ bits} + l_{\text{PHY}}$	Number of CW sizes, m'	5
Channel bit rate, C	11 Mbit/s	Short retry limit, m	6
Control rate, C_{control}	1 Mbit/s	Propagation delay, δ	$1 \mu\text{s}$

FIGURE 2: Throughput efficiency and packet delay: analysis versus simulation ($l = 1023$ bytes).FIGURE 3: Packet drop time and packet drop probability: analysis versus simulation ($l = 1023$ bytes).

demand low delivery delay but will not be sensitive to some loss of packets or may demand low loss but not small delay. For example, multimedia applications are not able to tolerate

high delay or jitter but may tolerate some packet loss whereas HTTP-like applications can tolerate delay but require minimum data loss.

TABLE 2: Packet delay and throughput efficiency for a small network size ($l = 1500$ bytes).

Number of stations	IEEE 802.11 standard $W = 32, m = 6, m' = 5$		$W = 64, m = 6, m' = 5$	
	Packet delay (s)	Throughput efficiency	Packet delay (s)	Throughput efficiency
$n = 2$	0.003779	0.577334	0.004049	0.538847
$n = 3$	0.005664	0.577849	0.005843	0.560091
$n = 4$	0.007624	0.572318	0.007683	0.567978
$n = 5$	0.009647	0.565203	0.009564	0.570292
$n = 6$	0.011722	0.557878	0.011485	0.569902

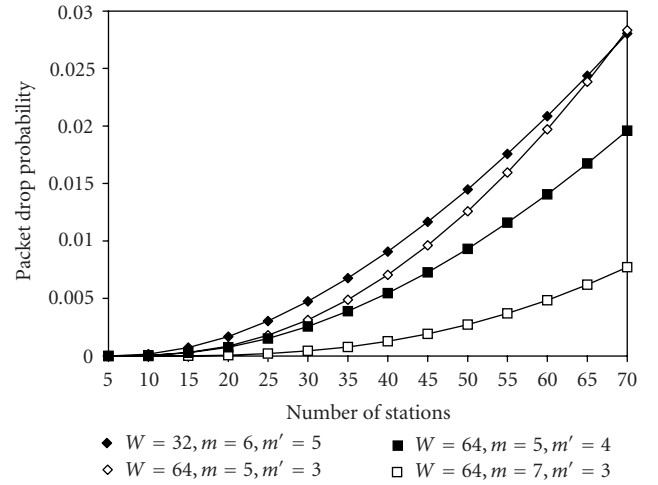
In order to fulfil specific communication needs, we propose the adjustment of certain protocol parameters to different values than those proposed by the IEEE standard. Three parameters are being examined: the initial contention size (W), the packet retry limit (m), and the number of back-off stages (m'). Our performance analysis examines the following metrics as good indicators for the performance of the IEEE 802.11 protocol, namely, the throughput efficiency, the average packet delay, the packet drop probability as well as the average time to drop a packet.

By employing the analytical model presented previously, various sets of protocol parameter values have been examined and compared with parameter values that the IEEE 802.11 standard proposes in order to identify potential improvements on protocol performance. After an extensive performance study, we have identified three sets of parameter values. Each set of parameter values achieves better performance on some particular metrics and it can be employed according to the specific communication needs. For example, one set of parameter values can significantly improve the throughput efficiency whereas another combination of parameters can considerably reduce the packet drop probability or the packet drop time.

The following three sets of parameter values that are being employed for the basic access scheme, for the case of “long” packets of $l = 1500$ bytes⁵ and compared with the values that the IEEE 802.11 protocol proposes ($W = 32, m = 6, m' = 5$) are

- (a) $W = 64, m = 5, m' = 4$,
- (b) $W = 64, m = 5, m' = 3$,
- (c) $W = 64, m = 7, m' = 3$.

In all considered cases, we increase the value of W to reduce the number of collisions. In the first case, the $CW_{\max}$ value that the standard proposes ($CW_{\max} = 1024$) is utilized by decreasing m' to 4; a lower retry limit ($m = 5$) is considered sufficient since increasing W to 64 reduces the collision probability. In the second set, we study the effect of reducing $CW_{\max}$ to 512 by decreasing m' to 3; this set is expected to

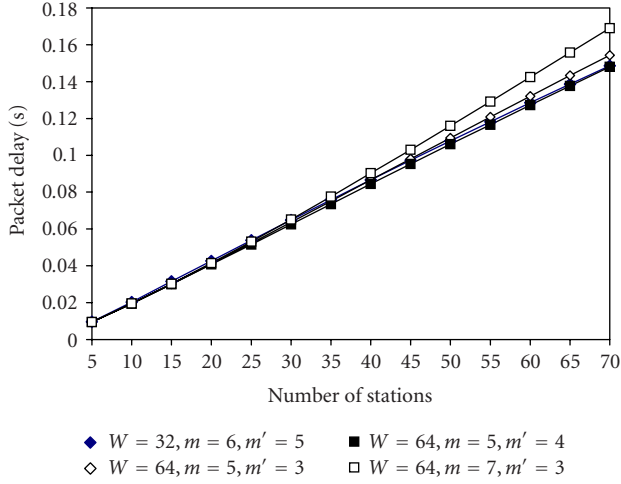
FIGURE 4: Packet drop probability against number of stations ($l = 1500$ bytes).

improve the average packet delay. Finally, in the last set, the retry limit is increased to the value of 7. As a result, a contending station utilizes two more times the (relatively) small last backoff stage ($CW_{\max} = 512$) aiming to reduce the packet drop probability while keeping a fairly low packet delay.

At a first glance, it might seem that the choice of a higher value for the initial CW size ($W = 64$) comparing to the value of the standard ($W = 32$) will cause a performance decrease in a small network scenario. A closer study to the case of a small network size ($2 \leq n \leq 6$) was performed and Table 2 presents the packet delay and throughput efficiency for the two different values of the initial contention window W . The table illustrates that the adjustment of W to a higher value does not cause a considerable effect on both the packet delay and throughput efficiency for very small networks; on the contrary performance is improved in networks with five or more contending stations.

The efficiency of each set of parameter values on the packet drop probability is explored in Figure 4 against the number of contending stations. When the standard proposed values are employed, a packet suffers the highest drop probability comparing to the other three cases. The choice of a higher W value improves the drop probability since fewer

⁵Results for the RTS/CTS scheme and other packet sizes such as “short” VoIP packets of $l = 200$ bytes have reached exactly the same conclusions, denoting that the proposed improvement does not depend on the employed access scheme or the packet payload size.

FIGURE 5: Packet delay against number of stations ($l = 1500$ bytes).

collisions are taking place. When $W = 64, m = 5, m' = 3$ are employed, the packet drop probability increases rapidly and gradually attains the same value with the standard proposed values in a large network scenario ($n = 70$). This is justified by noting that employing $W = 64$ and $m' = 3$, the maximum value of the CW size will be lower ($CW_{\max} = 512$) compared to the one that the IEEE standard proposes ($CW_{\max} = 1024$) resulting in an increased number of collisions when the number of contending stations is high. The lowest packet drop probability is achieved when $W = 64, m = 7$, and $m' = 3$ since the packet drop probability is reduced up to 75% compared to the IEEE standard proposed values despite of the decrease of $CW_{\max}$.

Figure 5 depicts that the packet delay increases when the network size grows in all cases due to the higher number of collisions. The figure also shows that the packet delay is not significantly affected by the employment of different parameter values. The only exception is when $W = 64, m = 7, m' = 3$, the packet delay increases faster than in the other cases when $n > 35$ and a packet experiences an increase on delay of up to 10% in a large network ($n = 70$). However, by means of Figure 4 the situation is easily explained since a larger number of packets are transmitted successfully and not discarded. The small increase of the packet delay is the small price we pay for significantly decreasing the packet drop probability.

Figure 6 plots the average time to drop a packet when it reaches the maximum retransmission limit against the number of contending stations. For all sets of parameter values, the packet drop time increases when the network size increases. The figure shows that the employment of any of the considered sets of parameter values, as compared to the IEEE standard parameters, results in a significant improvement on the packet drop time. The highest packet drop time is attained using the parameter values suggested in the standard, whereas the case of $W = 64, m = 5, m' = 3$ achieves the lowest packet drop time with a reduction of about 40% for a large network size ($n = 70$).

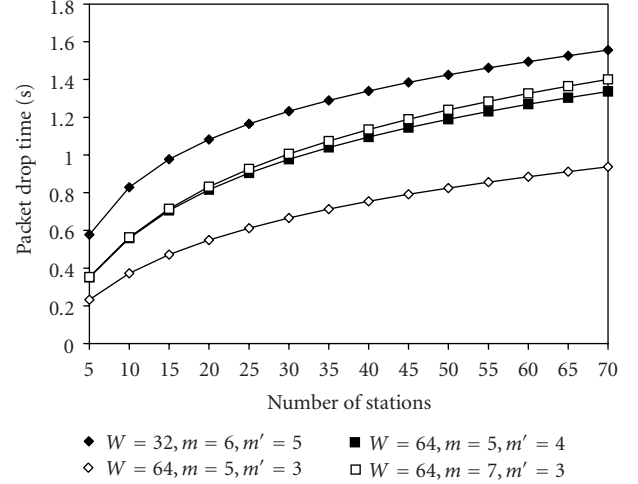
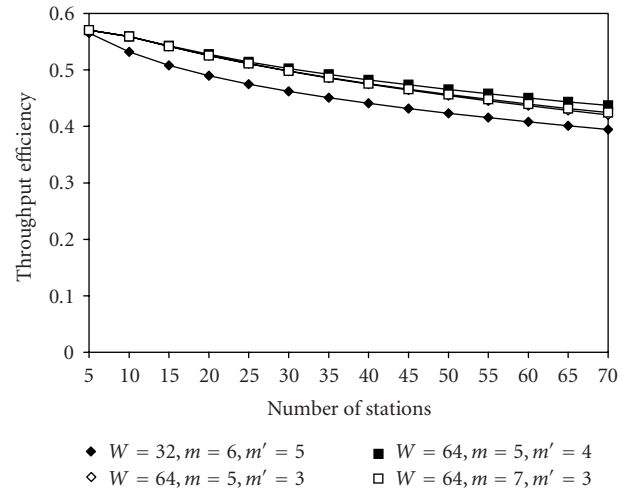
FIGURE 6: Packet drop time against number of stations ($l = 1500$ bytes).FIGURE 7: Throughput efficiency against number of stations ($l = 1500$ bytes).

Figure 7 examines the throughput efficiency that each considered set of parameter values achieves with varying the number of contending stations. When any of the proposed value sets is employed, the achievable throughput efficiency is higher compared to the standard parameter values mainly because the larger W value decreases the number of collisions. Especially when $W = 64, m = 5, m' = 4$, the increase on throughput can be up to 10% compared to the case when the standard parameter values are employed.

Finally, Figure 8 studies packet interarrival time, which is defined as the time interval between two successful packet receptions at the receiver. As expected, packet interarrival time for the standard parameter values is considerably higher than any other case. This can be easily justified by noting that packet interarrival time also includes the time for packets that have been discarded; this time is much greater for the case of $W = 32, m = 6, m' = 5$ due to the high drop probability values (Figure 4).

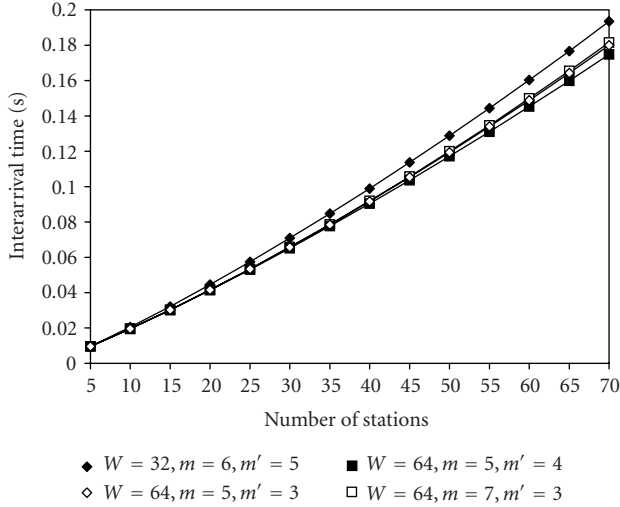


FIGURE 8: Packet interarrival time against number of stations ($l = 1500$ bytes).

Performance results reported in the previous figures show that when ($W = 64, m = 5, m' = 4$), lower packet drop probability, packet drop time, packet interarrival time, and better throughput performance are achieved compared to the values proposed by the standard. When the $CW_{\max}$ is decreased to a lower value ($CW_{\max} = 512$) for the same retry limit ($m = 5$), we attain the lowest packet drop time compared to any other case but the drop probability increases considerably. On the contrary, the adjustment of the retry limit to a higher value ($W = 64, m = 7, m' = 3$) results in the lowest packet drop probability and a small increase of packet drop time and delay due to the larger number of packets not being discarded and transmitted successfully. Each combination of parameters achieves an improved performance on some specific metrics compared to the standard proposed values and the choice of which set of protocol parameters should be employed depends on the specific communication requirements.

7. CONCLUSIONS

In this paper, we have focused on the performance enhancement of the IEEE 802.11 MAC protocol using several performance metrics such as the average packet delay, the packet drop probability, the average time to drop a packet, the packet interarrival time, and the throughput efficiency. Performance results obtained from our analysis fully agree with OPNET simulations confirming the improvements in accuracy when retry limits are considered. We also compared throughput and delay results for different models presented in the literature. With the infinite retry limit model [5], performance results deviate from simulations as the number of contending stations increases. Moreover, for the model [8] based on a different operational mode of IEEE 802.11 MAC results revealed that it overestimates packet delay performance.

We have also examined the effect of the initial contention window size on performance by employing a higher value ($W = 64$) compared to the standard proposed value ($W = 32$). Results indicate that this adjustment does not considerably degrade performance in very small WLANs but improves performance in networks with five or more contending stations. Based on performance results for the basic access scheme (the same conclusions are derived for the RTS/CTS scheme), we have proposed an appropriate tuning of the backoff algorithm to improve the services that the IEEE 802.11 protocol provides. We have shown that the high value of $CW_{\max}$ that the IEEE standard has proposed could be safely lowered and when combined with a higher retry limit, then the performance can be improved. Finally, we have proposed three sets of parameter values for initial contention window size, retry limit, and number of backoff stages and we have concluded that each proposed set achieves better performance on particular metrics and it could be employed to match specific communication needs.

APPENDIX

Let $b_{i,k} = \lim_{t \rightarrow \infty} P\{s(t) = i, b(t) = k\}$ be the stationary distribution of this Markov chain, where $i \in [0, m]$, $k \in [0, W_i - 1]$. Based on the two-dimensional Markov chain illustrated in Figure 1 and by considering that $b_{1,0} = p \cdot b_{0,0}$ and $b_{2,0} = p \cdot b_{1,0} = p^2 \cdot b_{0,0}$, we have the following relation for $b_{i,0}$:

$$b_{i,0} = p b_{i-1,0} = p^i b_{0,0}, \quad 0 < i \leq m. \quad (A.1)$$

Owing to chain regularities and by means of equation (A.1), all $b_{i,k}$ values are expressed as a function of $b_{0,0}$ and p as

$$b_{i,k} = \frac{W_i - k}{W_i} \cdot b_{i,0}, \quad 0 \leq i \leq m, \quad 0 \leq k \leq W_i - 1. \quad (A.2)$$

Applying the normalization condition for this stationary distribution

$$\begin{aligned} 1 &= \sum_{i=0}^m \sum_{k=0}^{W_i-1} b_{i,k} = \sum_{i=0}^m b_{i,0} \cdot \sum_{k=0}^{W_i-1} \frac{W_i - k}{W_i} \\ &= \sum_{i=0}^m b_{i,0} \cdot \frac{W_i + 1}{2} = \sum_{i=0}^m p^i \cdot b_{0,0} \cdot \frac{W_i + 1}{2} \\ &= \frac{b_{0,0}}{2} \cdot \left(\sum_{i=0}^m p^i \cdot W_i + \sum_{i=0}^m p^i \right), \end{aligned} \quad (A.3)$$

from which

$$b_{0,0} = \frac{2}{\left(\sum_{i=0}^m p^i \cdot W_i + \sum_{i=0}^m p^i \right)}, \quad (A.4)$$

and after some algebra,

$$b_{0,0} = \frac{2(1-2p)(1-p)}{W(1-(2p)^{m'+1})(1-p) + (1-2p)[W2^{m'}p^{m'+1}(1-p^{m-m'}) + 1 - p^{m+1}]}. \quad (\text{A.5})$$

By utilizing the Markov chain model, the probability τ that a station transmits a packet in a randomly chosen slot time is equal to

$$\tau = \sum_{i=0}^m b_{i,0} = \sum_{i=0}^m p^i \cdot b_{0,0} = b_{0,0} \cdot \frac{1-p^{m+1}}{(1-p)} \quad (\text{A.6})$$

and $b_{0,0}$ can be acquired from (A.5). From (A.6), we observe that the transmission probability τ depends on the conditional probability p , which is defined as the probability that a transmitted packet collides and is given by

$$p = 1 - (1 - \tau)^{n-1}. \quad (\text{A.7})$$

As we stated before, (A.6) and (A.7) represent a nonlinear system with two unknowns τ and p . This nonlinear system,

which has a unique solution, can be solved utilizing numerical methods evaluating t and p for a certain W , m , and m' combination. Since the system of the two equations is different from the one in [5], a detailed proof of the uniqueness of this solution is derived next.

Equation (A.7) can be rewritten as

$$\tau^*(p) : \tau = 1 - (1 - p)^{1/(n-1)}. \quad (\text{A.8})$$

The function $\tau^*(p)$ is a continuous and monotone increasing function in the range $p \in (0, 1)$. It increases from $\tau^*(0) = 0$ to $\tau^*(1) = 1$. Function $\tau(p)$ given by (A.6) is also continuous in the same range;⁶ continuity in correspondence of the critical value $p = 1/2$ is simply proven by using (A.5) as follows:

$$\begin{aligned} b_{0,0} &= \frac{2}{\sum_{i=0}^m (1/2)^i W_i + \sum_{i=0}^m (1/2)^i}, \\ &= \frac{2}{\left(\sum_{i=0}^{m'} (1/2)^i (2^i W) + \sum_{i=m'+1}^m (1/2)^i (2^{m'} \cdot W) + (1 - (1/2)^{m+1})/(1 - 1/2) \right)}, \\ &= \frac{2}{\left(\sum_{i=0}^{m'} W + (2^{m'} \cdot W) \sum_{i=m'+1}^m (1/2)^i + (1 - (1/2)^{m+1})/(1/2) \right)}, \\ &= \frac{2}{(W(m' + 1) + (2^{m'} \cdot W)((1 - (1/2)^{m-m'})/(1 - 1/2))(1/2)^{m'+1} + (1 - (1/2)^{m+1})/(1/2))}, \\ &= \frac{2}{(W(m' + 1) + W((2^{m-m'} - 1)/2^{m-m'})/(1/2))(1/2) + ((2^{m+1} - 1)/2^{m+1})/(1/2)}, \\ &= \frac{2}{(W(m' + 1) + W((2^{m-m'} - 1)/2^{m-m'}) + (2^{m+1} - 1)/2^m)}. \end{aligned} \quad (\text{A.9})$$

Therefore, when $p = 1/2$, (A.6) becomes

$$\begin{aligned} \tau\left(\frac{1}{2}\right) &= \sum_{i=0}^m b_{i,0} = \sum_{i=0}^m \left(\frac{1}{2}\right)^i b_{0,0} = \frac{2^{m+1} - 1}{2^m} b_{0,0} \\ &= \frac{2^{m+1} - 1}{2^{m-1}(W(m' + 1) + W((2^{m-m'} - 1)/2^{m-m'}) + (2^{m+1} - 1)/2^m)}. \end{aligned} \quad (\text{A.10})$$

Moreover, when $p = 1$ and by means of (A.5), we have

$$\begin{aligned}
 b_{0,0} &= \frac{2}{\left(\sum_{i=0}^m W_i + \sum_{i=0}^m 1^i\right)} \\
 &= \frac{2}{\left(\sum_{i=0}^{m'} (2^i \cdot W) + \sum_{i=m'+1}^m (2^{m'} \cdot W) + (m+1)\right)} \\
 &= \frac{2}{\left(\sum_{i=0}^{m'} (2^i \cdot W) + 2^{m'} \cdot W(m - m') + (m+1)\right)} \\
 &= \frac{2}{W((1 - 2^{m'+1})/(1 - 2)) + 2^{m'} W(m - m') + (m+1)} \\
 &= \frac{2}{W(2^{m'+1} - 1) + 2^{m'} W(m - m') + (m+1)}. \tag{A.11}
 \end{aligned}$$

Therefore, when $p = 1$, (A.6) becomes

$$\begin{aligned}
 \tau(1) &= \sum_{i=0}^m b_{i,0} = \sum_{i=0}^m b_{0,0} = (m+1)b_{0,0} \\
 &= \frac{2(m+1)}{W(2^{m'+1} - 1) + 2^{m'} W(m - m') + (m+1)}. \tag{A.12}
 \end{aligned}$$

Function $\tau(p)$ is continuous and monotone decreasing in the range $p \in (0, 1)$ since it decreases from $\tau(0) = 2/(W+1)$ to $\tau(1)$ given by (A.12). Uniqueness of the solution is proven by considering that $\tau(0) > \tau^*(0)$ and $\tau(1) < \tau^*(1)$.

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⁶Note that if $p = 1$ or $p = 1/2$, the expression for τ in (A.6) cannot be used.

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Special Issue on Quality of Service in Mobile Ad Hoc Networks

Call for Papers

Mobile ad hoc networking is a challenging task due to a lack of resources residing in the network as well as frequent changes in network topology. Although much research has been directed to supporting QoS in the Internet and traditional wireless networks, present results are not suitable for mobile ad hoc network (MANET). QoS support for mobile ad hoc networks remains an open problem, drawing interest from both academia and industry under military and commercial sponsorship. MANETs have certain unique characteristics that pose several difficulties in provisioning QoS, such as dynamically varying network topology, lack of precise state information, lack of central control, error-prone shared radio channels, limited resource availability, hidden terminal problems, and insecure media, and little consensus yet exists on which approaches may be optimal. Future MANETs are likely to be "multimode" or heterogeneous in nature. Thus, the routers comprising a MANET will employ multiple, physical-layer wireless technologies, with each new technology requiring a multiple-access (MAC) protocol for supporting QoS. Above the MAC layer, forwarding, routing, signaling, and admission control policies are required, and the best combination of these policies will change as the underlying hardware technology evolves.

The special issue solicits original papers dealing with state-of-the-art and up-to-date efforts in design, performance analysis, implementation and experimental results for various QoS issues in MANETs. Fundamental research results as well as practical implementations and demonstrators are encouraged.

Topics of interest include (but are not limited to):

- QoS models and performance evaluation of MANET
- QoS resource reservation signaling
- Various QoS routing protocols
- Flexible MAC protocols
- Robust modeling and analysis of MANET resource management
- Dynamic and hybrid resource allocation schemes
- Resource control and multimedia QoS support
- Channel characterization
- QoS management and traffic engineering

- Tools and techniques for MANET measurement and simulation
- Adaptive QoS provisioning issues
- Information assurance and reliability in MANET

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Special Issue on CMOS RF Circuits for Wireless Applications

Call for Papers

Advanced concepts for wireless communications present a vision of technology that is embedded in our surroundings and practically invisible, but present whenever required. From established radio techniques like GSM, 802.11, or Bluetooth to more emerging ones like ultra-wideband (UWB) or smart dust moats, a common denominator for future progress is underlying CMOS technology. Although the use of deep-submicron CMOS processes allows for an unprecedented degree of scaling in digital circuitry, it complicates implementation and integration of traditional RF circuits. The explosive growth of standard cellular radios and radically different new wireless applications makes it imperative to find architectural and circuit solutions to these design problems.

Two key issues for future silicon-based systems are scale of integration and ultra-low power dissipation. The concept of combining digital, memory, mixed-signal, and RF circuitry on one chip in the form of System-on-Chip (SoC) has been around for a while. However, the difficulty of integrating heterogeneous circuit design styles and processes onto one substrate still remains. Therefore, System-in-Package (SiP) concept seems to be gaining more acceptance.

While it is true that heterogeneous circuits and architectures originally developed for their native technologies cannot be effectively integrated “as is” into a deep-submicron CMOS process, one might ask the question whether those functions can be ported into more CMOS-friendly architectures to reap all the benefits of the digital design and flow. It is not predestined that RF wireless frequency synthesizers be always charge-pump-based PLLs with VCOs, RF transmit up-converters be I/Q modulators, receivers use only Gilbert cell or passive continuous-time mixers. Performance of modern CMOS transistors is nowadays good enough for multi-GHz RF applications.

Low power has always been important for wireless communications. With new developments in wireless sensor networks and wireless systems for medical applications, the power dissipation is becoming a number one issue. Wireless sensor network systems are being applied in critical applications in commerce, healthcare, and security. These sys-

tems have unique characteristics and face many implementation challenges. The requirement for long operating life for a wireless sensor node under limited energy supply imposes the most severe design constraints. This calls for innovative design methodologies at the circuit and system level to address this rigorous requirement.

Wireless systems for medical applications hold a number of advantages over wired alternatives, including the ease of use, reduced risk of infection, reduced risk of failure, reduced patient discomfort, enhanced mobility, and lower cost. Typically, applications demand expertise in multiple disciplines, varying from analog sensors to digital processing cores, suggesting opportunities for extensive hardware integration.

The special issue will address the state of the art in CMOS design in the context of wireless communication for 3G/4G cellular telephony, wireless sensor networks, and wireless medical applications.

Topics of interest include (but are not limited to):

- Hardware aspects of wireless networks
- Wireless CMOS circuits for healthcare and telemedicine
- Modulation schemes for low-power RF transmission
- RF transceiver architectures (low IF, direct conversion, super-regenerative)
- RF signal processing
- Phase-locked loops (PLLs)
- Digitally controlled oscillators
- LNAs, mixers, charge pumps, and VCOs in CMOS
- System-on-Chip (SoC) and System-in-Package (SiP) implementations
- RF design implementation challenges in deep-submicron CMOS processes

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Special Issue on Ultra-Wideband (UWB) Communication Systems—Technology and Applications

Call for Papers

The opening of unlicensed frequency band between 3.1 GHz and 10.6 GHz (7.5 GHz) for indoor wireless communication systems by the Federal Communications Commission (FCC) spurred the development of ultra-wideband (UWB) communications. Several wireless personal area networking (WPAN) products have been demonstrated recently. These products implement one of the two leading proposals to the IEEE 802.15.3a High-Speed WPAN Standards Committee. On the other hand, the IEEE 802.15.4a Standards Committee is focusing on low power, low bit rate applications, emphasizing accurate localization. This flurry of activity has demonstrated the feasibility of high-bit-rate and low-bit-rate/low-power UWB communications. Further improvement in UWB transmission speed and reductions in power consumption and UWB transceiver cost require a comprehensive investigation of UWB communications that simultaneously addresses system issues, analog and digital implementation constraints, and RF circuitry limitations. In the application area, coexistence with other wireless standards plays an important role.

The aim of this special issue is to present recent research in UWB communication systems with emphasis on future applications in wireless communications. Prospective papers should be unpublished and present novel innovative contributions from either a methodological or an application perspective.

Suggested topics include (but are not limited) to:

- UWB channel modeling and measurement
- High-bit-rate UWB communications
- UWB modulation and multiple access
- Synchronization and channel estimation
- Pulse shaping and filtering
- UWB transceiver design and signal processing
- Interference and coexistence
- Ultra-low-power UWB transmission
- MIMO-UWB
- Multiband UWB
- Spectral management

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- Ranging and positioning
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Special Issue on Wireless Network Security

Call for Papers

Recent advances in wireless network technologies have rapidly developed in recent years, as evidenced by wireless location area networks (WLANs), wireless personal area networks (WPANs), wireless metropolitan area networks (WMANs), and wireless wide area networks (WWANs), that is, cellular networks. A major impediment to their deployment, however, is wireless network security. For example, the lack of data confidentiality in wired equivalent privacy (WEP) protocol has been proven, and newly adopted standards such as IEEE 802.11i robust security network (RSN) and IEEE 802.15.3a ultra-wideband (UWB) are not fully tested and, as such, may expose unforeseen security vulnerabilities. The effort to improve wireless network security is linked with many technical challenges including compatibility with legacy wireless networks, complexity in implementation, and cost/performance trade-offs. The need to address wireless network security and to provide timely, solid technical contributions establishes the motivation behind this special issue.

This special issue will focus on novel and functional ways to improve wireless network security. Papers that do not focus on wireless network security will not be reviewed. Specific areas of interest in WLANs, WPANs, WMANs, and WWANs include, but are not limited to:

- Attacks, security mechanisms, and security services
- Authentication
- Access control
- Data confidentiality
- Data integrity
- Nonrepudiation
- Encryption and decryption
- Key management
- Fraudulent usage
- Wireless network security performance evaluation
- Wireless link layer security
- Tradeoff analysis between performance and security
- Authentication and authorization for mobile service network
- Wireless security standards (IEEE 802.11, IEEE 802.15, IEEE 802.16, 3GPP, and 3GPP2)

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Special Issue on Radio Resource Management in 3G+ Systems

Call for Papers

The 3G+ wireless systems can be characterized by aggregate bit rates in the range of Mbps, QoS support for interactive multimedia services, global mobility, service portability, enhanced ubiquity, and larger user capacity. All digital entirely packet-switched radio networks involving hybrid networking and access technologies are envisioned in 3G+ systems. In such systems, radio resource management (RRM) plays a major role in the provision of QoS and efficient utilization of scarce radio resources. With the required support for multimedia services to multiple users over diverse wireless networks and ever-increasing demand for high-quality wireless services, the need for effective and efficient RRM techniques becomes more important than ever. The addition of efficient packet data channels in both forward and reverse directions and QoS support in 3G standards leads to a more flexible network, but at the same time increases the complexity of determining the optimal allocation of resources especially on the radio interface. This special issue is devoted to addressing the urgent and important need for efficient and effective RRM techniques in the evolving next-generation wireless systems.

We are seeking original, high-quality, and unpublished papers representing the state-of-the-art research in radio resource management aspects of the next-generation wireless communication systems. Topics of interests include, but are not limited to:

- Resource optimization for multimedia services
- Rate allocation and adaptation
- Transmit power control and allocation
- Intelligent scheduling
- Subcarrier allocation in multicarrier systems
- Antenna selection techniques in MIMO systems
- Call admission control
- Load balancing, congestion, and flow control in radio networks
- Modeling and analysis of QoS in wireless networks
- Adaptive QoS control for wireless multimedia
- Delay and jitter management in wireless networks
- Handoff and mobility management
- RRM techniques in hybrid radio networks
- Distributed versus centralized RRM

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Special Issue on

Multuser Cooperative Diversity for Wireless Networks

Call for Papers

Multihop relaying technology is a promising solution for future cellular and ad-hoc wireless communications systems in order to achieve broader coverage and to mitigate wireless channels impairment without the need to use large power at the transmitter. Recently, a new concept that is being actively studied in multihop-augmented networks is multuser cooperative diversity, where several terminals form a kind of coalition to assist each other with the transmission of their messages. In general, cooperative relaying systems have a source node multicasting a message to a number of cooperative relays, which in turn resend a processed version to the intended destination node. The destination node combines the signal received from the relays, possibly also taking into account the source's original signal. Cooperative diversity exploits two fundamental features of the wireless medium: its broadcast nature and its ability to achieve diversity through independent channels. There are three advantages from this:

- (1) *Diversity*. This occurs because different paths are likely to fade independently. The impact of this is expected to be seen in the physical layer, in the design of a receiver that can exploit this diversity.
- (2) *Beamforming gain*. The use of directed beams should improve the capacity on the individual wireless links. The gains may be particularly significant if space-time coding schemes are used.
- (3) *Interference Mitigation*. A protocol that takes advantage of the wireless channel and the antennas and receivers available could achieve a substantial gain in system throughput by optimizing the processing done in the cooperative relays and in the scheduling of re-transmissions by the relays so as to minimize mutual interference and facilitate information transmission by cooperation.

The special issue solicits original research papers dealing with up-to-date efforts in design, performance analysis, implementation and experimental results of cooperative diversity networks.

We seek original, high-quality, and unpublished papers representing the state-of-the-art research in the area of multuser cooperative diversity as applied to the next generation

multihop wireless communication systems. We encourage submission of high-quality papers that report original work in both theoretical and experimental research areas.

Topics of interests include, but are not limited to:

- Information theoretic aspects of cooperative diversity
 - Cooperative diversity from the standpoint of multuser information theory: Shannon capacity
 - Cooperative diversity and its relation to network coding
 - Security aspects
- Physical layer and networking aspects of cooperative diversity
 - Cooperative protocols for wireless relay, ad hoc, and sensor multihop networks
 - Cross-layer protocol design
 - Power allocation in networks with cooperative diversity
 - Reducing transmission energy and extending terminal battery life in cooperative diversity networks
 - Relay networks architectures
- MIMO transmission and cooperative diversity networks
 - Cooperative systems with space-time coding
 - MIMO transmission in multihop networks
 - Cooperative MIMO

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Special Issue on

Signal Processing with High Complexity: Prototyping and Industrial Design

Call for Papers

Some modern applications require an extraordinary large amount of complexity in signal processing algorithms. For example, the 3rd generation of wireless cellular systems is expected to require 1000 times more complexity when compared to its 2nd generation predecessors, and future 3GPP standards will aim for even more number-crunching applications. Video and multimedia applications do not only drive the complexity to new peaks in wired and wireless systems but also in personal and home devices. Also in acoustics, modern hearing aids or algorithms for de-reverberation of rooms, blind source separation, and multichannel echo cancellation are complexity hungry. At the same time, the anticipated products also put on additional constraints like size and power consumption when mobile and thus battery powered. Furthermore, due to new developments in electroacoustic transducer design, it is possible to design very small and effective loudspeakers. Unfortunately, the linearity assumption does not hold any more for this kind of loudspeakers, leading to computationally demanding nonlinear cancellation and equalization algorithms.

Since standard design techniques would either consume too much time or do not result in solutions satisfying all constraints, more efficient development techniques are required to speed up this crucial phase. In general, such developments are rather expensive due to the required extraordinary high complexity. Thus, de-risking of a future product based on rapid prototyping is often an alternative approach. However, since prototyping would delay the development, it often makes only sense when it is well embedded in the product design process. Rapid prototyping has thus evolved by applying new design techniques more suitable to support a quick time to market requirement.

This special issue focuses on new development methods for applications with high complexity in signal processing and on showing the improved design obtained by such methods. Examples of such methods are virtual prototyping, HW/SW partitioning, automatic design flows, float to fix conversions, automatic testing and verification, and power aware designs.

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Special Issue on Field-Programmable Gate Arrays in Embedded Systems

Call for Papers

Field-Programmable Gate Arrays (FPGAs) are increasingly used in embedded systems to achieve high performance in a compact area. FPGAs are particularly well suited to processing data straight from sensors in embedded systems. More importantly, the reconfigurable aspects of FPGAs give the circuits the versatility to change their functionality based on processing requirements for different phases of an application, and for deploying new functionality.

Modern FPGAs integrate many different resources on a single chip. Embedded processors (both hard and soft cores), multipliers, RAM blocks, and DSP units are all available along with reconfigurable logic. Applications can use these heterogeneous resources to integrate several different functions on a single piece of silicon. This makes FPGAs particularly well suited to embedded applications.

This special issue focuses on applications that clearly show the benefit of using FPGAs in embedded applications, as well as on design tools that enable such applications. Specific topics of interest include the use of reconfiguration in embedded applications, hardware/software codesign targeting FPGAs, power-aware FPGA design, design environments for FPGAs, system signalling and protocols used by FPGAs in embedded environments, and system-level design targeting modern FPGA's heterogeneous resources.

Papers on other applicable topics will also be considered. All papers should address FPGA-based systems that are appropriate for embedded applications. Papers on subjects outside of this scope (i.e., not suitable for embedded applications) will not be considered.

Authors should follow the EURASIP JES manuscript format described at <http://www.hindawi.com/journals/es/>. Prospective authors should submit an electronic copy of their complete manuscript through the EURASIP JES manuscript tracking system at <http://www.mstracking.com/es/>, according to the following timetable:

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Special Issue on Synchronous Paradigm in Embedded Systems

Call for Papers

Synchronous languages were introduced in the 1980s for programming reactive systems. Such systems are characterized by their continuous reaction to their environment, at a speed determined by the latter. Reactive systems include embedded control software and hardware. Synchronous languages have recently seen a tremendous interest from leading companies developing automatic control software and hardware for critical applications. Industrial success stories have been achieved by Schneider Electric, Airbus, Dassault Aviation, Snecma, MBDA, Arm, ST Microelectronics, Texas Instruments, Freescale, Intel The key advantage outlined by these companies resides in the rigorous mathematical semantics provided by the synchronous approach that allows system designers to develop critical software and hardware in a faster and safer way.

Indeed, an important feature of synchronous paradigm is that the tools and environments supporting development of synchronous programs are based upon a formal mathematical model defined by the semantics of the languages. The compilation involves the construction of these formal models, and their analysis for static properties, their optimization, the synthesis of executable sequential implementations, and the automated distribution of programs. It can also build a model of the dynamical behaviors, in the form of a transition system, upon which is based the analysis of dynamical properties, for example, through model-checking-based verification, or discrete controller synthesis. Hence, synchronous programming is at the crossroads of many approaches in compilation, formal analysis and verification techniques, and software or hardware implementations generation.

We invite original papers for a special issue of the journal to be published in the first quarter of 2007. Papers may be submitted on all aspects of the synchronous paradigm for embedded systems, including theory and applications. Some sample topics are:

- Synchronous languages design and compiling
- Novel application and implementation of synchronous languages
- Applications of synchronous design methods to embedded systems (hardware or software)

- Formal modeling, formal verification, controller synthesis, and abstract interpretation with synchronous-based tools
- Combining synchrony and asynchrony for embedded system design and, in particular, globally asynchronous and locally synchronous systems
- The role of synchronous models of computations in heterogeneous modeling
- The use of synchronous modeling techniques in model-driven design environment
- Design of distributed control systems using the synchronous paradigm

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