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Performance analysis of the IEEE 802.11 MAC protocol for wireless LANs

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SUMMARY

Wireless local area networks (WLANs) are extremely popular being almost everywhere including business, office and home deployments. The IEEE 802.11 protocol is the dominating standard for WLANs. The essential medium access control (MAC) mechanism of 802.11 is called distributed co-ordination function (DCF). This paper provides a simple and accurate analysis using Markov chain modelling to compute IEEE 802.11 DCF performance, in the absence of hidden stations and transmission errors. This mathematical analysis calculates in addition to the throughput efficiency, the average packet delay, the packet drop probability and the average time to drop a packet for both basic access and RTS/CTS medium access schemes. The derived analysis, which takes into account packet retry limits, is validated by comparison with OPNET simulation results. We demonstrate that a Markov chain model presented in the literature, which also calculates throughput and packet delay by introducing an additional transition state to the Markov chain model, does not appear to model IEEE 802.11 correctly, leading to ambiguous conclusions for its performance. We also carry out an extensive and detailed study on the influence on performance of the initial contention window size (CW), maximum CW size and data rate. Performance results are presented to identify the dependence on the backoff procedure parameters and to give insights on the issues affecting IEEE 802.11 DCF performance.

KEY WORDS: IEEE 802.11; wireless LANs; DCF; packet delay; performance evaluation; retry limits

1. INTRODUCTION

In recent years, wireless local area networks (WLANs) have played a key role in the data communications and networking areas, having witnessed a significant development. Technological and regulatory developments have allowed the issues of high prices, low data rates and licensing requirements to be addressed driving the popularity of wireless LANs to grow significantly. With wireless networking, regardless of where end users are, they can have

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network connectivity, being a mouse-click away from key information and applications. Moreover, recent advances in wireless technology and mobile communications have equipped wireless capability portable devices including palmtop computers, laptops and personal digital assistants (PDAs) [1, 2].

The Institute of Electrical and Electronics Engineers (IEEE) has developed the 802.11 standard family [3–5], in order to deal with the modern wireless connectivity needs. Over the years, the IEEE 802.11 protocol has become a mature technology, achieved worldwide acceptance and turned into the dominating standard for WLANs. The IEEE 802.11a standard [4] operating on the 5 GHz radio frequency band and the IEEE 802.11b standard [5] using the 2.4 GHz frequency band, provide up to 54 Mbit/s and 11 Mbit/s data rates, respectively.

The IEEE 802.11 standard includes detailed specifications for both the medium access control (MAC) and the physical layer (PHY). The MAC incorporates two different medium access methods for WLANs; the compulsory distributed co-ordination function (DCF) and the optional point co-ordination function (PCF). The contention-based DCF is an asynchronous data transmission function, which best suits delay insensitive data (e.g. email, ftp). On the other hand, the polling-based point co-ordination function (PCF) is utilized in delay sensitive data transmissions (e.g. real-time audio or video).

DCF defines two access mechanisms to employ packet transmission. The default scheme is called the basic access mechanism, in which stations transmit data packets after deferring when the medium is busy. The 802.11 standard also provides an optional way of transmitting data packets, namely, the request to send/clear to send (RTS/CTS) reservation scheme. This scheme uses small RTS/CTS packets to reserve the medium before large packets are transmitted in order to reduce the duration of a collision. Moreover, the RTS/CTS reservation scheme is utilized to combat the hidden stations problem [6]. This phenomenon takes place when stations are unable to hear each other and transmit simultaneously, resulting in a packet collision at the receiver. The presence of hidden stations in an IEEE 802.11 network may result in significant performance degradation [7] and could cause unfairness in accessing the medium because a station's location may result in a higher transmission privilege.

Due to the wide acceptance and use of WLANs, extensive research has been carried out to model and study the IEEE 802.11 protocol [8–18]. Several simulation studies of the 802.11 protocol performance are presented in References [8–10]. Recently, considerable research activity has concentrated on the analytical modelling of DCF. Bianchi in Reference [11] and Wu in Reference [12] employ Markov chain models to analyse DCF operation and calculate the saturated throughput of 802.11 protocol. In particular, Bianchi [11] models the idealistic assumption that packet retransmissions are unlimited and a packet is being retransmitted continuously until its successful reception. Wu [12] extends Bianchi's analysis to include the finite packet retry limits as specified in the IEEE 802.11 standard. Nevertheless, neither Reference [11] nor Reference [12] deals with packet delay, packet drop probability or drop time of a transmitted packet using the 802.11 protocol.

In Reference [13], we derived the average packet delay for Bianchi's model. Additionally, in Reference [14] we have identified the network and traffic conditions for Bianchi's model that render the RTS/CTS mechanism beneficial, achieving lower packet delay with respect to the basic access mechanism. Other papers in the literature [15–18] have attempted to derive the average packet delay performance. Cali in Reference [15] makes the assumption that the backoff time is independent of the number of packet retransmissions and sampled from a geometric distribution. Under these assumptions, [15] develops a mathematical model that calculates the

DCF throughput and the packet virtual transmission time, which is defined as the time interval between two consecutive successful transmissions from (perhaps different) contending stations. Vishnevsky in Reference [16] extends Bianchi's model [11] and Cali's model [15] by developing a new mathematical model in order to take into account the Seizing Effect. This effect takes place when a station that has just completed its transmission successfully seizes the channel since it has a better chance of winning in the next competition than other stations. This mathematical model, utilizing the geometrically distributed backoff time used in Reference [15], calculates throughput, packet virtual transmission time and seizing probability in order to study the unfairness emerging from the seizing effect. However, both References [15, 16] develop complex analytical formulas utilizing several assumptions. In addition, comparison with simulation results in Reference [16] shows that Vishnevsky's model is not very accurate. Ziouva in Reference [17] develops a Markov chain model that introduces an additional transition state to the models of References [11, 12]. This additional state represents the case that a station transmits a new packet without entering the backoff procedure if it detects that its previous transmitted packet was successfully received and the channel is idle. Thus, the model in Reference [17], which calculates throughput and packet delay, actually allows stations to transmit consecutive packets without activating the backoff procedure. 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 Reference [17] and subsequent work [18], based on Reference [17], lack any validation using simulation results. In addition, average packet delay calculation in Reference [17] utilizes a very complicated approach since it calculates the average number of packet collisions before a successful reception and the average time a station's backoff timer remains stopped.

In this paper, we extend Wu's analysis [12] in order to thoroughly study the performance of the IEEE 802.11 protocol for both the basic access and RTS/CTS access mechanisms. We present a simple, elegant and intuitive analysis that takes into account packet retry limits and leads to simple equations for additional performance metrics to throughput efficiency such as the average packet delay, the packet drop probability and the average time to drop a packet for the IEEE 802.11 DCF. As in Reference [12], the key assumption of the mathematical model is that the collision probability of a transmitted packet is constant and independent of the retransmissions that this packet has suffered in the past. OPNET simulation results validate the accuracy of our performance analysis. Moreover, a performance comparison of (a) the proposed delay analysis in Reference [17], (b) our delay analysis and (c) simulation results demonstrates the inaccuracy of the analysis of [17] and subsequent work [18], which is based on Reference [17]. Furthermore, utilizing our proposed mathematical analysis, we explore the dependency of protocol performance on the initial contention window (CW) size, the maximum CW size (by varying the CW increasing factor) and the data rate for both access mechanisms. Performance results are presented to highlight the characteristics of each medium access scheme and to examine the behaviour of the exponential backoff algorithm that affects DCF performance.

The paper is outlined as follows. Section 2 presents IEEE 802.11 DCF and describes both the basic access and RTS/CTS access mechanisms. Section 3 presents the analytical model for DCF performance, which is based on a Markov chain by focusing on the backoff procedure. In Section 4, the analytical model is utilized to carry out a performance analysis, calculating throughput, packet delay, packet drop probability and packet drop time. Section 5 validates the proposed analysis by comparing analytical outcome with OPNET simulation results. Section 6 provides a performance evaluation of both DCF access mechanisms and studies the influence of the backoff and system parameters on protocol performance. Finally, Section 7 concludes the paper.

2. OVERVIEW OF IEEE 802.11 DCF

IEEE 802.11 DCF includes carrier-sensing mechanisms in both the physical and MAC layers. On the physical layer, carrier sensing is performed by detecting any channel activity caused by other stations. On the MAC sub-layer, virtual carrier sensing is achieved by using time fields in the packets. These time fields indicate the duration of an ongoing transmission to other stations. All stations that hear the RTS or the CTS packets update their network allocation vector (NAV) according to the value of the duration field in the received packet and do not transmit for the indicated time period. This duration field also incorporates the SIFS and the ACK packet transmission time period following the data packet, ensuring that the station will sense the medium after the current transmission is over.

In IEEE 802.11 WLANs, priority access to the wireless medium is managed by the use of inter-frame space (IFS) time intervals between the packet transmissions. The IFS time intervals are mandatory periods of idle time on the transmission medium before a station may start transmitting a certain type of packet. Three different IFS intervals have been specified to provide various priority levels for access to the wireless medium; the short IFS (SIFS), the point co-ordination function IFS (PIFS) and the distributed co-ordination function IFS (DIFS). The SIFS is the shortest time interval and is used for the transmission of control packets (RTS, CTS) and acknowledgements (ACK), which have the highest priority. The time intervals PIFS and DIFS are utilized to separate the PCF and DCF modes, giving a higher priority to the former.

The techniques used for packet transmission in DCF, the basic access and the RTS/CTS reservation scheme, are described next.

2.1. The basic access method

According to DCF, each station with a new packet ready for transmission monitors the channel activity. If the channel is idle for a time interval equal to DIFS, the station transmits. Otherwise, if the channel is sensed busy (either immediately or during the DIFS), the station persists to monitor the channel until it is determined idle for more than DIFS. The station then initializes its backoff timer and defers transmission for a randomly selected backoff interval in order to minimize collisions. 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. The station whose backoff timer expires first begins transmission and the other stations freeze their timers and defer transmission. Once the current station completes transmission, the backoff process repeats again and the remaining stations reactivate their backoff timers.

A station that receives a data packet, replies by sending a positive acknowledgement (ACK) packet after a SIFS time interval, confirming the successful reception of the data packet. If the source station does not receive an ACK within a specified time, the data packet is assumed to have been lost and a retransmission is scheduled according to the specified backoff rules. Moreover, in order to avoid channel capture, a station must wait a random backoff time between two consecutive packet transmissions. After a successful packet transmission, if the station still has packets buffered for transmission, it must execute a new backoff process [3–5].

2.2. The RTS/CTS access method

In 802.11, DCF also specifies an optional way of transmitting data packets which involves transmission of special short RTS and CTS packets prior to the transmission of the data packet. The RTS/CTS scheme is mainly used to minimize the amount of time wasted when a collision occurs and to combat the hidden station problem. Before initiating the transmission of a data packet, the source station sends a short control packet, called RTS, announcing the duration of the upcoming transmission. When the destination station receives the RTS packet, it replies with a CTS packet after a SIFS interval, echoing the duration of the upcoming transmission. After the successful RTS/CTS exchange, the source station transmits the data packet. The receiver responds with an ACK packet to acknowledge a successful reception of the data packet.

The RTS/CTS scheme addresses the hidden station problem since all the stations are capable of updating their NAVs, based on the receipt of either the RTS or the CTS control packets. Thus, if a station is hidden from either the transmitting or the receiving station, by detecting just one packet between the RTS and CTS packets, it can suitably defer transmission, and hence avoid collision. Since collisions may occur only on the RTS packets and are detected by the lack of CTS response, the RTS/CTS scheme increases the system performance by reducing the duration of a collision, especially when long data packets are transmitted. On the other hand, RTS/CTS decreases efficiency since it transmits two additional packets without any payload. In particular, when short data packets are transmitted, the use of the RTS/CTS reservation scheme might not be advantageous over the basic access scheme. Hence, the standard specifies a manageable object *RTS_Threshold* that indicates the data length under which the data packets should be sent without RTS/CTS. The suitable choice of the *RTS_Threshold* parameter is essential in determining the optimal use of the RTS/CTS mechanism, which can become highly beneficial for the performance of IEEE 802.11 WLANs.

2.3. The binary exponential backoff (BEB) of DCF

IEEE 802.11 DCF is based on a carrier sense multiple access with collision avoidance (CSMA/CA) technique. A contention resolution method, namely binary exponential backoff (BEB), is utilized to randomize moments at which stations are trying to access the wireless medium. By means of this random backoff mechanism, the probability of collisions due to multiple simultaneous transmissions is minimized. In fact, the time following an idle DIFS is slotted and a station is allowed to transmit only at the beginning of each slot. The value of the backoff timer for each station is a uniformly distributed integer number of slots in the interval $[0, W_i - 1]$, where W_i is the current contention window (CW) size and i is the backoff stage. The value of W_i depends on the number of failed transmissions of a packet. The backoff timer is decremented when the medium is sensed idle. A station initiates a packet transmission when its backoff timer reaches zero.

At the first transmission attempt of a packet, W_i is set equal to $W_0 = \mathrm{CW}_{\min}$, which is called the minimum contention window size. If two or more stations start transmission simultaneously in the same slot, a collision takes place. After a packet collision, the contention window is doubled up to a maximum value, $W_{m'} = \mathrm{CW}_{\max} = 2^{m'} \cdot W$, where m' is the CW increasing factor. Once W_i reaches $\mathrm{CW}_{\max}$, it will remain at the value of $\mathrm{CW}_{\max}$ until it is reset to $\mathrm{CW}_{\min}$.

Therefore, the current contention window (CW) size is given by

$$\begin{aligned} W_i &= 2^i \cdot W, \quad i \leqslant m' \\ W_i &= 2^{m'} \cdot W, \quad i > m' \end{aligned} \tag{1}$$

where $i \in [0, m]$ and m represents the station's short retry count.¶ Here m is also the maximum backoff stage. The contention window is reset to CW_{min} in the following cases: (a) after the successful transmission of a data packet; (b) when SSRC reaches the short retry limit. The SSRC is reset to 0 whenever a packet is discarded or a CTS is received in response to an RTS or an ACK is received in response to a data packet when RTS/CTS is not used.

3. ANALYTICAL MODEL

The mathematical analysis makes use of the same assumptions as in References [11–14], in order to analyse and study the performance of the IEEE 802.11 protocol. We assume that the network consists of a finite number of n contending stations using the same channel access mechanism in ideal channel conditions (no channel bit errors or hidden stations). We also consider saturation conditions; every station always has a packet ready for transmission (its transmission queue is always non-empty), immediately after every successful packet transmission. The key assumption of our analysis is that the collision probability p of a data packet transmission is constant and independent of the number of collisions the packet has suffered in the past. Moreover, we use the same discrete-time Markov chain model for depicting the backoff procedure followed by each station as in References [12, 13].

Let $b(t)$ be the stochastic process that represents the backoff timer for a specific station and $s(t)$ be the stochastic process representing the backoff stage $[0, \ldots, m]$ for a given station at time t, where m is the packet retry limit. A discrete integer time scale is adopted; t and $t+1$ correspond to the beginning of two consecutive slot times and the backoff timer of each station decrements at the beginning of each slot time. The process $b(t)$ corresponds to the number of the remaining slot times before a packet transmission and does not represent the remaining time before a transmission attempt. Since a successful packet transmission may take place between two consecutive slot times, the adopted discrete time scale does not directly relate to system time. More specifically, as explained earlier, the backoff timer is frozen when the medium is sensed busy and is reactivated again when the medium is sensed idle. For this reason, the time interval between two consecutive slot times for a station may be much longer than the slot time size σ, due to the fact that it could include a packet transmission by another station. Note that with the term slot time we will refer to either the (constant) value σ or the (variable) time interval between two consecutive backoff timer decrements. The backoff timer of each station depends on the number of collisions and successful transmissions experienced in the past. As a result, the stochastic process $b(t)$ is non-Markovian.

As in References [11–14], the key approximation of our analysis is that each packet collides with the same constant probability p regardless of the number of retransmissions the packet has

¶Each station maintains a station short retry count (SSRC), which takes an initial value of zero for every new packet. The short retry count indicates the maximum number of retransmission attempts of an RTS packet or of a data packet when RTS/CTS is not used. When this retry limit is reached, retry attempts shall cease and this packet is discarded.

suffered in the past. This assumption is more accurate as long as W and n get larger. Based on this assumption, we utilize the discrete-time Markov chain depicted in Figure 1 to model the bi-dimensional process $\{b(t), s(t)\}$.

We adopt the same short notation $P\{i_1, k_1 \mid i_0, k_0\} = P\{s(t+1) = i_1, b(t+1) = k_1 \mid s(t) = i_0, b(t) = k_0\}$ used in Reference [11]. The state transition diagram for this Markov chain model has the following non-null one-step transition probabilities:

$$\begin{aligned} &P\{i, k \mid i, k+1\} = 1, \quad k \in [0, W_i - 2], \quad i \in [0, m] \\ &P\{0, k \mid i, 0\} = (1-p)/W_0, \quad k \in [0, W_0 - 1], \quad i \in [0, m-1] \\ &P\{i, k \mid i-1, 0\} = p/W_i, \quad k \in [0, W_i - 1], \quad i \in [1, m] \\ &P\{0, k \mid m, 0\} = 1/W_0, \quad k \in [0, W_0 - 1] \end{aligned} \tag{2}$$

The first equation in (2) accounts for the case that the slot time i is idle and the backoff timer is decremented. The second equation represents the fact that after a successful packet transmission at backoff stage i, a new packet starts from backoff stage 0. The new value of the backoff timer is uniformly chosen in the interval $[0, W_0 - 1]$. The third equation shows that when an unsuccessful transmission occurs at backoff stage $i-1$, the backoff increases and the new value of the backoff timer is uniformly chosen in the range $[0, W_i - 1]$. The last equation represents the fact that at the maximum backoff stage m, the contention window (CW) is reset to

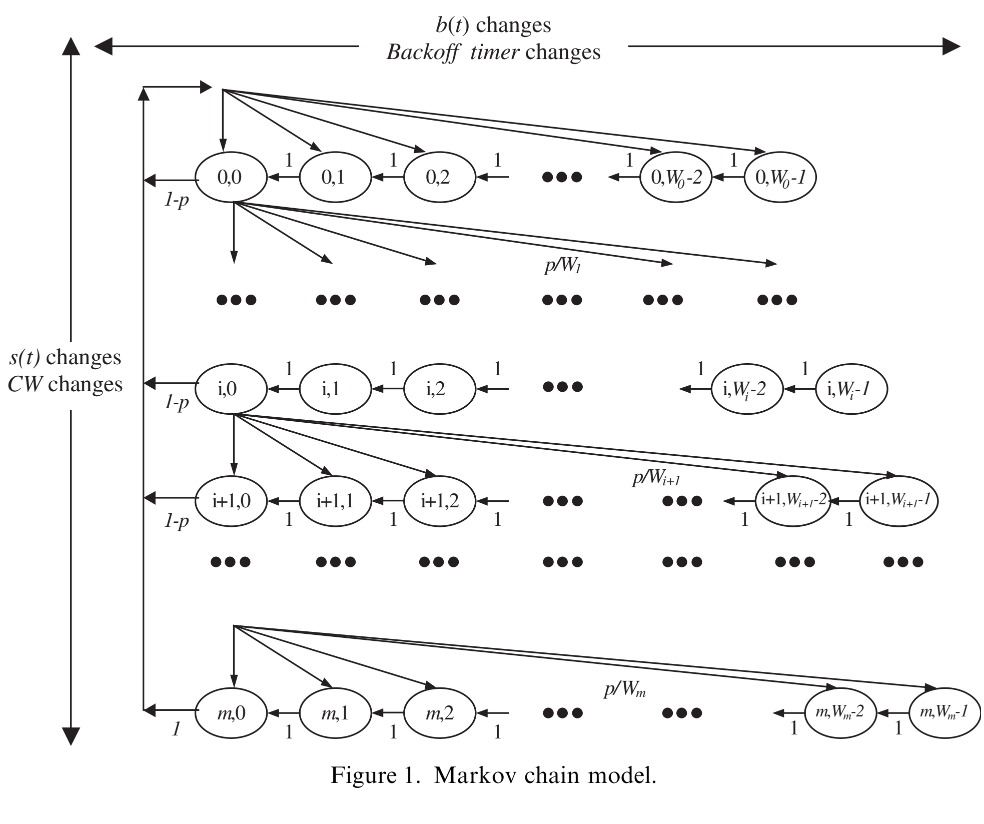

Figure 1. Markov chain model.

$\mathrm{CW_{min}} = W_0$ either due to a successful packet transmission or because the retry limit is reached. In the latter case, the packet is discarded and the backoff mechanism is invoked for a new packet from backoff stage 0.

In order to obtain a closed-form solution for the considered Markov chain, let $b_{i,k} = \lim_{t\to\infty} P\{s(t) = i, b(t) = k\}$ be the stationary distribution of this Markov chain, where $i \in [0, m]$, $k \in [0, W_i - 1]$. 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 relations for $b_{i,0}$:

$$b_{i,0} = p \cdot b_{i-1,0}, \quad 0 < i \leqslant m \tag{3}$$

$$b_{i,0} = p^i \cdot b_{0,0}, \quad 0 < i \leqslant m \tag{4}$$

Owing to chain regularities, the values of $b_{i,k}$ are given by

$$b_{i,k} = \frac{W_i - k}{W_i} \cdot \begin{cases} (1-p) \cdot \sum_{j=0}^{m-1} b_{j,0} + b_{m,0}, & i = 0 \\ p \cdot b_{i-1,0}, & 0 < i \leqslant m \end{cases} \tag{5}$$

By means of Equations (1), (3), (4) and imposing that $\sum_{j=0}^{m-1} b_{j,0} = b_{0,0} \cdot (1-p^m)/(1-p)$, Equation (5) becomes

$$b_{i,k} = \frac{W_i - k}{W_i} \cdot b_{i,0}, \quad 0 \leqslant i \leqslant m, \ 0 \leqslant k \leqslant W_i - 1 \tag{6}$$

Equations (4) and (6) express all $b_{i,k}$ values as a function of $b_{0,0}$ and p. Applying the normalization condition for this stationary distribution:

$$\begin{aligned} 1 &= \sum_{i=0}^{m} \sum_{k=0}^{W_i-1} b_{i,k} = \sum_{i=0}^{m} b_{i,0} \cdot \sum_{k=0}^{W_i-1} \frac{W_i - k}{W_i} \\ &= \sum_{i=0}^{m} b_{i,0} \cdot \frac{W_i + 1}{2} = \sum_{i=0}^{m} p^i \cdot b_{0,0} \cdot \frac{W_i + 1}{2} \\ &= \frac{b_{0,0}}{2} \cdot \left(\sum_{i=0}^{m} p^i \cdot W_i + \sum_{i=0}^{m} p^i \right) \end{aligned} \tag{7}$$

We have to distinguish two different cases according to the values of m and m'.

- When $m > m'$ and by taking into account Equation (1), Equation (7) becomes

$$\begin{aligned} 1 &= \frac{b_{0,0}}{2} \cdot \left[\sum_{i=0}^{m'} ((2p)^i \cdot W) + \sum_{i=m'+1}^{m} (p^i \cdot 2^{m'} \cdot W) + \sum_{i=0}^{m} p^i \right] \\ &= \frac{b_{0,0}}{2} \cdot \left[\frac{1-(2p)^{m'+1}}{1-2p} \cdot W + 2^{m'} \cdot W \cdot p^{m'+1} \cdot \frac{1-p^{m-m'}}{1-p} + \frac{1-p^{m+1}}{1-p} \right] \end{aligned}$$

from which

$$b_{0,0} = \frac{2 \cdot (1-2p) \cdot (1-p)}{W \cdot (1-(2p)^{m'+1}) \cdot (1-p) + (1-2p) \cdot (1-p^{m+1}) + W \cdot 2^{m'} \cdot p^{m'+1} \cdot (1-2p) \cdot (1-p^{m-m'})}$$

- When $m \leqslant m'$ and by considering Equation (1), Equation (7) turns into

$$1 = \frac{b_{0,0}}{2} \cdot \left[\sum_{i=0}^{m} ((2p)^i \cdot W) + \sum_{i=0}^{m} p^i \right]$$
$$= \frac{b_{0,0}}{2} \cdot \left[\frac{1-(2p)^{m+1}}{1-(2p)} \cdot W + \frac{1-p^{m+1}}{1-p} \right]$$

from which

$$b_{0,0} = \frac{2 \cdot (1-2p) \cdot (1-p)}{W \cdot (1-(2p)^{m+1}) \cdot (1-p) + (1-2p) \cdot (1-p^{m+1})}$$

Finally, $b_{0,0}$ is given by Equation (8) and depends on the values of m and m'.

$$b_{0,0} = \begin{cases} \dfrac{2 \cdot (1-2p) \cdot (1-p)}{W \cdot (1-(2p)^{m+1}) \cdot (1-p) + (1-2p) \cdot (1-p^{m+1})}, & m \leqslant m' \\ \dfrac{2 \cdot (1-2p) \cdot (1-p)}{W \cdot (1-(2p)^{m'+1}) \cdot (1-p) + (1-2p) \cdot (1-p^{m+1}) + W \cdot 2^{m'} \cdot p^{m'+1} \cdot (1-2p) \cdot (1-p^{m-m'})}, \\ m > m' \end{cases} \tag{8}$$

Using the previous analysis, we can derive the probability τ that a station transmits a packet in a randomly chosen slot time. Note that a packet transmission occurs when the backoff timer of the transmitting station is equal to zero, regardless of the backoff stage. By utilizing the previous Markov chain model, the probability τ that a station transmits a packet in a randomly chosen slot time is equal to

$$\tau = \sum_{i=0}^{m} b_{i,0} = \sum_{i=0}^{m} p^i \cdot b_{0,0} = b_{0,0} \cdot \frac{1-p^{m+1}}{(1-p)} \tag{9}$$

and $b_{0,0}$ can be acquired from Equation (8). From Equation (9) we observe that the transmission probability τ depends on the collision probability p, which is still unknown, and it will be derived next. The probability p that a transmitted packet encounters a collision is the probability that at least one of the $n-1$ remaining stations transmit in the same time slot. If we assume that all stations see the system at steady state and transmit with probability τ, the collision probability p is given by

$$p = 1 - (1-\tau)^{n-1} \tag{10}$$

Equations (9) and (10) represent a non-linear system with two unknowns τ and p, which can be solved utilizing numerical methods. Note that $p \in [0, 1]$ and $\tau \in [0, 1]$. As it has been shown in Reference [11] throughout a detailed proof, this non-linear system has a unique solution.

4. MATHEMATICAL ANALYSIS

Our performance analysis considers the following metrics, which are good indicators of the IEEE 802.11 protocol performance; throughput, average packet delay, probability of a packet being discarded when it reaches the maximum retransmission limit and the average time to drop a packet. The derived performance analysis is applicable to both the basic access and RTS/CTS medium access mechanisms.

4.1. Saturation throughput

This paper utilizes the concept of 'saturation throughput' for a finite number of n contending stations. We assume that a station transmits a data packet of fixed payload size of l bits at a data rate of C Mbit/s. The saturation throughput is defined as the limit reached by the system throughput as the offered load increases and represents the maximum load that the system can carry in stable conditions [11]. In particular, as the offered load increases, the throughput grows up to a maximum value, referred to as maximum throughput. However, a further increase of the offered load leads to a decrease in the system throughput. More details about the mathematical formulation and interpretation of the unstable behaviour of several random access schemes could be found in References [11, 19].

In order to compute the system throughput S, we analyse what happens in a randomly chosen slot time. Let P_{tr} be the probability that at least one station transmits a packet in the considered slot time. Since each station transmits with probability τ, the probability P_{tr} is given by

$$P_{tr} = 1 - (1 - \tau)^n \tag{11}$$

A packet collision takes place when two or more contending stations initiate simultaneously a packet transmission in the same slot time. The probability P_s that an occurring packet transmission is successful is given by the probability that exactly one station transmits and the remaining $n - 1$ stations defer transmission, conditioned on the fact that at least one station (out of n stations) transmits.

$$P_s = \frac{n \cdot \tau \cdot (1 - \tau)^{n-1}}{P_{tr}} = \frac{n \cdot \tau \cdot (1 - \tau)^{n-1}}{1 - (1 - \tau)^n} \tag{12}$$

Considering that a random slot is empty with probability $(1 - P_{tr})$, contains a successful transmission with probability $P_{tr} \cdot P_s$ and a collision with probability $P_{tr} \cdot (1 - P_s)$, the average length of a slot time $E[\text{slot}]$ is equal to

$$E[\text{slot}] = (1 - P_{tr}) \cdot \sigma + P_{tr} \cdot P_s \cdot T_s + P_{tr} \cdot (1 - P_s) \cdot T_c \tag{13}$$

where σ is the duration of an empty slot time, and T_s and T_c are the time durations when the medium is sensed busy due to a successful transmission and a collision, respectively.

If we follow the same reasoning with Reference [11], the system throughput S can be expressed by dividing the successfully transmitted payload information in a slot time, with the average length of a slot time as follows:

$$S = \frac{P_{tr} \cdot P_s \cdot l}{(1 - P_{tr}) \cdot \sigma + P_{tr} \cdot P_s \cdot T_s + P_{tr} \cdot (1 - P_s) \cdot T_c} \tag{14}$$

The values of T_s and T_c depend on the medium access mechanism and are defined for the basic access and the RTS/CTS access mechanisms as follows:

$$\left\langle \begin{aligned} T_s^{bas} &= \mathrm{DIFS} + T_{header} + \frac{l}{C} + \mathrm{SIFS} + T_{ACK} \\ T_c^{bas} &= \mathrm{DIFS} + T_{header} + \frac{l}{C} + \mathrm{SIFS} + T_{ACK} \end{aligned} \right. \tag{15}$$

$$\left\langle \begin{aligned} T_s^{RTS} &= \mathrm{DIFS} + T_{RTS} + \mathrm{SIFS} + T_{CTS} + \mathrm{SIFS} + T_{header} + \frac{l}{C} + \mathrm{SIFS} + T_{ACK} \\ T_c^{RTS} &= \mathrm{DIFS} + T_{RTS} + \mathrm{SIFS} + T_{CTS} \end{aligned} \right. \tag{16}$$

where C is the data rate, T_{header}, T_{ACK}, T_{RTS} and T_{CTS} are the times required to transmit the packet payload header, the ACK, RTS and CTS control packets, respectively. The above time intervals are given by

$$T_{header} = \frac{\mathrm{MAC}_{hdr}}{C} + \frac{\mathrm{PHY}_{hdr}}{C_{control}}, \quad T_{ACK} = \frac{l_{ACK}}{C_{control}} \tag{17}$$

$$T_{RTS} = \frac{l_{RTS}}{C_{control}}, \quad T_{CTS} = \frac{l_{CTS}}{C_{control}} \tag{18}$$

where MAC_{hdr} and PHY_{hdr} is the MAC and the physical header (in bits), $C_{control}$ is the rate at which the control packets are transmitted l_{ACK}, l_{RTS}, and l_{CTS} the lengths of ACK, RTS and CTS control packets, respectively. Note that the data C and the control $C_{control}$ rates may not be the same.

4.2. Packet drop probability

The drop probability is defined as the probability that a packet is dropped when the retry limit is reached. A packet reaches the last backoff stage m, if it encounters m collisions in the previous stages and this packet will be dropped if it experiences another collision. Consequently, the packet drop probability is independent of the employed access mechanism (basic access or RTS/CTS) and is given by

$$p_{drop} = p^{m+1} \tag{19}$$

4.3. Average time to drop a packet

A packet is dropped when it reaches the last backoff stage and experiences another collision. The average value of slots the station will utilize in the i stage (including the transmission slot) is given by

$$d_i = \frac{W_i + 1}{2}, \quad i \in [0, m] \tag{20}$$

Moreover, the average number of slots $E[T_{drop}]$ required for a packet to experience $m+1$ collisions in the $(0, 1, \ldots, m)$ stages is given by

$$E[T_{drop}] = \sum_{i=0}^{m} d_i = \begin{cases} \dfrac{W \cdot (2^{m+1} - 1) + (m+1)}{2}, & m \leqslant m' \\ \dfrac{W \cdot (2^{m'+1} - 1) + W \cdot 2^{m'} \cdot (m - m') + (m+1)}{2}, & m > m' \end{cases} \tag{21}$$

Finally, the average time to drop a packet $E[D_{\text{drop}}]$ because its retry limit is reached can be found by

$$E[D_{\text{drop}}] = E[T_{\text{drop}}] \cdot E[\text{slot}] \tag{22}$$

4.4. Average packet delay

Our analysis considers the delay D for a successfully transmitted packet, which is defined to be the time interval from the time a 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

$$E[D] = E[X] \cdot E[\text{slot}] \tag{23}$$

where $E[X]$ is the average number of slot times required for a successful packet transmission. $E[X]$ can be found by multiplying the number of slots d_i the packet is delayed in each backoff stage by the probability q_i that a packet that is not dropped reaches the i backoff stage:

$$E[X] = \sum_{i=0}^{m} d_i \cdot q_i \tag{24}$$

The probability q_i that a packet reaches the i backoff stage, provided that this packet is not to be discarded, is equal to

$$q_i = \frac{(p^i - p^{m+1})}{1 - p^{m+1}}, \quad i \in [0, m] \tag{25}$$

where p^i is the probability that a packet reaches the i stage, and p^{m+1} and $(1 - p^{m+1})$ are the probabilities that a packet is dropped or not, respectively. Combining Equations (20), (25) and (24), $E[X]$ is given by

$$E[X] = \sum_{i=0}^{m} \left[\frac{(p^i - p^{m+1}) \cdot (W_i + 1)/2}{1 - p^{m+1}} \right] \tag{26}$$

After some algebra, Equation (26) becomes

$$E[X] = \begin{cases} \dfrac{W \cdot (1-(2p)^{m+1}) \cdot (1-p) + (1-2p) \cdot (1-p^{m+1})}{2 \cdot (1-2p) \cdot (1-p) \cdot (1-p^{m+1})} - \dfrac{p^{m+1}}{1-p^{m+1}} \cdot E[T_{\text{drop}}], & m \leqslant m' \\ \dfrac{W \cdot (1-(2p)^{m'+1}) \cdot (1-p) + W \cdot 2^{m'} \cdot p^{m'+1} \cdot (1-p^{m-m'}) \cdot (1-2p) + (1-2p) \cdot (1-p^{m+1})}{2 \cdot (1-2p) \cdot (1-p) \cdot (1-p^{m+1})} & \\ \quad - \dfrac{p^{m+1}}{1-p^{m+1}} \cdot E[T_{\text{drop}}], & m > m' \end{cases} \tag{27}$$

Finally, if we substitute Equations (27), (21) and (13) into Equation (23), the average packet delay $E[D]$ can be easily calculated.

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 modelling and simulation software package from OPNET Technologies (formerly MIL3 Inc). OPNET Modeler is an event-driven simulator and provides a powerful graphical tool to display simulation statistics. OPNET uses hierarchically linked domains to denote a network design and stations are defined in the network domain, which is the top-level domain. Each station has a set of processes and each process can represent a layer in the protocol stack. A process can be defined by a finite state machine. The transmission of packets across network links is controlled by pipeline-stage C/C + + coded routines. The user can produce and add C code to be executed when entering and exiting each state. 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, etc. The simulator closely follows all timer values and packet element transmission times defined by IEEE 802.11 specifications. Furthermore, we have suitably modified the standard library of the OPNET 802.11 simulator in order to employ saturation conditions, i.e. all stations always have a packet ready for transmission.

The Markov chain analysis, presented in the previous section, 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 Reference [12] and are summarized in Table I. The system parameter values are those specified for the Direct Spread Sequence Spectrum (DSSS) physical layer utilized in IEEE 802.11b [5].

Figures 2–4 confirm the accuracy of the considered assumptions in our mathematical analysis. The figures provide performance results (throughput efficiency, packet delay, packet drop time and packet drop probability) versus the number of contending stations for the basic access and RTS/CTS mechanisms. The channel data rate is equal to 1 Mbit/s. Figures 2 and 3 illustrate that the analytical model that considers retry limits predicts very accurately DCF throughput

Table I. DSSS system parameters in 802.11b.

Parameter	Value
Packet payload, l	8184 bits
Slot time, σ	20 μs
MAC header	224 bits
PHY header	192 bits
RTS packet	160 bits + PHY header
CTS packet	112 bits + PHY header
ACK packet	112 bits + PHY header
DIFS	50 μs
SIFS	10 μs
Channel data rate	1, 5.5 and 11 Mbit/s
Control rate	1 Mbit/s
Minimum CW, W_0	32
Number of CW sizes, m'	5
Short retry limit	6

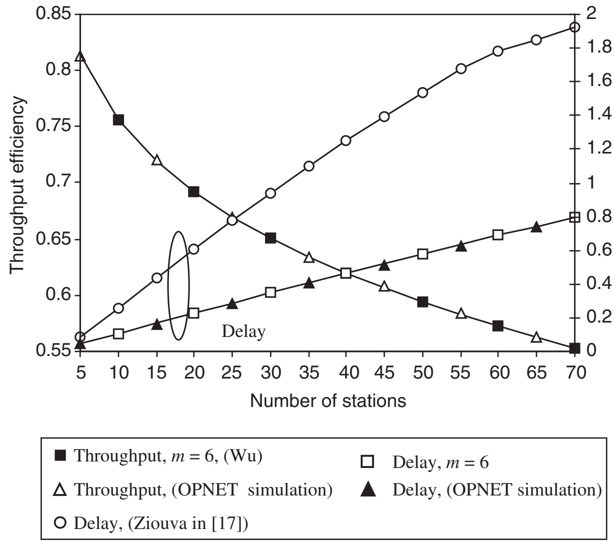

Figure 2. Throughput efficiency and packet delay for basic access: analysis versus simulation.

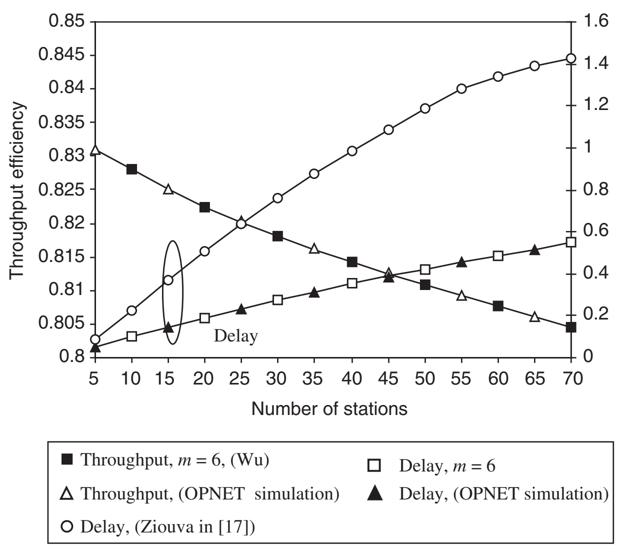

Figure 3. Throughput efficiency and packet delay for RTS/CTS: analysis versus simulation.

performance, a conclusion not clearly drawn in Reference [12] which added packet retry limits in the analytical model in Reference [11]. Note that simulation results are acquired with a 95% confidence interval lower than 0.002. Figures 2 and 3 also display packet delay calculated using our delay analysis as well as Ziouva's [17] model against OPNET simulation results. The performance comparison shows that our packet delay analysis gives results in high agreement

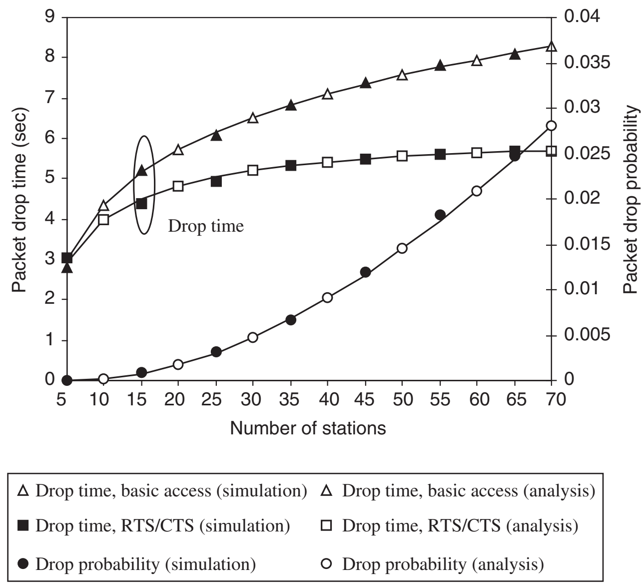

Figure 4. Packet drop time and packet drop probability: analysis versus simulation.

with OPNET simulations and reveals that the model in Reference [17], which is less conformant to the IEEE 802.11 standard than our model, causes a high packet delay overestimation due to the adoption of the incorrect additional transition state as well as due to the absence of packet retry limits. Thus, the results derived from the model in Reference [17] and subsequent work [18], which is based on Reference [17], lead to ambiguous conclusions for the performance of IEEE 802.11 protocol.

Figure 4 validates our analysis for the other two considered performance metrics; packet drop time and packet drop probability. Moreover, Figures 2, 3 and 4 show that the RTS/CTS reservation scheme achieves higher throughput, lower packet delay as well as lower packet drop time as compared with the basic access mechanism, for the specific large packet size and the data rate, as a result of shorter collision duration. Furthermore, an interesting observation is that packet drop probability is the same for both basic access and RTS/CTS since it is independent of the employed medium access scheme.

6. PERFORMANCE EVALUATION

Figures 5–8 investigate the dependency of packet drop probability, packet drop time, packet delay and throughput efficiency on the initial contention window size W. The figures study both the basic access and the RTS/CTS mechanisms and report three different network sizes ($n = 5$, 25 and 50). Figure 5 shows that the adjustment of the initial contention window size to higher values in large network scenarios highly benefits packet drop probability; fewer packets are discarded since higher values of W reduce the number of collisions. On the other hand, for a small number of stations ($n = 5$), the packet drop probability is not considerably affected as a

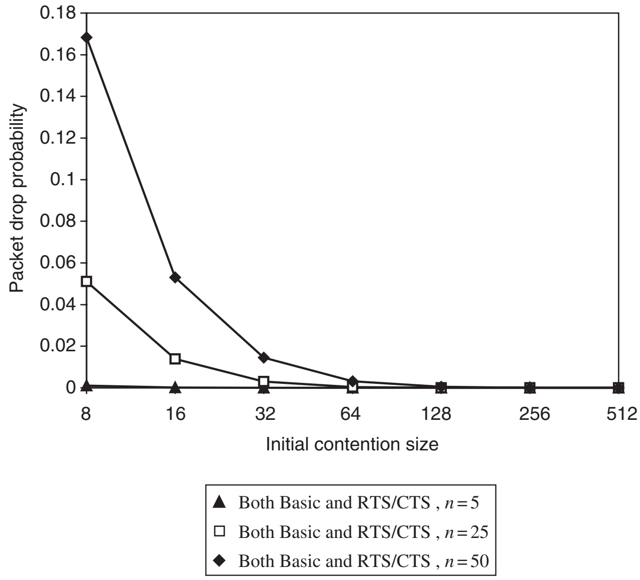

Figure 5. Packet drop probability for basic access and RTS/CTS.

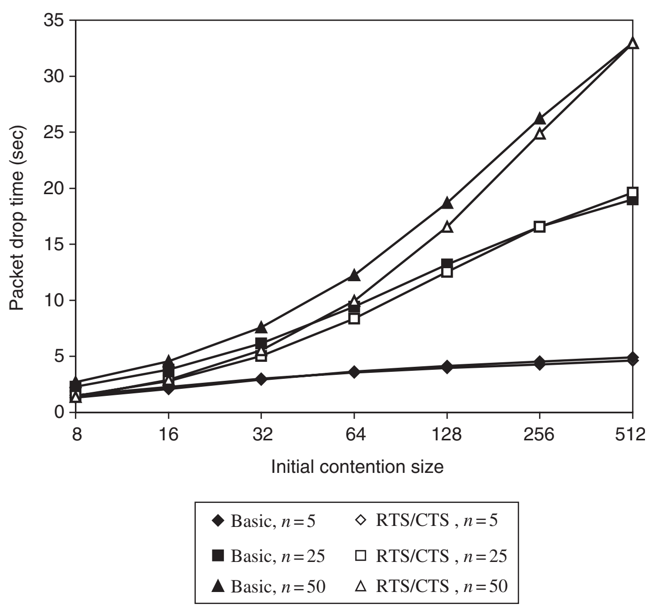

Figure 6. Packet drop time for basic access and RTS/CTS.

result of the low collision probability. Moreover, Figure 6 illustrates that higher values of W cause an increase on packet drop time for both the basic access and RTS/CTS mechanisms mainly due to the increase of idle slots.

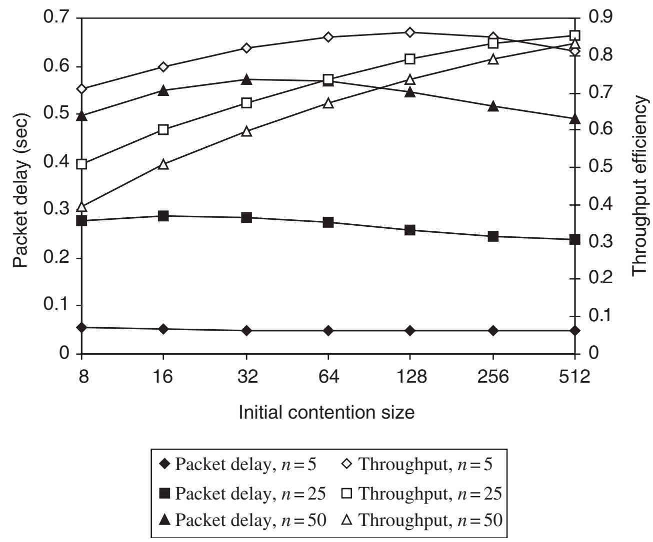

Figure 7. Packet delay and throughput efficiency for basic access.

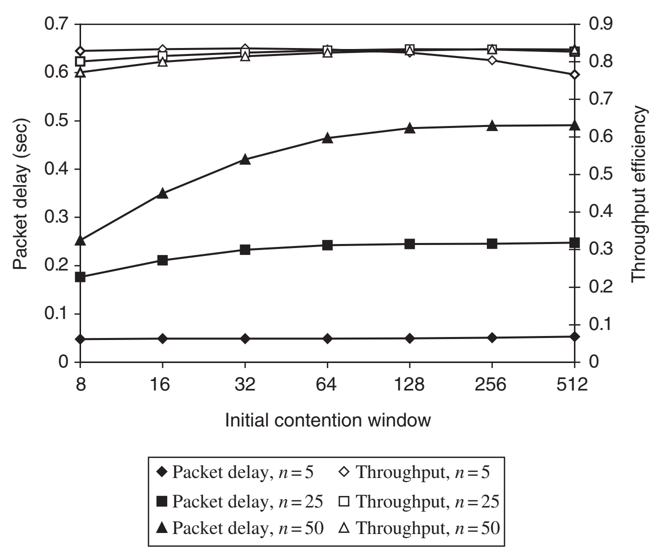

Figure 8. Packet delay and throughput efficiency for RTS/CTS.

Figure 7 plots packet delay and throughput efficiency versus initial contention window size for the basic access scheme and for various network sizes. Figure 7 shows that when the basic access mechanism is employed, throughput improves as initial contention window increases. The situation is explained since when W increases, the number of collisions decreases and the system throughput gets higher. The only exception is when $n = 5$ and $W \geqslant 128$, throughput drops off due to the increased number of idle slots. Furthermore, packet delay is not greatly affected by the increase of the initial contention window, in small network sizes. In large network scenarios, when initial contention size grows, more packets are transmitted successfully (Figure 5). A notable result is that packet delay increases with the increase of initial contention size, especially when $W \leqslant 32$, as a result of the fact that the additional packets contain large delays. In the case of $W \geqslant 64$, packet delay drops off as a result of fewer collisions that take place. The figure also indicates that a very small initial contention window is not effective for large networks due to the increased number of collisions. In contrast, a large value of W is unsuitable for a small network size ($n \leqslant 5$) due to many idle slots.

Figure 8 explores the effect of W when the RTS/CTS is employed and indicates that the choice of initial contention window does not significantly affect throughput due to the shorter collision duration of the RTS packets. When $n = 5$ and $W \geqslant 128$, throughput slightly decreases due to the increase of idle slots. For a large network size, the throughput improves for high values of W but remains constant as long as W is greater than 64 due to the fact that these high W values can effectively cope with the increased number of collisions. In contrast, for large network sizes, packet delay increases when W increases due to the fact that more packets are being successfully transmitted as illustrated in Figure 5 through packet drop probability. However, the choice of the initial contention window size does not affect packet delay when $W \geqslant 128$ with a large network size. For a small number of contending stations ($n = 5$), packet delay is not affected by changing the values of initial contention window size since the number of packets that are transmitted successfully is about the same regardless the value of W (Figure 5).

Figures 9–12 study the effect of the maximum CW size (by varying the CW increasing factor m') on packet drop probability, packet drop time, packet delay and throughput efficiency for the basic access and the RTS/CTS mechanisms and for three different network sizes. Figure 9 plots packet drop probability against the CW increasing factor for various network sizes. The figure illustrates that the increase of m' is beneficial for packet drop probability; fewer packets are dropped since higher values of m' deal with the increased number of packet collisions.

Figure 10 reveals that packet drop time is also significantly affected from the change on the CW increasing factor values. Moreover, it is obvious that the medium access mechanism also has a significant effect on packet drop time and the RTS/CTS scheme achieves a considerably lower packet drop time compared to the basic access scheme.

In Figure 11, packet delay is plotted against the CW increasing factor for both basic access and RTS/CTS mechanisms. The figure depicts that packet delay mainly depends on the number of contending stations and increases when n increases. Moreover, the use of the RTS/CTS scheme appears to be beneficial for packet delay especially in large networks, while the basic access experiences higher packet delay values. In all cases, packet delay is not significantly affected when $m' \geqslant 5$ for any network size and access scheme. The reason is that when $m' \geqslant 5$, less collisions are taking place and many packets are transmitted successfully as shown in Figure 9.

Figure 12 studies the effect of the CW increasing factor on throughput efficiency for three different networks sizes, for both basic access and RTS/CTS mechanisms. The throughput performance of the basic access scheme increases as the CW increasing factor increases, whereas the RTS/CTS mechanism appears more robust and constantly achieves high throughput values. Moreover, the CW increasing factor does not affect throughput efficiency when $m' \geqslant 5$ and $m' \geqslant 4$ for the basic access and RTS/CTS schemes, respectively.

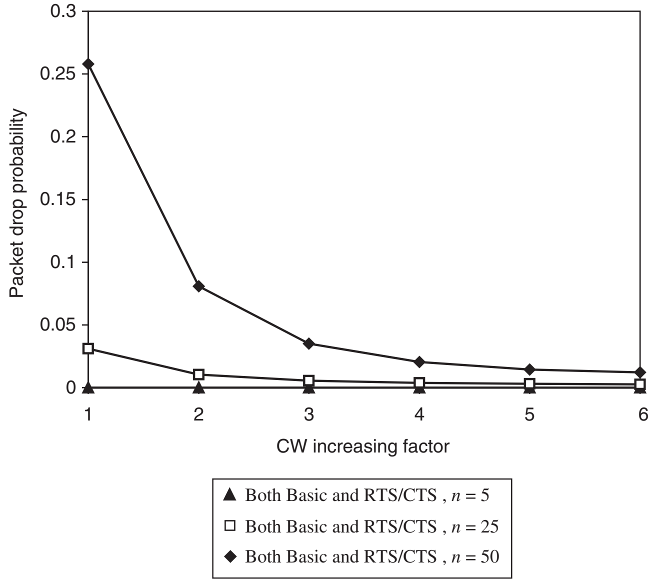

Figure 9. Packet drop probability for basic access and RTS/CTS.

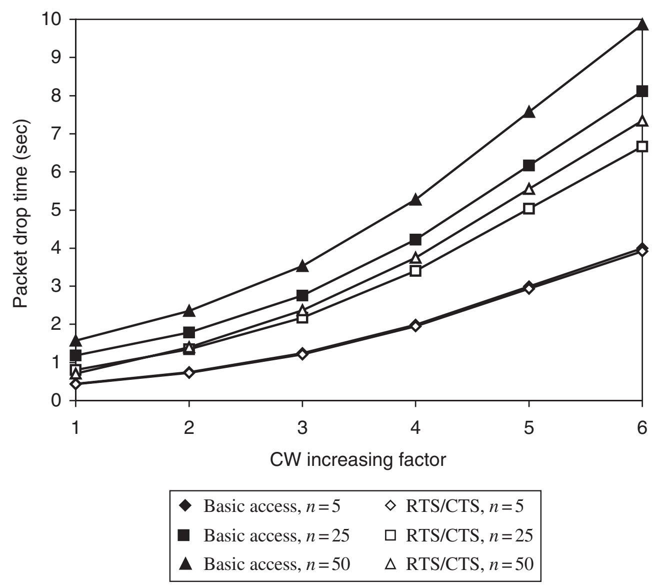

Figure 10. Packet drop time for basic access and RTS/CTS.

Since the IEEE 802.11b standard specifies various data rates, it is interesting to study how performance is influenced by the medium data rate. More specifically, the effect of data rate on packet delay and packet drop probability is illustrated in Figure 13, where packet delay is

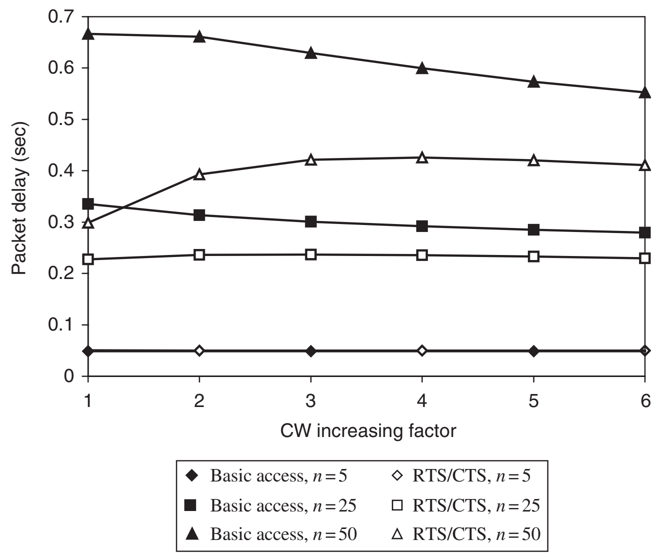

Figure 11. Packet delay for basic access and RTS/CTS.

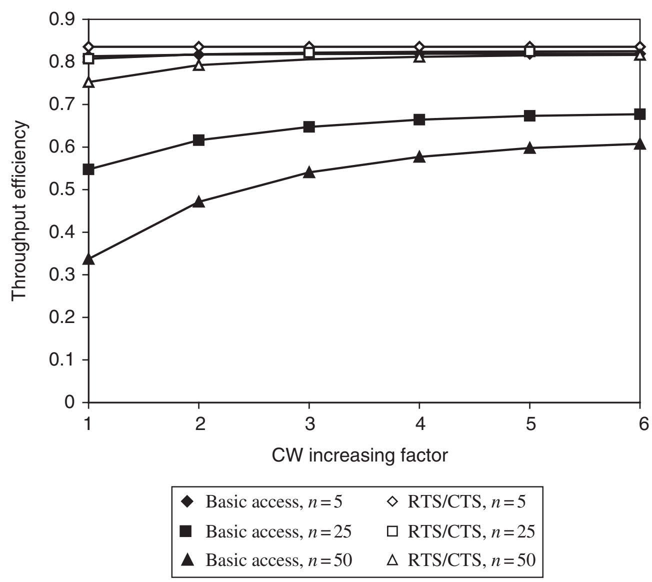

Figure 12. Throughput efficiency for basic access & RTS/CTS.

plotted against the number of stations for three different data rates ($C = 1$, 5.5 and 11 Mbit/s). The figure depicts that the packet delay significantly drops off when the data rate increases since packet transmission time is reduced. Another observation is that the use of the RTS/CTS

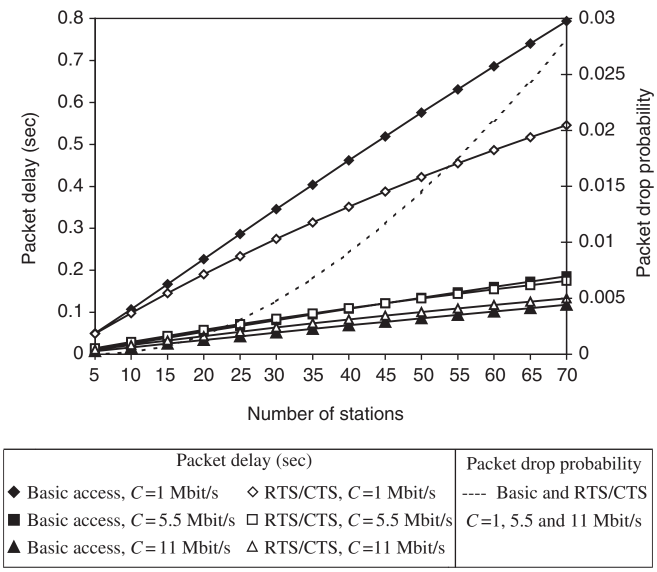

Figure 13. Packet delay and drop probability for various data rates.

scheme results in a considerable decrease on packet delay compared to the basic access scheme when $C = 1$ Mbit/s due to the shorter collision duration. However, when higher data rates are utilized, the surprising result is that the RTS/CTS reservation scheme either is beneficial when the number of stations is greater than 50 ($C = 5.5$ Mbit/s) or even degrades performance ($C = 11$ Mbit/s). This is justified since RTS and CTS control packets are always transmitted at the control rate (1 Mbit/s), causing a considerable communication delay, especially when data rate is high. On the other hand, packet drop probability is independent of data rate. As can be observed in Equation (19), packet drop probability only depends on collision probability p and retry limit m and not on data rate or medium access mechanism.

Figure 14 illustrates that the use of high data rates significantly reduces the average time to drop a packet that reaches its retransmission limit when the basic access scheme is employed. On the contrary, throughput efficiency decreases as the number of the stations increases because more collisions take place. Moreover, throughput efficiency decreases when data rate increases. The situation is explained considering that the time spent on packet transmission is reduced but the duration of DIFS, SIFS and the slot time is independent of medium data rate and remains the same. Thus, the time spent on DIFS, SIFS and backoff delay increases in relation to packet transmission time, resulting in throughput efficiency degradation.

Figure 15 plots throughput efficiency and packet drop time against network size for the RTS/CTS access scheme. The use of the RTS/CTS appears to be more robust and weakly depends on the number of stations for any medium data rate. Moreover, packet drop time is significantly affected when a relatively low data rate of $C = 1$ Mbit/s is employed and increases as network size grows. When a higher data rate is used ($C = 5.5$ or 11 Mbit/s), network size does not considerably influence packet drop time, which is not affected noticeably when $n \geqslant 25$.

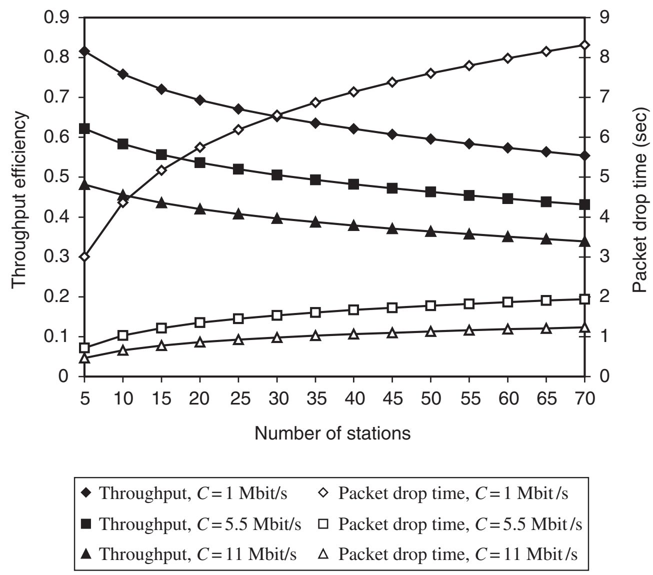

Figure 14. Throughput efficiency and packet drop time for various data rates and basic access.

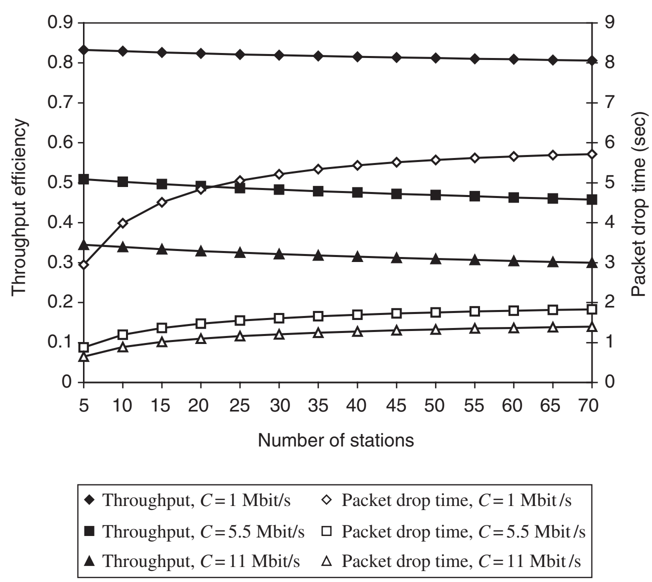

Figure 15. Throughput efficiency and packet drop time for various data rates and RTS/CTS.

7. CONCLUSIONS

This paper presents a simple and intuitive performance analysis that calculates throughput, packet delay, packet drop probability and packet drop time for the IEEE 802.11 protocol.

Performance results obtained from our analysis fully agree with OPNET simulations. Comparison with a model that considers consecutive packet transmissions without activating the backoff procedure reveals that this model overestimates packet delay and does not appear to model IEEE 802.11 correctly, leading to ambiguous conclusions for its performance. Our work becomes important and meaningful in the sense that it predicts 802.11 protocol performance very accurately. According to the results of our analytical approach, the initial contention window size, the maximum CW size and the data rate considerably affect performance of both access mechanisms. High values of initial contention window size improve performance in terms of lower packet drop probability and higher throughput values but increase packet drop time and in certain cases packet delay. Moreover, the increase on the maximum CW size enhances performance since the number of packet collisions is significantly decreased. Furthermore, increasing the data rate in which packets are transmitted results in a considerable degradation of packet delay and packet drop time. Conversely, the increase in data rate does not affect by any means packet drop probability, and increases throughput but causes the decrease of throughput efficiency. This is justified since control packets are always transmitted at the low control rate causing a considerable communication delay. Finally, performance results suggest that the RTS/CTS scheme proves its superiority and is extremely beneficial with respect to the basic access mechanism, mainly in large network scenarios and low data rates.

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