

Performance Analysis of IEEE 802.11 DCF in Imperfect Channels

Yu Zheng, Kejie Lu, Dapeng Wu, and Yuguang Fang

Abstract—IEEE 802.11 is the most important standard for wireless local area networks (WLANs). In IEEE 802.11, the fundamental medium access control (MAC) scheme is distributed coordination function (DCF). To understand the performance of WLANs, it is important to analyze IEEE 802.11 DCF. Recently, several analytical models have been proposed to evaluate the performance of DCF under different incoming traffic conditions. However, to the best of our knowledge, there is no accurate model that takes into account both the incoming traffic loads and the effect of imperfect wireless channels, in which unsuccessful packet delivery may occur due to bit transmission errors. In this paper, we address this issue and provide an analytical model to evaluate the performance of DCF in imperfect wireless channels. In our study, we consider the impact of different factors together, including the binary exponential backoff mechanism in DCF, various incoming traffic loads, distribution of incoming packet size, queueing system at the MAC layer, and the imperfect wireless channels, which has never been done before. Extensive simulation and analysis results show that our analytical model can accurately predict the delay and throughput performance of IEEE 802.11 DCF under different channel and traffic conditions.

Index Terms—Wireless LAN, IEEE 802.11, MAC, DCF, analysis, throughput, delay

I. INTRODUCTION

Wireless local area networks (WLANs) have been widely deployed in recent years. In WLANs, the most important standard is IEEE 802.11 [1], in which the fundamental medium access control (MAC) scheme is distributed coordination function (DCF), which is a carrier sense multiple access with collision avoidance (CSMA/CA) protocol. To better understand the performance of WLANs, a critical challenge is how to analyze IEEE 802.11 DCF. This topic has attracted a lot of research interests in the literature.

In [2]–[4], the authors studied the performance of IEEE 802.11 DCF in a simplified scenario, where every node in the network always has a packet to transmit at any time, known as the *saturated condition*. With the saturation assumption, these studies can accurately model the behavior of the binary

exponential backoff mechanism used in DCF and can provide insightful results. However, the saturation assumption may not be valid in practice since the number of packets to be transmitted depends on the incoming traffic loads.

Recently, a number of models have been proposed in the literature to address the performance of IEEE 802.11 DCF in more general, *unsaturated* traffic conditions [5]–[10]. The models in [5] and [6] are direct extensions of the saturated model in [2] in that both of them inherit the same discrete-time Markov chain model introduced in [2]. In these two models, the unsaturated traffic conditions are modelled by adding one or more *idle* states that represent the situation that there are no packets to be transmitted. Although these two models take into account the unsaturated traffic conditions, both of them assume that the queue length of the MAC system is zero, which may not be practical. Similar assumption is also used in [7], which applies the linear feedback model introduced in [11]. Clearly, the models in [2]–[7] cannot provide accurate delay analysis since they all ignore the queueing system at the MAC layer.

The queueing behavior of IEEE 802.11 DCF is studied in [8]–[10]. The model in [8] is based on a G/G/1 queue. Due to computational complexity, this model depends on several approximated parameters, such as the probability that a node has no packet to transmit. Consequently, the analysis results have a large deviation in comparison to the simulation results. Moreover, this model is not suitable for high traffic load condition since the queue size is assumed to be infinite, which may not be valid in practice.

The finite capacity of the queue is studied in [9] and [10]. The models in [9] and [10] are common in that both of the analyses are based on the M/G/1/K queue. In addition, both of them require iterative algorithms. This is because, to solve the M/G/1/K model, the service time distribution is required; while to calculate the service time distribution, a required parameter is the probability that a node has no packet to transmit, which can be achieved by solving the M/G/1/K model. The main difference of these two models is how to calculate the service time distribution. In [9], Ozdemir and McDonald proposed to use the Markov modulated general independent (MMGI) model. In contrast, Zhai, Kwon and Fang [10] used a transfer-function approach to calculate the service time distribution directly. In [12], we developed a more accurate and tractable algorithm using a similar technique as [10].

From the discussion above, we note that an important and realistic condition — imperfect wireless channels, has not been addressed adequately. The only study that takes the wireless channel errors into account is [4], where the analysis,

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however, is based on the saturated condition. To the best of our knowledge, there is no analytical model that considers the channel error conditions in the unsaturated performance analysis for IEEE 802.11 DCF.

In this paper, we provide an analytical model to evaluate the performance of DCF in imperfect wireless channels. In this study, we consider the impact of different factors together, including the binary exponential backoff mechanism in DCF, various incoming traffic loads, distribution of incoming packet size, queueing system at the MAC layer, and the imperfect wireless channels, which has never been done before. Extensive simulation and analysis results show that our analytical model can accurately predict the delay and throughput performance of IEEE 802.11 DCF under different channel and traffic conditions.

The rest of the paper is organized as follows. In Section II, we first provide an overview of the system at the MAC layer; we then briefly describe the access mechanism of IEEE 802.11 DCF. In Section III we will focus on the analytical model for the unsaturated performance of DCF in a realistic WLAN. Simulation and numerical result will be shown in Section IV. Finally, Section V concludes the paper.

II. SYSTEM OVERVIEW

In this section, we briefly describe the MAC protocol and the DCF functions in IEEE 802.11 to facilitate the analysis in the next section.

A. The MAC Protocol

In this study, we consider there is a queue at the MAC layer. Specifically, we assume that the queue can store a finite number of K packets. In addition to the queue, we consider there is a transmission buffer at the MAC layer, in which a packet can be temporarily stored and waiting for transmission. It is important to note that at most $K + 1$ packet can be stored in the system at a certain time.

To simplify the discussion, in this paper we consider only one class of traffic. With this assumption, the function of the MAC protocol can be described as follows. When a packet arrives at the MAC layer in the source node, it will be dropped if the queue is full; it will be put in the transmission buffer directly if the buffer is empty; and it will be put at the tail of the queue otherwise. All queued packets are served in a first in first out (FIFO) manner, which means that the MAC will move the head-of-queue packet into the transmission buffer after a packet transmission is finished. The packet transmission procedure is defined by the IEEE DCF.

B. IEEE 802.11 Distributed Coordination Function

In IEEE 802.11 DCF, there are two options for medium access: the basic access scheme and the request-to-send/clear-to-send (RTS/CTS) scheme. The basic scheme uses DATA/ACK two-way handshaking to determine whether the DATA packet is successfully transmitted in the channel; while the RTS/CTS mechanism tries to reserve the channel by smaller control packets (RTS and CTS) before the DATA transmission. The

main functions for both channel access schemes are common. These functions include the carrier sensing mechanism, the virtual carrier sensing mechanism, and the binary exponential backoff mechanism.

In the carrier sensing mechanism, a node that has packet to send will continuously sense if the channel is busy. If the channel is idle, the node will start or resume the backoff procedure. If the channel is busy, it will wait until the channel becomes idle, which means that there is no transmission in a certain duration, denoted as DIFS, which depends on the physical layer specification. For example, in IEEE 802.11b direct sequence spread spectrum (DSSS) mode [13], DIFS is set to $50 \mu s$.

In the virtual carrier sensing mechanism, a node will set up a timer, namely, the network allocation vector (NAV), if it receives a packet that indicates the duration of a transmission between two other nodes. The node will resume its backoff procedure after the NAV timeout.

In the binary exponential backoff mechanism, a node will transmit its packet only if its backoff counter is zero. The backoff counter is an integer value that is uniformly chosen, within $[0, CW-1]$ where CW denotes the contention window size, when the node receives a new packet from the upper layer and when the node notices that its transmission has failed. During the backoff procedure, the backoff counter will be decreased by one after the channel has been idle for a certain duration called time slot, denoted as σ . Similar to the setting of DIFS, σ also depends on the physical layer specification. For instance, σ is set to $20 \mu s$ in IEEE 802.11b DSSS mode. The value of CW depends on the status of backoff. Particularly, when a node receives a packet from upper layer, CW is set to the minimum value CW_{min} . Before each re-transmission, CW will be doubled until it reaches the maximum value CW_{max} .

Let $W = CW_{min}$, let $M' = \log_2 (CW_{max}/CW_{min})$, let M be the retry limit for a packet, and let W_m be the CW on backoff stage m . We can summarize the contention window as follows.

$$W_m = \begin{cases} 2^m W & 0 \leq m \leq M' \\ 2^{M'} W & M' \leq m \leq M. \end{cases} \quad (1)$$

Finally, it is worth noting that the finish of a transmission can occur in two cases: the transmission is successful and the transmission is a failure after a certain number of retries.

III. ANALYTICAL MODEL FOR IEEE 802.11 DCF

In this section, we present an analytical model to evaluate the performance of IEEE 802.11 DCF with imperfect channel conditions. In this model, we also take into consideration a number of realistic conditions, such as incoming traffic loads, packet size distribution, and the queueing behavior. Similar to [9] and [10], in our study we will decompose the MAC into two subsystems: the *queueing subsystem* that takes care of the queueing behavior based on the M/G/1/K model; and the *service subsystem* that characterizes the service time distribution.

The rest of this section is organized as the following. We first give the assumptions of the analytical model in Section III-A, followed by the description of the iterative

algorithm in Section III-F. We then elaborate on the queueing subsystem in Section III-C and on the service subsystem in Section III-D, respectively. Finally, in Section III-E we will discuss how to achieve the throughput and delay performance.

A. Assumptions

To facilitate our discussion, we make the following assumptions:

- There are N identical nodes in the network.
- At each node, packet arrivals are Poisson with rate λ (packets/sec). Here we notice that the same assumption has been widely used to keep to tractability of the analytical model [9], [10].
- The size of the packets (in bytes) from the upper layer is a random variable with probability distribution $f(n)$, where $f(n) = 0$ for $n < N_{min}$ or $n > N_{max}$. In other words, the length of any packet is bounded.
- The queue at the MAC layer can store up to K packets, which does not include the packet in the transmission buffer.
- The MAC header and the data packet are transmitted with rate R_d (in bits/second), while RTS, CTS, ACK packets and the preambles are transmitted with rate R_c (in bits/second).
- The channel is not perfect. Every bit within the transmitted data packets encounters error with a fixed probability ϵ . To simplify the discussion we assume that control packets and frame headers of data packets are error free.
- The probability that one transmission attempt of a packet fails, denoted as p , does not depend on the backoff stage of the node.
- The packet service time is an integer multiple of a preset time unit τ (in seconds). This integer has an upper bound I_{max} as a server only tries to send one packet for finite number of times and each time the attempt has a finite duration.
- To simplify the discussion, we assume that the propagation delay is negligible.

B. Notation

We list the key notations we are going to use as follows.

- p_I denotes the probability that a node has no packet to transmit in one time slot. Here we follow [2] and partition the continuous time axis into slots, where two consecutive slots are delimited by the event of a value change in the backoff counter.
- p_t denotes the probability that a node will transmit in one time slot.
- p_k^d denotes the steady state probability that there are k packets left in the queue at the time instance just before a packet departures. Here we note that the departure in this paper means that the transmission is finished.
- q_i denotes the steady state probability that the packet service time is $i\tau$.
- S is the MAC throughput.
- T is the average packet delay at the MAC layer, including the queueing delay and the service delay.

C. The Queueing Subsystem

Based on the assumptions in Section III-A, the queueing system can be modelled as $M/G/1/K$. Following [12], we let $\xi(t)(t \geq 0)$ be the state of the queueing system at time t . The state space of $\xi(t)$ can then be defined as

$$S = \{I, A_0, A_1, A_2, \dots, A_K\}, \quad (2)$$

where A_k means that the server is busy and there are k packets waiting in the queue; I means the server is idle, or in other words, the queue and the transmission buffer are empty.

Let δ_n be the time instance of the n -th packet departure. We now consider the embedded Markov process ξ_n , where ξ_n is the state of the queueing system just before δ_n , which is

$$\xi_n = \xi(\delta_n^-). \quad (3)$$

This embedded Markov chain has state space $S' = S - I = \{A_0, A_1, A_2, \dots, A_K\}$. Let p_{ij} be the steady-state transition probability from state A_i to A_j for $\forall i, j \in [0, K]$, i.e.:

$$p_{ij} = \lim_{n \rightarrow \infty} Pr[\xi_{n+1} = A_j | \xi_n = A_i]. \quad (4)$$

To calculate p_{ij} , we can use the service time distribution and the packet arrival rate. Define $\alpha(k)$ as the probability that k packets arrive during one packet service time. Since the packet arrival is a Poisson process with rate λ , we have

$$\alpha(k) = \sum_{i=0}^{I_{max}} q_i \frac{(\lambda i \tau)^k e^{-\lambda i \tau}}{k!}. \quad (5)$$

Consequently, p_{ij} can be calculated as

$$p_{ij} = \begin{cases} \alpha(j) & i = 0, j < K \\ 1 - \sum_{k=0}^{K-1} \alpha(k) & i = 0, j = K \\ 0 & i > 0, j < i - 1 \\ \alpha(j - i + 1) & i > 0, j < K \\ 1 - \sum_{k=0}^{K-1} \alpha[k - i + 1] & i > 0, j = K. \end{cases} \quad (6)$$

Now let p_k^d ($0 \leq k \leq K$) be the steady-state probability that $\xi_n = A_k$. Clearly, p_k^d can be obtained by solving the embedded Markov chain with all p_{ij} .

Based on the M/G/1/K model, we can also calculate p_I through

$$p_I = \frac{p_0^d}{\lambda T^s + p_0^d}. \quad (7)$$

where T^s denotes the average packet service time, which is

$$T^s = \sum_{i=1}^{I_{max}} q_i \times (i\tau). \quad (8)$$

D. The Service Subsystem

In this subsection, we first analyze the binary exponential backoff scheme of the DCF protocol using the Markov modelling technique introduced in [2]; we then calculate the service time distribution by the transfer-function approach [10], [12].

1) *Binary Exponential Backoff*: Similar to [3], we can formulate a two-dimensional discrete time embedded Markov chain with state $\{s_n, b_n\}$, where n is the index of a slot, b_n is the value of the backoff counter in slot n , and s_n is the index of the backoff stage in slot n . Let the steady state probability of state $\{s_n = m, b_n = i\}$ be

$$b_{m,i} = \lim_{n \rightarrow \infty} Pr[s_n = m, b_n = i]. \quad (9)$$

Based on the the Markov chain, a closed-form solution for all $b_{m,i}$ can be derived. Since a transmission is initiated in slot n if and only if $b_n = 0$, we can obtain the first relationship between p_t and p as Eq. (10).

In addition to the relationship between p and p_t described in Eq. (10), we note that a successful packet delivery can occur only when there is neither collision nor bit error in a transmission attempt. Therefore, we can calculate p through

$$p = 1 - (1 - p_c)(1 - p_e), \quad (11)$$

where p_c is the collision probability in any slot and p_e is the packet error probability, which can be calculated through Eq. (12) and Eq. (13), respectively.

$$p_c = 1 - [1 - (1 - p_I)p_t]^{N-1}. \quad (12)$$

$$p_e = 1 - \sum_{n=N_{min}}^{N_{max}} f(n)(1 - \varepsilon)^{8n}. \quad (13)$$

By solving Eqs. (10) and (11), we can get p_t and p with a given p_I .

2) *Service Time Distribution*: Let $Q(z)$ be the *probability generating function* (PGF) of q_i , which is

$$Q(z) = \sum_i z^i \cdot q_i. \quad (14)$$

Due to the simplicity of notation in the z -transform domain and the one-to-one correspondence between $Q(z)$ and $\{q_i\}$, we will discuss how to calculate $Q(z)$ instead of individual q_i .

Similar to [12], we let X_n be the length of slot n and let X'_n be the length of the time interval (within slot n) during which the server is busy. Note that for saturated condition $X_n \equiv X'_n$, while for unsaturated cases $X'_n \leq X_n$. We can then apply the transfer-function approach, in which the packet transmission process is characterized by a linear system, shown in Fig. 1.

In Fig. 1, $C(z)$ denotes the PGF of X'_n given that a collision occurred when the current node transmits a packet; $S(z)$ denotes the PGF of X'_n given that the current node has successfully transmitted a packet; and $H(z)$ denotes the PGF of X'_n given that the server of current node is busy but not transmitting. To simplify the notation, in Fig. 1, we define $H_m(z)$ as following

$$H_m(z) = \frac{1}{W_m} \sum_{i=0}^{W_m-1} H^i(z), \quad 0 \leq m \leq M. \quad (15)$$

From Fig. 1 we can derive the transfer function of the linear system, which is $Q(z)$, as Eq.(16).

We now discuss how to obtain $C(z)$, $S(z)$, and $H(z)$. To simplify the discussion, we use only the RTS/CTS scheme defined in [13] hereafter. Since the RTS/CTS scheme is used, the collision can only occur amongst RTS packets. Therefore, we can derive

$$C(z) = z^{\lfloor \frac{T_{co}}{\tau} \rfloor} \quad (17)$$

where T_{co} is the time overhead for a collision. According to IEEE 802.11b, T_{co} can be calculated through

$$T_{co} = 2T_{sync} + T_{SIFS} + T_{DIFS} + \frac{1}{R_c}(2L_{PH} + L_{RTS} + L_{CTS}), \quad (18)$$

where T_{sync} denotes the synchronization time, T_{SIFS} denotes the time duration of SIFS, T_{DIFS} denotes the time duration of DIFS, L_{PH} denotes the length of the physical frame header (excluding the synchronization preamble), L_{RTS} denotes the length of the RTS packet, and L_{CTS} denotes the length of the CTS packet. Here we note that all the values of length are in bits.

To calculate $S(z)$, we can use

$$S(z) = \sum_{n=N_{min}}^{N_{max}} f(n)z^{\lfloor \frac{1}{\tau} \times (T_{so} + 8n/R_d) \rfloor}, \quad (19)$$

where T_{so} is the time overhead for a successful transmission or an error transmission. According to IEEE 802.11b, we have

$$T_{so} = 4T_{sync} + 3T_{SIFS} + T_{DIFS} + \frac{1}{R_d} \times L_{MH} + \frac{1}{R_c}(4L_{PH} + L_{RTS} + L_{CTS} + L_{ACK}), \quad (20)$$

where L_{MH} denotes the length of the MAC frame header and L_{ACK} denotes the length of the ACK packet.

To calculate $H(z)$, we define the following probabilities as functions of p and p_t , given that the current node is not going to transmit in slot n :

- q_t denotes the probability that there is at least one packet transmission in $N - 1$ other nodes in slot n , which is

$$q_t = 1 - [1 - (1 - p_I)p_t]^{N-1}. \quad (21)$$

- q_s denotes the probability that there is only one packet transmission in $N - 1$ other nodes in slot n , which is

$$q_s = (N - 1) [(1 - p_I)p_t] [1 - (1 - p_I)p_t]^{N-2}. \quad (22)$$

Finally, we have

$$H(z) = (1 - q_t)z^{\lfloor \frac{\sigma}{\tau} \rfloor} + q_s S(z) + (q_t - q_s)C(z). \quad (23)$$

E. Throughput and Delay

Based on the M/G/1/K model, we can calculate the throughput of the system through

$$S = \frac{\lambda(1 - p^{M+1})\bar{P}}{\lambda T^s + p_0^d}, \quad (24)$$

where \bar{P} is the average packet length in bits and can be calculated through

$$\bar{P} = \sum_{n=N_{min}}^{N_{max}} 8nf(n). \quad (25)$$

$$p_t = \sum_{m=0}^M b_{m,0} = \begin{cases} \frac{2(1-2p)(1-p^{M+1})}{(1-2p)(1-p^{M+1}) + W(1-p)[1-(2p)^{M+1}]} & M \leq M' \\ \frac{2(1-2p)(1-p^{M+1})}{(1-2p)(1-p^{M+1}) + W(1-p)[1-(2p)^{M'+1}] + W2^{M'}p^{M'+1}(1-2p)(1-p^{M-M'})} & M > M'. \end{cases} \quad (10)$$

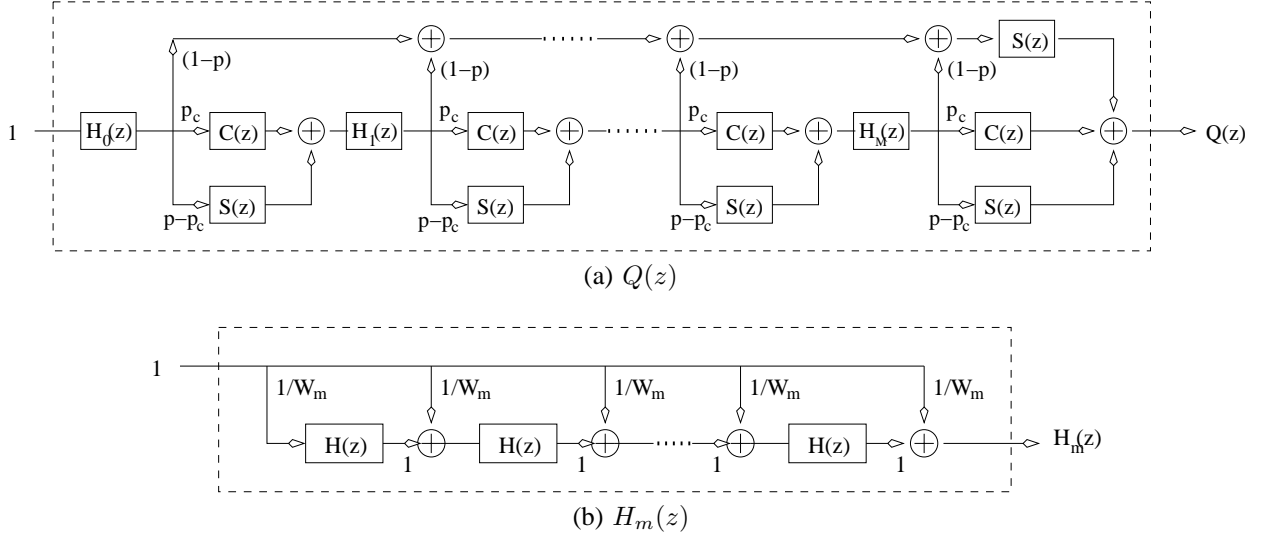


Fig. 1. Service system diagram.

$$Q(z) = (1-p_c)S(z) \sum_{m=0}^M \left[[p_c C(z)]^m \prod_{i=0}^m H_i(z) \right] + [p_c C(z)]^{M+1} \times \prod_{i=0}^M H_i(z). \quad (16)$$

Using the result for finite M/G/1/K queue [14], taking into consideration that our system has a transmission buffer, we can relate the probability of queue length seen by arriving packets and the steady state probability for the embedded Markov chain by the following equations

$$p_i = \frac{p_k^d}{p_0^d + \lambda T^s}, \quad 0 \leq i \leq K$$

$$p_{K+1} = 1 - \frac{1}{\lambda T^s + p_0^d},$$

therefore in a similar way as [9] we can calculate the average packet delay T as

$$T = \frac{1}{\lambda} \left[\sum_{k=1}^K k p_k^d + (K+1)(\lambda T^s + p_0^d - 1) \right]. \quad (26)$$

F. The Iterative Algorithm

To calculate the performance metrics of IEEE 802.11 DCF, we apply the iterative algorithm illustrated in Fig. 2.

The iterative steps are outlined as follows:

- Step 1: Initialize $p_I = 0$, which is the saturated condition.
- Step 2: With p_I , calculate p_t and p according to the model for binary exponential backoff, which will be discussed in Section III-D.1.
- Step 3: Calculate q_i through the transfer-function approach using p_t and p , which will be discussed in Section III-D.2.

- Step 4: Calculate p_k^d based on the M/G/1/K model, which will be discussed in Section III-C.
- Step 5: Calculate a new p_I based on p_k^d , which will also be discussed in Section III-C.
- Step 6: Calculate the throughput S and delay T as shown in Section III-E.
- Step 7: If S and T converge with the previous values, then stop the algorithm; otherwise go to Step 2 with the updated p_I .

In the above subsections, we have discussed how to calculate all parameters list above. It is important to note that, although we have not been able to prove the convergence of the algorithm, the algorithm can always yield converged results in practice.

IV. SIMULATION AND NUMERICAL RESULTS

In this section, we evaluate the performance of IEEE 802.11 DCF under different channel and traffic conditions through simulation and analytical results. Table I lists the values of the control parameters used in the simulations and numerical analysis.

In addition to the setting in Table I, we assume that all nodes in the network are located in a small area so that the propagation delay can be ignored. In our experiments, we let the packet arrivals to any node be a Poisson process with the same rate λ (packets/s). Consequently, the total incoming

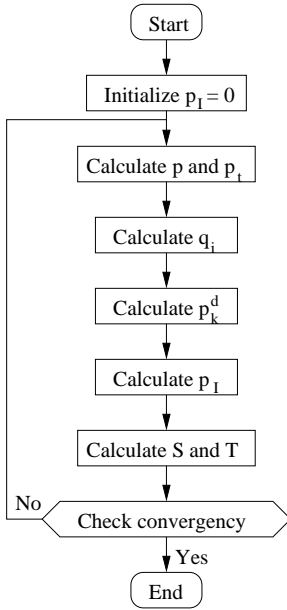


Fig. 2. The iterative algorithm.

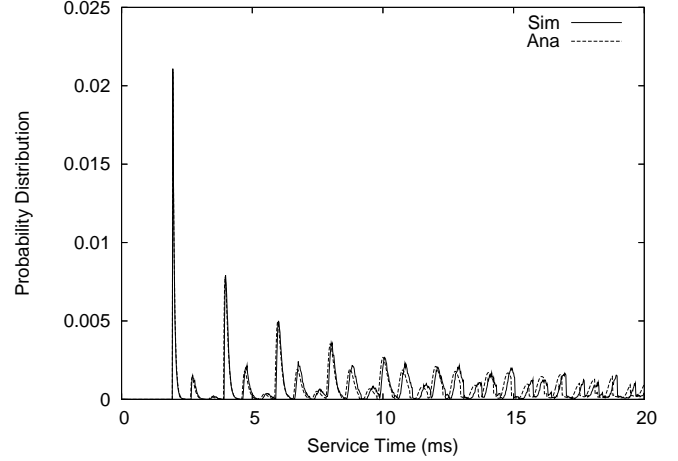
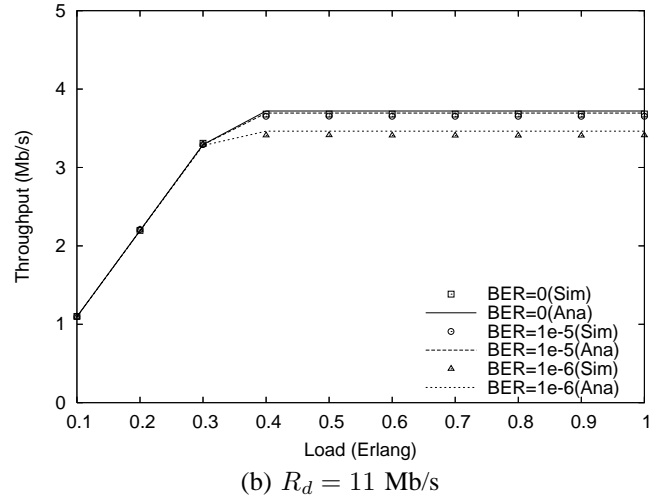
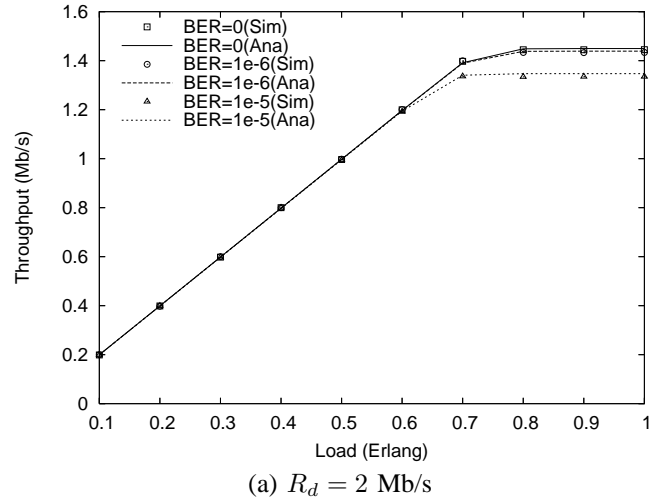
TABLE I
SETTING OF IEEE 802.11 DCF.

Minimum contention window size	31
Maximum contention window size	1023
σ	20 μ s
SIFS	10 μ s
DIFS	50 μ s
Access scheme	RTS/CTS
Retry limit	4
Queue size	50

traffic data rate is $R_i = N\bar{P}\lambda$ (bits/s). We further define the total incoming traffic load as $\rho = R_i/R_d$. Unless specified otherwise, we assume that the size of all packet are fixed to 1000 bytes. For the analytical model, we let time unit $\tau = \sigma$, and let the maximum service time be 60000 time units.

Fig. 3 compares the simulation and analytical results of the service time distribution of IEEE 802.11 DCF, where we let $N = 10$, $R_d = 11$ Mb/s, and $\rho = 1$ Erlang. We can observe that the results from our analytical model have a good match with the simulation results, which validates the sampling technique we utilize in the analytical model.

Fig. 4 shows the throughput versus traffic load of DCF with different channel bit error ratios. It can be observed that, our analytical model can accurately predict the throughput performance of DCF under different traffic and channel conditions. From Fig. 4 we can see a common trend for all channel conditions: if the traffic load is small, the overall throughput of DCF will be equal to the increase of incoming traffic data rates; and if the traffic load is higher enough, the throughput will become saturated. We can further observe from Fig. 4 that larger channel bit error probability will lead to smaller saturated throughputs. Particularly, in Fig. 4 (a), the throughput for $\rho = 1$ Erlang is about 1.41 Mb/s for $\epsilon = 10^{-6}$, and is about 1.32 Mb/s for $\epsilon = 10^{-5}$. Another interesting observation is

Fig. 3. Service time distribution ($N = 10$, $R_d = 11$ Mb/s, and $\rho = 1$ Erlang).Fig. 4. Throughput vs. Load ($N = 10$).

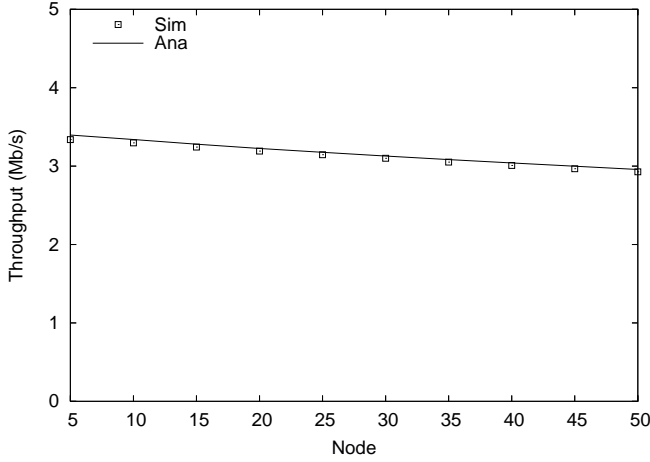


Fig. 5. Throughput vs. Node ($BER=10^{-5}$, $R_d = 10$ Mb/s, Load=1 Erlang).

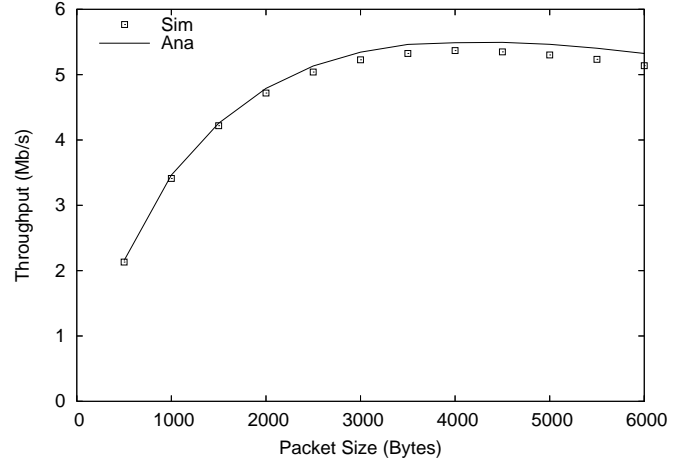


Fig. 7. Throughput vs. Packet Size ($BER=10^{-5}$, $R_d = 11$ Mb/s, Load=1 Erlang).

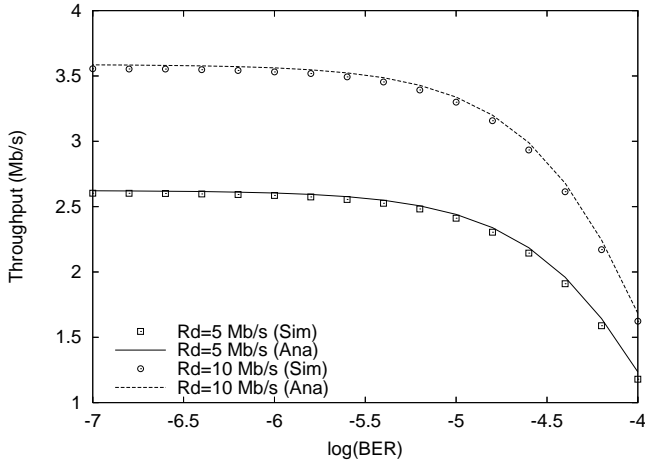


Fig. 6. Throughput vs. BER ($N = 10$, Load=1 Erlang).

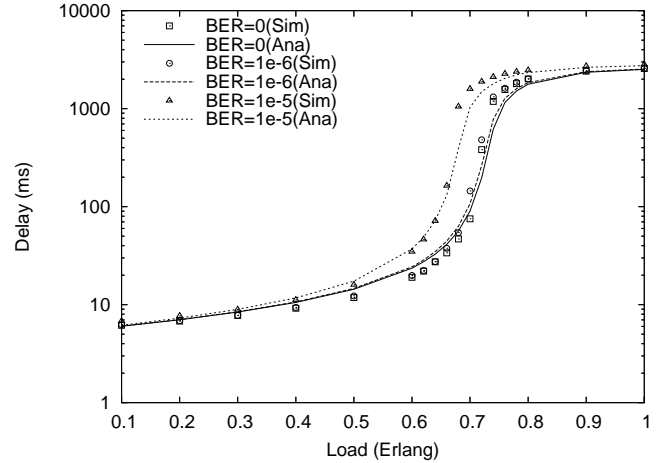


Fig. 8. Average delay vs. loads ($N = 10$, $R_d = 2$ Mb/s).

that, the relative throughput, i.e. the ratio of throughput to R_d decreases with the increase of R_d . This phenomenon is primarily because the control overhead is relatively larger for a larger R_d .

Fig. 5 and Fig. 6 demonstrate that our analytical model is also accurate for different number of nodes in the networks, as well as various BER conditions. In addition, Fig. 6 also demonstrate the impact of BER, where we let $N=10$ and Load=1 Erlang. We can observe that, for both $R_d = 5$ Mb/s and 10 Mb/s, the throughput of DCF only decreases slightly with the increase of BER if the BER is less than $10^{-5.5}$. However, if the BER is greater than $10^{-5.5}$, then the throughput will decrease dramatically.

In Fig. 4, we have discovered that the relative throughput will be decreased if the channel data rate is higher. To overcome this problem, a possible approach is to increase the average length of incoming packets. However, as the size of packets increases, the probability that a certain transmission is failure due to bit errors is also increase, which will lead to a degradation of throughput performance. Intuitively, there may exist an optimal packet size which can result in the maximum

throughput. This intuition is confirmed in Fig. 7, where we let $BER=10^{-5}$ and $R_d = 11$ Mb/s. We can observe from Fig. 7 that the maximum throughput can be achieved if the packet size is about 4000 Bytes.

Finally, in Fig. 8, we show the delay performance versus traffic loads under different BER conditions, where we apply the same setting as that of Fig. 4 (a). We can observe that, our model can accurately predict the delay performance, in addition to the throughput performance.

V. CONCLUSION

In this paper, we provide an accurate analytical model to evaluate the performance of distributed coordination function (DCF), which is the fundamental medium access control (MAC) scheme in IEEE 802.11. The main contribution of our study is that we consider the impact of different realistic factors together, including binary exponential backoff, various incoming traffic loads, queueing system at the MAC layer, and the imperfect wireless channels, which has never been addressed in a comprehensive manner before. Extensive simulation and analysis results show that our analytical model

can accurately predict the delay and throughput performance of IEEE 802.11 DCF under different channel and traffic conditions.

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