Performance Analysis of IEEE 802.11 DCF in Imperfect Channels

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***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 the 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 the authors' 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, the authors address this issue and provide an analytical model to evaluate the performance of DCF in imperfect wireless channels. The authors 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 the proposed analytical model can accurately predict the delay and throughput performance of IEEE 802.11 DCF under different channel and traffic conditions.**

***Index Terms*—Analysis, delay, distributed coordination function (DCF), IEEE 802.11, medium access control (MAC), throughput, wireless local area networks (LANs).**

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], wherein the fundamental medium access control (MAC) scheme is the 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 many research interests in the literature.

In [2]–[4], the performance of IEEE 802.11 DCF was studied 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 modeled by adding one or more "idle" states that represent the situation where 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. A 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 with the simulation results. Moreover, this model is not suitable for high traffic load conditions 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 *et al.* [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 Zhai *et al.* [10].

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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, 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 this paper is organized as follows: In Section II, we first provide an overview of the system at the MAC layer and 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 results will be shown in Section IV. Finally, Section V concludes this 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. MAC Protocol

In this study, we consider that 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 that 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 $K+1$ packets 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, will be put in the transmission buffer directly if the buffer is empty, and 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 DCF

In IEEE 802.11 DCF, there are two options for medium access, namely 1) the basic access scheme and 2) 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 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 μs.

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 $[0, \mathrm{CW}-1]$, 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 μs in IEEE 802.11b DSSS mode. The value of CW depends on the status of backoff. Particularly, when a node receives a packet from the upper layer, CW is set to the minimum value CWmin. Before each retransmission, CW will be doubled until it reaches the maximum value CWmax.

Letting $W = \mathrm{CWmin}$, $M' = \log_2$ (CWmax/CWmin), M be the retry limit for a packet, and W_m be the CW on the backoff stage m, we can summarize the CW as

$$W_m = \begin{cases} 2^m W, & 0 \le m \le M' \\ 2^{M'} W, & M' \le m \le M \end{cases}. \tag{1}$$

Finally, it is worth noting that the end of a transmission can occur in two cases, namely 1) the transmission is successful and 2) 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 queueing behavior. Similar to [9] and [10], in our study, we will decompose the MAC into two subsystems, namely 1) the "queueing subsystem" that takes care of the queueing behavior based on the M/G/1/K model, and 2) the "service subsystem" that characterizes the service time distribution.

The rest of this section is organized as follows: We first give the assumptions of the analytical model in Section III-A and 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. Finally, in Section III-E, we will discuss how to achieve 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 (in packets per second). Here, we notice that the same assumption has been widely used to keep the 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 per second), while RTS, CTS, ACK packets, and the preambles are transmitted with rate R_c (in bits per 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. Queueing Subsystem

Based on the assumptions in Section III-A, the queueing system can be modeled 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

$$S = \{I, A_0, A_1, A_2, \ldots, A_K\} \tag{2}$$

where A_k means that the server is busy and there are k packets waiting in the queue, and 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 nth 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

$$\xi_n = \xi(\delta_n^-). \tag{3}$$

This embedded Markov chain has state space $S' = S - I = \{A_0, A_1, A_2, \ldots, A_K\}$. 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.,

$$p_{ij} = \lim_{n\to\infty} \Pr\left[\xi_{n+1} = A_j | \xi_n = A_i\right]. \tag{4}$$

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

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

Consequently, p_{ij} can be calculated as

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

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}.

$$p_t = \sum_{m=0}^{M} b_{m,0} = \begin{cases} \dfrac{2(1-2p)(1-p^{M+1})}{(1-2p)(1-p^{M+1}) + W(1-p)\left[1-(2p)^{M+1}\right]} & M \le M' \\ \dfrac{2(1-2p)(1-p^{M+1})}{(1-2p)(1-p^{M+1}) + W(1-p)\left[1-(2p)^{M'+1}\right] + W2^{M'}p^{M'+1}(1-2p)(1-p^{M-M'})} & M > M'. \end{cases}$$

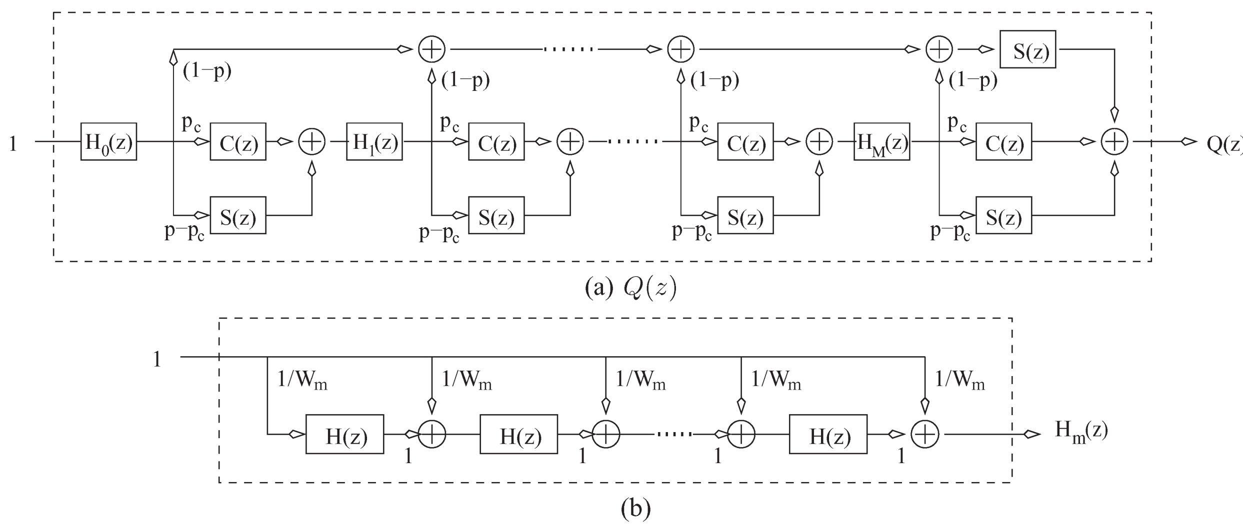

Fig. 1. Service system diagram.

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

$$p_I = \frac{p_0^d}{\lambda T^s + p_0^d} \tag{7}$$

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

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

D. Service Subsystem

In this subsection, we first analyze the binary exponential backoff scheme of the DCF protocol using the Markov modeling technique introduced in [2] and 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 (2-D) 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

$$b_{m,i} = \lim_{n\to\infty} \Pr[s_n = m, b_n = i]. \tag{9}$$

Based on 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, shown in (10) at the bottom of the next page.

In addition to the relationship between p and p_t described in (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

$$p = 1 - (1-p_c)(1-p_e) \tag{11}$$

where p_c is the collision probability in any slot, and p_e is the packet error probability, which can be calculated through

$$p_c = 1 - \left[1 - (1-p_I)p_t\right]^{N-1} \tag{12}$$

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

By solving (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

$$Q(z) = \sum_i z^i \cdot q_i. \tag{14}$$

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 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, $X'_n \le X_n$. We can then apply the transfer-function approach in which the packet transmission process is characterized by a linear system, as 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 the current node is busy but not transmitting. To simplify the notation, in Fig. 1, we define $H_m(z)$ as

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

From Fig. 1, we can derive the transfer function of the linear system as

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

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 among RTS packets. Therefore, we can derive

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

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

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

where $T_{\rm sync}$ denotes the synchronization time, $T_{\rm SIFS}$ denotes the time duration of short inter-frame space (SIFS), $T_{\rm DIFS}$ denotes the time duration of DIFS, $L_{\rm PH}$ denotes the length of the physical frame header (excluding the synchronization preamble), $L_{\rm RTS}$ denotes the length of the RTS packet, and $L_{\rm CTS}$ 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

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

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

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

where $L_{\rm MH}$ denotes the length of the MAC frame header, and $L_{\rm ACK}$ 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

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

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

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

Finally, we have

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

E. Throughput and Delay

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

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

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

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

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

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

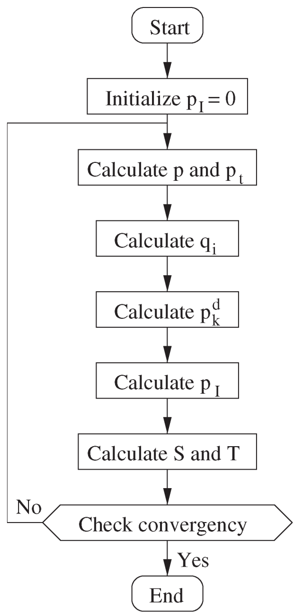

Fig. 2. Iterative algorithm.

and the steady-state probability for the embedded Markov chain by

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

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

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

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

F. 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 the parameters listed 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.

TABLE I
SETTING OF IEEE 802.11 DCF

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

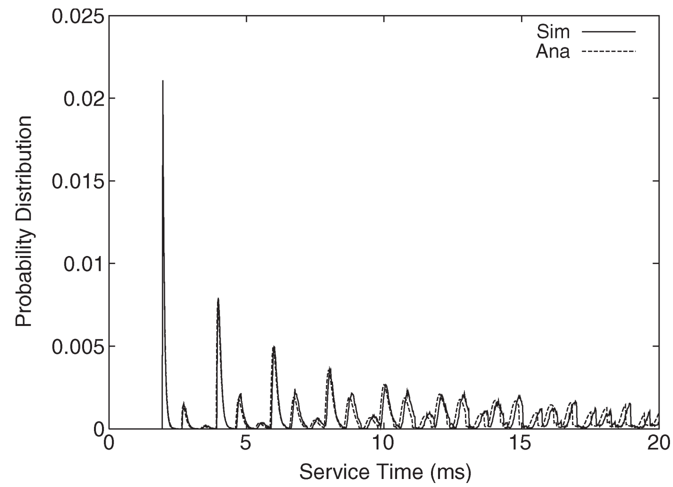

Fig. 3. Service time distribution ($N = 10$, $R_d = 11$ Mb/s, and $\rho = 1$ Erlang).

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 (in packets per second). Consequently, the total incoming traffic data rate is $R_i = N\overline{P}\lambda$ (in bits per second). We further define the total incoming traffic load as $\rho = R_i/R_d$. Unless specified otherwise, we assume that the size of all packets is fixed to 1000 B. For the analytical model, we let the time unit $\tau = \sigma$ and the maximum service time be 60 000 time units.

Fig. 3 compares the simulation and analytical results of the service time distribution of IEEE 802.11 DCF, where we let

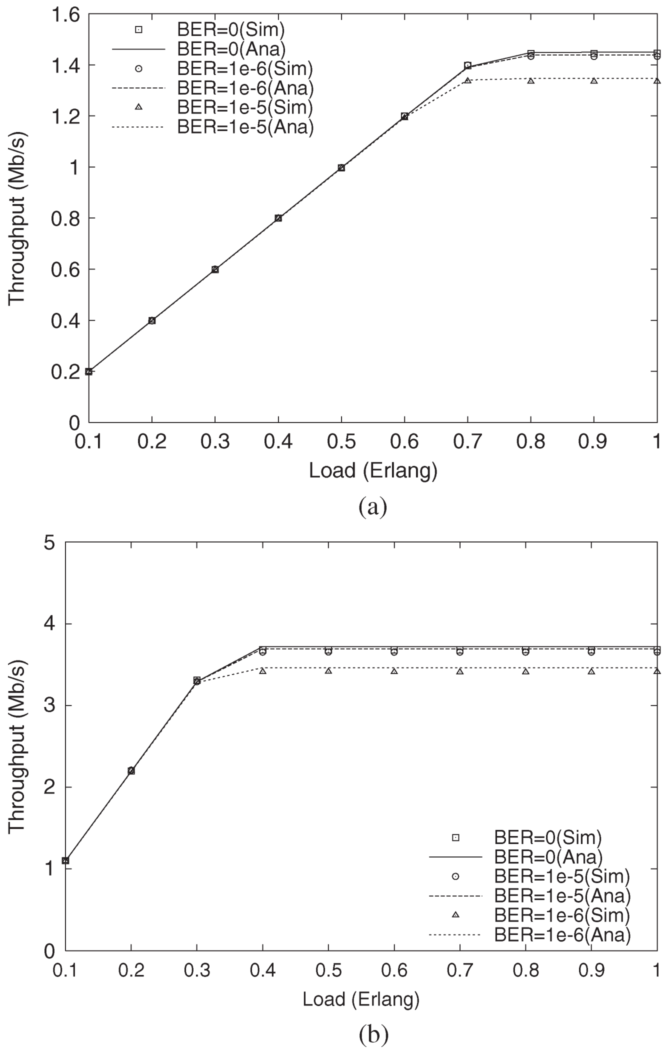

Fig. 4. Throughput versus load ($N = 10$). (a) $R_d = 11$ Mb/s. (b) $R_d = 2$ Mb/s.

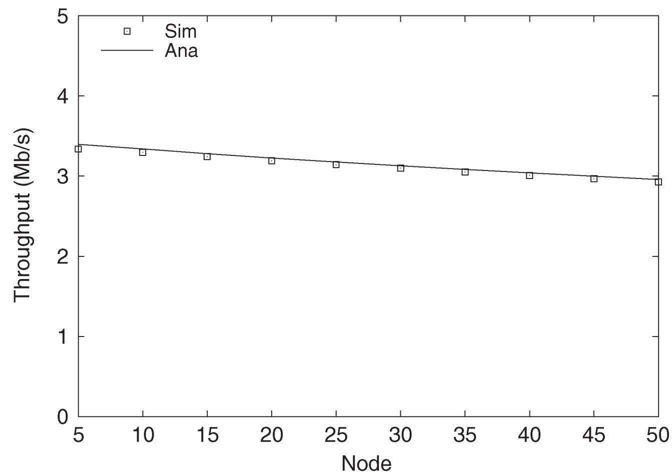

Fig. 5. Throughput versus node (BER $= 10^{-5}$, $R_d = 10$ Mb/s, and Load $=$ 1 Erlang).

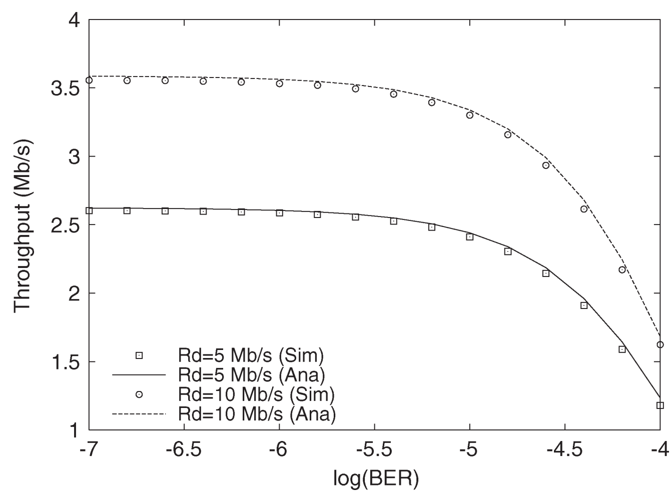

Fig. 6. Throughput versus BER ($N = 10$ and Load $= 1$ Erlang).

$N = 10$, $R_d = 11$ 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 high enough, the throughput will become saturated. We can further observe from Fig. 4 that a 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 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.

Figs. 5 and 6 demonstrate that our analytical model is also accurate for different number of nodes in the networks as well as various bit error rate (BER) conditions. In addition, Fig. 6 also demonstrates the impact of BER, where we let $N = 10$ and Load $= 1$ Erlang. We can observe that, for both $R_d = 5$ 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 a failure due to bit errors also increases, which will lead to a degradation of throughput performance. Intuitively, there may exist an optimal packet size that 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 B.

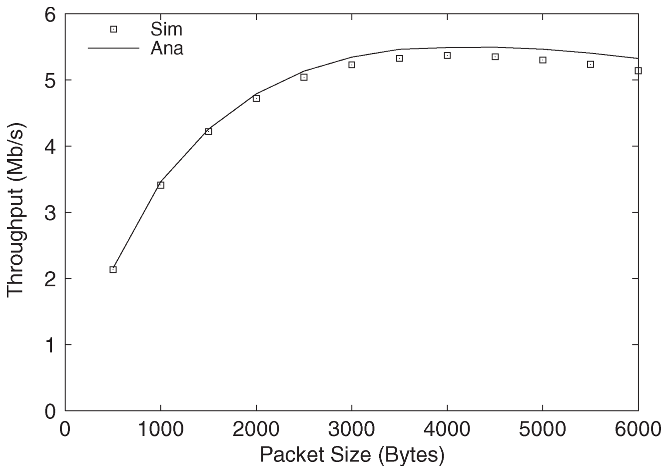

Fig. 7. Throughput versus packet size (BER $= 10^{-5}$, $R_d = 11$ Mb/s, and Load $= 1$ Erlang).

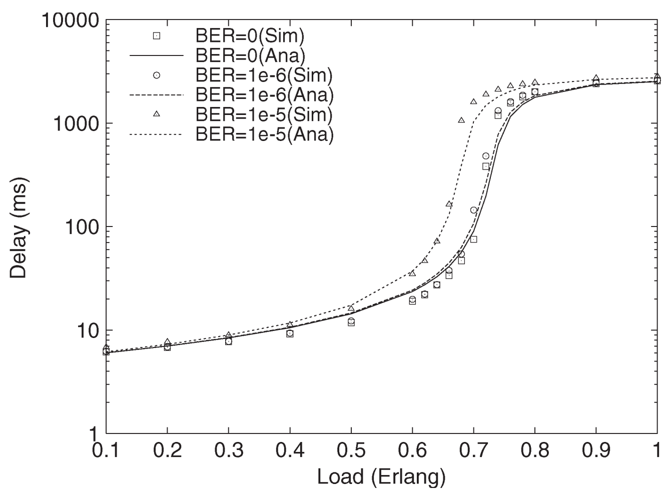

Fig. 8. Average delay versus loads ($N = 10$ and $R_d = 2$ Mb/s).

Finally, in Fig. 8, we show the delay performance versus traffic loads under different BER conditions, where we apply the same setting as that in 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 DCF, which is the fundamental 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 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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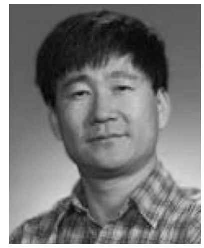

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