A Simple Performance Model of IEEE 802.11 WLAN with Arbitrary Buffer Size and Traffic Load

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***Abstract*—In literature, most existing performance models of the IEEE 802.11 networks assume a saturated traffic load, where every station always has a frame to send. Some other models capture a specific case of an unsaturated traffic load, where each station can buffer at most one frame at a time. There are, however, few works that model the performance for an arbitrary traffic load and buffer size using a three-dimensional Markov chain. But those models are very complex due to the huge number of transitions between the states with different queue sizes. In this work, we propose a new and much simpler performance model for arbitrary traffic loads and buffer sizes. The model is based on a very simple but valid assumption, which decreases the number of transitions in a three-dimensional chain, making it more readable and easier to calculate. The performance evaluations show that the proposed model has better accuracy than other well-known models under both *unsaturated* and *saturated* traffic loads.**

***Index Terms*—802.11, 802.11a, 802.11ac, 802.11ax, 802.11b, 802.11g, 802.11n, WLAN, performance model, throughput, Markov chain.**

I. Introduction

TODAY, it is difficult to imagine our daily lives without IEEE 802.11 wireless local area networks (WLANs), also known as Wi-Fi. Since they are very easy to deploy and use, we widely use them in homes, public places, and offices for broadband access to the Internet and to other local services, such as shared printers or file servers.

Early WLAN technologies such as IEEE 802.11a/b/g use the distributed coordination function (DCF) protocol that is based on carrier-sense multiple access with collision avoidance (CSMA/CA) [1]–[3]. Later technologies such as IEEE 802.11n/ac were improved by higher order modulation and coding rates, multiple input/multiple output (MIMO), frame aggregation, and block acknowledgment mechanisms [4]–[9]. The most recent, IEEE 802.11ax, introduced orthogonal frequency division multiple access (OFDMA) [10]. But the medium access mechanism of modern 802.11n/ac/ax technologies still uses the CSMA/CA-based DCF protocol with some changes. Therefore, the correct performance modeling of the basic DCF protocol is the key to obtaining more accurate performance models of later technologies such as IEEE 802.11n/ac/ax. However, the existing performance models of the DCF protocol have the following major drawbacks:

- Most existing models consider saturated traffic only [3], [11]–[15], where the stations always have at least one frame to send. Such models are a good tool to analyze saturation throughput, but in real-world applications, the stations have unsaturated traffic loads most of the time.
- Some other works model a very specific case of unsaturated traffic assuming a *bufferless station*, i.e., a station can have at most one frame at a time [16]–[19]. This is an impractical assumption, since 802.11 stations usually have a queue at least for a few dozens of frames. Thus, these models underestimate the probability of collision and overestimate the system's performance.
- However, there are some works that model a more general case where stations have an arbitrary load and buffer size [20]–[23]. These models are based on a three-dimensional (3-D) Markov chain and produce better accuracy. But they are overly complex due to the enormous number of transitions between different states and the lack of concise equations for steady-state probabilities.

In this work, we propose a new and much simpler performance model of IEEE 802.11 WLANs for a general case where stations have an arbitrary traffic load and buffer size. Our model is also based on a 3-D Markov chain, but unlike other 3-D models, it is much simpler and easier to calculate. We make a very simple assumption, such that *new frame arrivals are taken into account immediately after the current head-of-line frame is served*. This assumption does not affect the normal operation of the protocol but greatly decreases the number of transitions between different states. Performance evaluations show that the proposed model has better accuracy than other well-known models in both the *unsaturated* and *saturated* regions of the offered traffic load.

II. Background and Related Work

A. The distributed coordination function (DCF) protocol

DCF is a medium access control (MAC) protocol, a fundamental component of the IEEE 802.11 standard. It is based on CSMA/CA, which means that stations listen to the channel before transmitting to see if it is busy. If the channel is busy, the station waits a random amount of time and then transmits.

The sender station, which has a frame to send, continuously monitors the channel. If the channel is idle for a period called DCF inter-frame space (DIFS), it randomly extracts a new backoff counter from the $[0, W_0)$ range, where W_0 is the size of the contention window at the backoff stage 0. The sender decrements the counter at the end of every idle backoff slot and freezes the counter if the channel is detected to be busy. The backoff counter decrementing is resumed only after an idle DIFS period and an idle backoff slot.

When the backoff counter expires, that is, becomes 0, the sender transmits its frame. If the receiver successfully receives the frame, it replies with an acknowledgement (ACK) frame after a period called short inter-frame space (SIFS). If the sender receives the ACK within a timeout, it resets its backoff stage to 0. Otherwise, the sender concludes that its frame was not received due to a collision and therefore increments its backoff stage.

When the backoff stage is incremented, the sender doubles the size of its contention window to decrease the probability of the collision, that is, $W_i = \min(2^i \times W_0, W_{max})$ where $i \in [0, r]$. After an idle DIFS period, the sender randomly extracts a new backoff counter from the $[0, W_1)$ range and retransmits the frame after the counter expires. However, if the ACK is not received even after transmission from the last backoff stage r, the data frame is eventually dropped, and the sender resets its backoff stage [1]–[3], [6].

B. Existing performance models

Performance modeling of IEEE 802.11 WLANs has long been a hot research topic. Existing performance models in the literature can be divided into three categories. The models in the first category assume that stations always have saturated transmit buffers and thus always compete for channel access. First, in [3], Bianchi proposed a new two-dimensional (2-D) Markov chain model of backoff stage and backoff counter evolution with the following assumptions: all stations have saturated buffers, each transmission encounters the collision with constant probability regardless of the backoff stage value, ideal channel conditions, and infinite retransmission attempts.

In the following years, many papers improved various aspects of Bianchi's model by relaxing different assumptions. For example, in [11], the authors improved Bianchi's model by making the retransmission attempts finite. Vishnevsky et al. improved Bianchi's model by accounting for channel errors [12]. In [13], Raptis et al. extended previous models to obtain a more accurate distribution of access delay under saturated traffic conditions. In [14], [15], the authors improved the original Bianchi model, taking into account the more correct rule for decrementing the backoff counter. Models in the first category can be a handy tool to analyze the upper limits of saturation throughput, but in real-world applications, stations have unsaturated traffic loads most of the time.

The works in the second category relax Bianchi's saturated buffer assumption. They model a very specific case of unsaturated traffic assuming a *bufferless station*, i.e., a station can have at most one frame at a time [16]–[19]. With such an assumption, the authors effectively avoid introducing a third dimension (for queue size evolution) in their Markov chains. However, this is an impractical assumption since 802.11 stations usually have a buffer at least for a few dozens of frames, and, at any given time, queues may have multiple pending frames. Moreover, due to such an assumption, these models underestimate the probability of collision, leading to overestimated throughput [20].

The works in the third category extend the previous models to a general unsaturated traffic case [20]–[23]. For example, in [20], Liu et al. introduce a third dimension in the Markov chain to track changes in queue length. In [21], Sutton et al. extend Liu's model to include channel errors and the capture effect. In [22], [23], Martorell et al. extended Liu's model by accounting for a more correct rule for decrementing the backoff counter. Using these models, one can calculate system throughput, collision probability, and access delay with greater precision. However, the biggest drawback of these models is their extremely high complexity. They have too many possible states and a huge number of transitions between the states with different queue sizes, which makes it difficult to formulate the steady-state probabilities in terms of concise equations.

III. System Model

As in [3], [11], [13]–[17], [20], we also assume that every station is within range of each other, the channel is ideal, and thus the receivers fail to decode the frames due only to collisions. However, we also introduce a new assumption, such that *new frame arrivals are taken into account immediately after the service of the current head-of-line (HOL) frame finishes*. This assumption helps to simplify the complexity problems that exist in conventional 3-D Markov chain modeling approaches such as [20]–[23]. In these models, new frame arrivals are accounted for at the end of each slot, i.e., the queue size can change after each slot, resulting in an enormous number of transitions between the states of the different queue sizes [9]. This approach complicates the performance model, making it less understandable and difficult to estimate.

Generally, when a frame becomes HOL, the station immediately starts the backoff process, i.e., starts the service of this frame. The service ends when the station either receives an acknowledgement (ACK) for this frame or drops it due to an exhausted retry limit. According to our new assumption, the queue size does not change during the service. This does not mean that new frames cannot arrive during this period, but new frame arrivals are accounted for right after the service finishes. That is, the queue size can change only at the end of slots in which the station either successfully delivers the frame or drops it due to an exhausted retry limit. This approach greatly decreases the number of transitions and simplifies the 3-D chain, making it easier to understand and estimate.

A. New 3-D Markov Chain Model

As in [3] and other related works, our model is based on a discrete and integer timescale, where time indexes t and $t + 1$ correspond to the beginning of two consecutive slots.

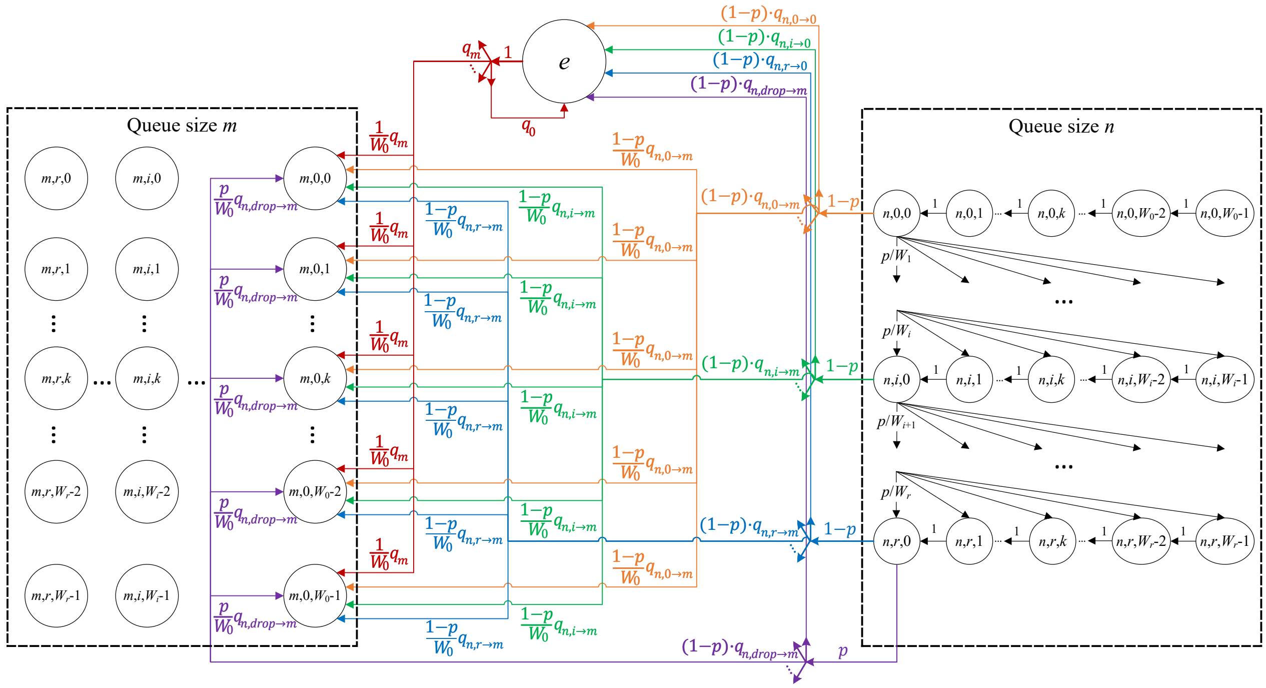

Fig. 1. Proposed Markov-chain of queue size, backoff stage, and backoff counter.

Fig. 1 depicts the newly proposed 3-D Markov chain, which is composed of stochastic processes of the queue size ($q(t)$), the backoff stage ($s(t)$), and the backoff counter ($c(t)$). A generic state in the chain is represented by $\{n, i, k\}$, where n, i, and k are random variables representing the current queue size, the backoff stage, and the backoff counter, respectively. There is also a special state $\{e\}$, representing the case where the station does not have any frames in the queue, that is, the *empty state*. Due to the lack of space and for better readability, Fig. 1 shows only the states of queue size n and m, and the empty state e. Furthermore, the figure depicts only (i) incoming transitions to states with queue size m from states with queue size n and the state $\{e\}$ and (ii) incoming transitions to the state $\{e\}$ from the states with queue size n. We believe that readers can get any transition from/to states with any queue size. Note that W_i is the size of the contention window at backoff stage i, and r is the retry limit.

B. One-step state transition probabilities

Let us start deriving the probabilities of incoming one-step transitions into state $\{e\}$ that are given in (1).

$$\begin{cases} P\{e|e\} = q_0 \\ P\{e|n,i,0\} = (1-p)\cdot q_{n,i\to 0}, \quad i\in[0,r]\ . \\ P\{e|n,r,0\} = p\cdot q_{n,drop\to 0} \end{cases} \qquad (1)$$

The first expression is the probability of the incoming transition from $\{e\}$ itself, where q_n is the probability of n new frame arrivals during the average slot duration denoted by E_s and will be discussed later in more detail. The second expression is the probability that the station successfully transmits the frame from state $\{n, i, 0\}$ and then finds no frame in the queue, where p is a conditional collision probability introduced in [3] and $q_{n,i\to m}$ is a probability that the queue has m frames after a successful transmission in the state $\{n, i, 0\}$. Finally, the last expression is the probability that the retransmitted frame from state $\{n, r, 0\}$ encounters a collision and there is no frame in the queue, where $q_{n,drop\to m}$ is a probability that the station has m frames in its queue after dropping/discarding the HOL frame due to the exhausted retry limit.

Transition to stage 0 is triggered whenever the queue is busy right after the service of the HOL frame finishes, and then the backoff counter is selected randomly from $[0, W_0)$. The transition probabilities to the state $\{m, 0, k\}$ are given in (2), where $k \in [0, W_0)$, $n, m \in [1, Q]$, and Q is the queue/buffer size limit (in frames). The first equation is the probability that m frames arrive into an empty queue during E_s. The second is the probability that the station has m frames in its queue immediately after successfully transmitting a frame. Finally, the last is the probability that the station has m frames in its queue right after it drops a frame upon failure at the final retry.

$$\begin{cases} P\{m,0,k|e\} = \frac{q_m}{W_0} \\ P\{m,0,k|n,i,0\} = \frac{1-p}{W_0}\cdot q_{n,i\to m}, \quad i\in[0,r]\ . \\ P\{m,0,k|n,r,0\} = \frac{p}{W_0}\cdot q_{n,drop\to m} \end{cases} \qquad (2)$$

We now examine one-step transition probabilities between the states of the same queue size, given in (3). The first equation represents the probability that the station decrements its nonzero backoff counter at the end of idle slots. The second is the probability that the transmitted frame encounters a collision and, as a result, the sender increments its backoff stage and selects a new backoff counter k.

$$\begin{cases} P\{n,i,k-1|n,i,k\}=1, & i\in[0,r],k\in[1,W_i) \\ P\{n,i,k|n,i-1,0\}=\frac{p}{W_i}, & i\in[1,r],k\in[0,W_i) \end{cases}. \tag{3}$$

C. Steady-state probabilities

Let $\pi_{n,i,k}$ be the steady state probability of state $\{n,i,k\}$. Using (3), we can derive the following relationships for $n\in[1,Q]$, $i\in[0,r]$, and $k\in[0,W_i)$,

$$\begin{cases} \pi_{n,i,k}=\frac{W_i-k}{W_i}\pi_{n,i,0} \\ \pi_{n,i,0}=\pi_{n,0,0}\cdot p^i \end{cases}. \tag{4}$$

After applying (1) to the balance equation of the state $\{e\}$, we can obtain the following expression for the steady-state probability of the state $\{e\}$ denoted by π_e,

$$\pi_e=\frac{1}{1-q_0}\sum_{n=1}^{Q}\pi_{n,0,0}\Big((1-p)\sum_{i=0}^{r}p^i\cdot q_{n,i\to 0} + p^{r+1}\cdot q_{n,drop\to 0}\Big). \tag{5}$$

Using (2) and (4), we can derive the steady-state probability of state $\{n,0,0\}$ as follows

$$\pi_{n,0,0}=\pi_e\cdot q_n+\sum_{m=1}^{Q}\pi_{m,0,0}\Big((1-p)\sum_{i=0}^{r}p^i\cdot q_{m,i\to n} + p^{r+1}\cdot q_{m,drop\to n}\Big). \tag{6}$$

By utilizing that the sum of the all steady-state probabilities is one, the following relationship is derived.

$$\sum_{n=1}^{Q}\pi_{n,0,0}=\frac{2\cdot(1-\pi_e)}{\sum_{i=0}^{r}p^i(W_i+1)}. \tag{7}$$

D. Transmission probability and throughput

A station transmits the HOL frame whenever its backoff counter expires and thus the transmission probability τ is given by

$$\tau=\sum_{n=1}^{Q}\sum_{i=0}^{r}\pi_{n,i,0}=\frac{2\cdot(1-p^{r+1})\cdot(1-\pi_e)}{(1-p)\sum_{i=0}^{r}p^i\cdot(W_i+1)}. \tag{8}$$

As in [3], the conditional collision probability is given by

$$p=1-(1-\tau)^{N-1}, \tag{9}$$

where N is the number of stations in the network.

A generic time slot is idle if none of the stations transmits; it contains a successful transmission if only one of the stations transmits; if two or more stations transmit simultaneously, a time slot contains a collision. Thus, the probability of idle slot is $P_i=(1-\tau)^N$, the probability of successful transmission is $P_s=N\cdot\tau\cdot(1-\tau)^{N-1}$, and the probability of collision is $P_c=1-P_s-P_i$. The duration of idle slot, T_i, simply equals the duration of standard slot duration σ. In the basic access scheme, the durations of slots containing successful transmission and collision can be obtained as $T_s=T_c=T_{frame}+\delta+T_{SIFS}+T_{ACK}+\delta+T_{DIFS}$, where $T_{frame}=T_{preamble}+T_{header}+T_{payload}$ is the time spent while transmitting the data frame, and $T_{preamble}$, T_{header}, T_{SIFS}, T_{ACK}, and T_{DIFS} are the durations of preamble, header, short inter-frame space (SIFS), ACK, and distributed inter-frame space (DIFS), respectively. $T_{payload}$ denotes the transmission time of a single data payload. Then, the normalized system throughput is calculated as the average time for successful payload transmission over the average slot time.

$$S=\frac{P_s\cdot T_{payload}}{P_i\cdot\sigma+P_s\cdot T_s+P_c\cdot T_c}. \tag{10}$$

E. Offered load and service time relationship

To calculate the throughput S, we need τ and p, which require q_m, $q_{n,i\to m}$, $q_{n,drop\to m}$ for all n,m,i.

Let λ denote the frame arrival rate at each station. Assuming Poisson traffic, let us denote the probability of j frame arrivals during the interval θ by $\alpha_j(\theta)=((\lambda\cdot\theta)^j\cdot e^{-\lambda\cdot\theta})/j!$. Then the probability that an empty queue will have m frames during an interval E_s can be obtained as follows

$$q_m=\begin{cases} \alpha_m(E_s), & m\in[0,Q) \\ 1-\sum_{j=0}^{Q-1}q_j, & m=Q \end{cases}. \tag{11}$$

On the other hand, when a frame is transmitted successfully in the state $\{n,i,0\}$ where $n\in[1,Q]$, the average service time of the frame is $T_{n,i}=E_s\sum_{j=0}^{i}(W_j-1)/2+i\cdot T_c+T_s$. Then, the probability that a queue has m frames right after this successful transmission, $q_{n,i\to m}$, is

$$q_{n,i\to m}=\begin{cases} \alpha_m(T_{n,i}), & m\in[n-1,Q) \\ 1-\sum_{k=n-1}^{Q-1}q_{n,i\to k}, & m=Q \\ 0, & \text{otherwise} \end{cases}. \tag{12}$$

Let $T_{n,drop}$ be the average time to drop a frame due to exhausted retry limit, $T_{n,drop}=E_s\sum_{j=0}^{r}(W_j-1)/2+(r+1)\cdot T_c$. Then, the probability $q_{n,drop\to m}$ can be also calculated using (12) but by replacing $T_{n,i}$ with $T_{n,drop}$.

Next, we should get E_s, which is the average slot duration when all other stations except the tagged station compete for channel access. E_s was first used in [13] and is calculated as $E_s=\overline{P_i}\cdot\sigma+\overline{P_s}\cdot T_s+\overline{P_c}\cdot T_c$, where $\overline{P_i}=(1-\tau)^{N-1}$ denotes the probability that none of the remaining $N-1$ stations transmit, $\overline{P_s}=(N-1)\cdot\tau\cdot(1-\tau)^{N-2}$ represents the probability that only one of the $N-1$ stations transmits in the given slot, and $\overline{P_c}=1-\overline{P_s}-\overline{P_i}$ denotes the probability that two or more stations simultaneously transmit in the given slot.

TABLE I
PARAMETER SETTINGS

Preamble duration ($T_{preamble}$)	192 μs
Propagation delay (δ)	2 μs
Idle slot duration (σ)	20 μs
DIFS duration (T_{DIFS})	50 μs
SIFS duration (T_{SIFS})	10 μs
ACK frame duration (T_{ACK})	30 μs
frame payload duration ($T_{payload}$)	745 μs
Header duration (T_{header})	45 μs
Minimum contention window size (W_0)	32
Maximum contention window size (W_{max})	1024
Retry limit (r)	7
Queue/buffer size limit (Q)	10 frames

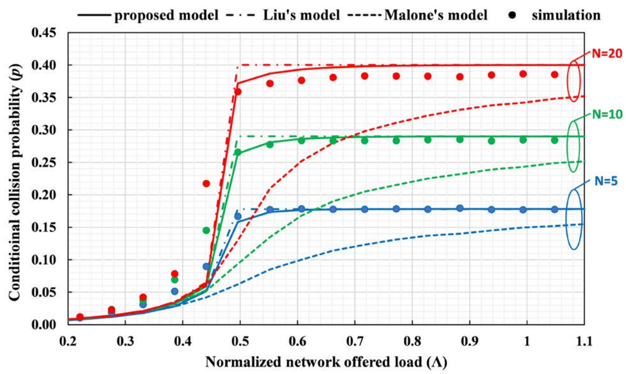

Fig. 2. Conditional collision probability.

Now, by numerically solving the equations from (5) to (9), we can get τ, p, π_e, and $\pi_{n,0,0}$ for $n \in [1, Q]$. Then, we easily calculate the system throughput.

IV. PERFORMANCE EVALUATION

We compare the performance of our newly proposed model with the other well-known models proposed by Malone et al. (further referred to as Malone's model) [17] and Liu et al. (further referred to as Liu's model) [20]. We used the NS-3.40 simulator to validate the results produced by our model. Script files can be found in our GitHub repository [24]. TABLE I includes the simulation and model parameter settings.

A. Discussions on collision probability

Fig. 2 shows the comparison of the conditional collision probability of different models and the measured / empirical collision probability in the simulation for different numbers of nodes (N). The horizontal axis represents the normalized total/network offered load.

We can roughly divide the offered load values into two regions: *unsaturated* and *saturated* offered load regions. In the *unsaturated* region, the station buffers are not always filled with frames. In *saturated* region, however, all buffers are always busy with at least one frame.

The *unsaturated* region in Fig. 2 approximately corresponds to the region $\Lambda \leq 0.65$, where Λ represents the normalized network offered load. At small offered loads, such that $\Lambda \leq 0.4$, the stations have fewer frames to send and thus a smaller probability of transmission (τ). Clearly, the probability of collision (p) is also small due to its relationship with τ given in (9). All three models show similar performance for small offered loads. As Λ increases, τ also increases, and consequently, so does the p. When N increases, p also increases, but the amount of increase is not noticeable for such small loads. Surprisingly, the simulator shows a much higher collision probability compared to the models. We also observed the same situation in [20], [22], [23].

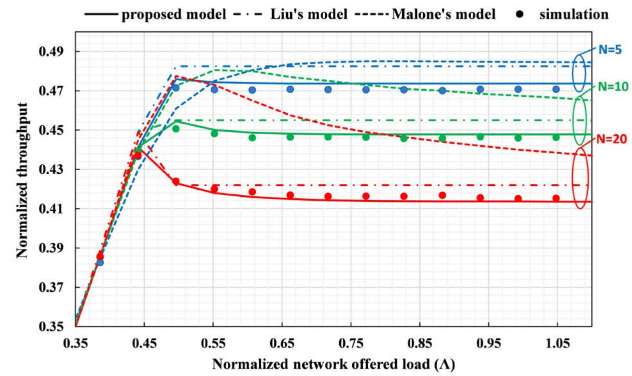

Fig. 3. Normalized system throughput.

As Λ increases beyond 0.40, the probability of collision increases dramatically, indicating that the network is becoming saturated with transmissions. The proposed model and Liu's model correctly reflect this process by producing sharp increases in collision probability. Malone's model, however, shows a much slower increase since it has a *bufferless station* assumption, so buffers can have at most one frame at a time. Such an assumption decreases the actual input traffic load, the number of transmissions, and consequently the probability of collisions. However, in the proposed and Liu's models, the stations can buffer more than one frame and thus produce more realistic collision probabilities for high offered loads. Note that the proposed model shows a more accurate collision probability compared to Liu's model, showing a smooth transition from *unsaturated* region to *saturated* region.

In the *saturated* region, all buffers are always busy with at least one frame; therefore, stations always compete for channel access, and thus the network is fully saturated with transmissions, which corresponds to the traffic condition modeled in [3], [11]–[15] and many others. An additional increase in the offered load does not increase the transmission probability or the collision probability due to (9). Therefore, the proposed and Liu's models produce constant collision probabilities even when the load increases. The empirical collision probability, however, fluctuates, but it does so around some mean value. In contrast, the collision probability produced by Malone's model still continues to increase. It will converge with the other two models and simulation at some very high value of the offered load, which is not included in Fig. 2.

B. Discussions on system throughput

Fig. 3 shows the performance comparison for system throughput. In the *unsaturated* region, where $\Lambda \leq 0.4$, the network can serve all frames well, and thus the network throughput is the same as the offered load in all models and simulator. As Λ increases, the throughput curves start reaching their maximum. The exact value of the offered load, which results in the maximum throughput, depends on the number of nodes and was analyzed in [25]. After reaching maximum, the throughput starts decreasing, and this continues until saturation point, which occurs at $\Lambda \approx 0.65$.

Malone's model produces significantly higher throughput than the simulator and other models. Due to a *bufferless station* assumption, the actual offered load and the probability of collision increase slower; therefore, the throughput also increases slower but eventually achieves a much higher maximum compared to two models and a simulator. Liu's model produces higher maximum and follow-up throughput. We suspect this is due to its additional assumption, which ignores more than one frame arrival during a system slot time. Note that a system slot can be either an empty idle slot, a collision slot, or a success slot. On the contrary, our model predicts the maximum and follow-up decline in throughput with more precision.

In the *saturated* region, the throughput is supposed to stay constant since the number of nodes is fixed and the buffers are (almost) full all the time, so the increase in the offered load will only increase frame drops due to buffer overflow. We can confirm this from the simulation results, where throughput fluctuates around some mean value. Liu's model produces a constant but a little higher throughput than the simulator. Malone's model produces even higher throughput. The proposed model, on the other hand, produces more accurate saturation throughput.

V. Conclusion

Existing 3-D Markov chain-based performance models of the IEEE 802.11 DCF protocol for arbitrary traffic load and buffer size are overly complex due to the huge number of transitions between states of different queue sizes. In this work, we proposed a new 3-D chain-based performance model with the assumption such that *new frame arrivals are taken into account immediately after the current HOL frame is served*. Our 3-D chain is much simpler due to the extremely decreased number of transitions between the states with various queue sizes. Performance evaluation showed that the proposed model has better accuracy than other well-known models in both *unsaturated* and *saturated* regions of the offered traffic load.

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