Performance Analysis of $m$-retry BEB based DCF under Unsaturated Traffic Condition

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Abstract—The IEEE 802.11 standard offers a cheap and promising solution for small scale wireless networks. Due to the self configuring nature, WLANs do not require large scale infrastructure deployment, and are scaleable and easily maintainable which incited its popularity in both literature and industry. In real environment, these networks operate mostly under unsaturated condition. We investigate performance of such a network with $m$-retry limit BEB based DCF. We consider imperfect channel with provision for power capture. Our method employs a Markov model and represents the most common performance measures in terms of network parameters making the model and mathematical analysis useful in network design and planning. We also explore the effects of packet error, network size, initial contention window, and retry limit on overall performance of WLANs.

I. INTRODUCTION

Since its introduction in 1997, WLAN standard IEEE 802.11 received tremendous attention from both researchers and industry. Its widespread commercial adoption attributes to its ad-hoc nature of operation and interoperability. To arbitrate access to the channel, DCF uses a Binary Exponential Backoff (BEB) algorithm which can be either infinite-retry type or $m$-retry type. If the sender keeps trying to send a packet until it succeeds, the backoff algorithm is termed infinite-retry BEB where a packet can suffer repeated failures and hog the channel for long time. $m$-retry BEB drops a packet after $m$ failures giving the next packet a greater chance of reaching destination in time so that delay sensitive applications perform better. In our current work, we analyze $m$-retry BEB based DCF.

A number of works studied performance of WLANs under various conditions. Bianchi [1] used a Markov model analysis to investigate saturation throughput of an infinite-retry limit BEB based DCF assuming ideal channel and ignoring capture effect. Ziouva and Antonakopoulos [2] presented a similar study with modified assumptions on idle channel condition. Bianchi’s model was later extended by Wu et al. [3] to model $m$-retry BEB which is further extended by Chatzimiosis et al. [4] to incorporate the effect of imperfect channel condition.

Although real networks operate under unsaturated load for a considerable amount of time [6], all the above works investigated throughput under saturation condition only. But the study of unsaturated network for both finite and infinite retransmission limit BEB is of great importance in practical terms. Barowski et al. [7] and Daneshgaran et al. [8] calculated throughput under unsaturated load for an infinite-retry limit BEB. Additionally, capture effect is incorporated in [6]. Liaw et al. [8] presented unsaturated throughput for $m$-retry limit but the model did not consider capture effect or imperfect channel. Most commercial 802.11 products support power capture today which brings a
positive impact on overall network performance. But to the best of the authors’ knowledge, no existing work in the current literature presents an analytical model of $m$-retry BEB based DCF considering power capture. Moreover, queueing analysis is mostly ignored which plays a crucial role in determining capacity of delay and loss sensitive applications [9], [10].

No existing study presents a comprehensive model and theoretical performance analyses of 802.11 WLANs considering all four factors concomitantly, namely, (i) unsaturated traffic (ii) finite retry limit, (iii) imperfect channel, and (iv) power capture. In this paper, we propose a simplistic Markov model considering all the above factors and investigate the performance of an unsaturated IEEE 802.11 compliant WLAN in details, both theoretically and through simulation. Consideration of imperfect channel and capture effect makes our model more suitable to reflect real world scenario closely. Our model can accommodate both saturated and unsaturated traffic and is equally applicable for both basic and RTS/CTS type handshake. Common performance measures are expressed in terms of network configuration which makes the model useful in network design and planning. Medium access delay and queuing loss are also estimated which will be very useful in devising techniques to ensure QoS of delay sensitive applications over 802.11 networks.

II. ANALYTICAL MODEL

We develop a Markov model for DCF mechanism employing $m$-retry BEB to analyze performance of a homogeneous WLAN. The model considers effects of imperfect channel and power capture and is applicable to both Basic and RTS/CTS type DCF. It considers unsaturated traffic but can also model saturated traffic, as shown later.

A. The Markov Model

Let $n$ be the number of contending stations, $m$ the retry limit, $m'$ the number of retry stages and $p_F$ the packet error rate in an imperfect channel. Generally, $p_F$ depends on the modulation scheme and device parameters such as transmission rate, noise, and header/data length, etc. The model considers channel error as packet error rate and thus it can be used with any modulation scheme. If $P\{Event\}$ is the probability of Event then we define the following necessary notations and their relations for the most common probabilities which are used in the model.

\[
\tau=\sum_{i=1}^{n-1} i(\tau)^{n-i-1}(1+\tau)^{-i}\]

Capture probability $p_C$ is the probability of a collided frame being received correctly by power capture i.e., a frame with higher power can be received even when more than one frame collided. Assuming an 11 chip Barker sequence being used for code spreading, the spreading factor $s$ is 11. The receiver uses a co-relator to de-spread the original signal. The inverse of processing gain for correlation receiver, denoted by $g$, is given by $g=2(3s)^{-1}$. If power capture in Rayleigh fading channel is enabled, the capture probability for a simultaneous $i$ interfering packets is given by [11],

\[
p_C = \frac{m'}{1-\tau^m}\to \sum_{i=1}^{n-1} i(\tau)^{n-i-1}(1+\tau)^{-i}\]

where $z$ is the capture threshold.

$p_F$ is the probability that a transmission failed, i.e., an ACK is not received and the packet requires a retransmission. This can happen due to a reception with error, a collision without capture, or both. $p_F$ can be defined as,

\[
p_F = p_C + (p_F - p_F)\]

Here $p_F$ is the probability that the queue is non-empty. $p_Q$ is a function of the packet arrival rate $\lambda$ and channel access delay $d_C$. Assuming a $M/M/1$ queue with length $l_Q$ with Markovian arrival and departure, $p_Q$ can be shown to be

\[
p_Q = \frac{\lambda d_C - (\lambda d_C)^{l_Q+1}}{1-\lambda d_C}\]

Here $d_C$ is the time required to serve each packet which is given by the interval between the time when a packet becomes the head of the queue and the time when it is acknowledged and removed from the queue.

The Markov model we used to model DCF with $m$-retry BEB is shown in Fig. 1 The states (ellipses) in a row are in the same retry stage while retry stages increase from top to bottom. The columns denote different counter values increasing from left to right. The state $E$ at the top left corner of the figure represents an empty queue scenario. This state is the key to modeling unsaturated network. When the queue is empty, the system stays in state $E$. Although our model resembles those presented in [5] and [6], it differs in a number of ways. [6] modeled infinite retry BEB where a packet is retransmitted until success. Therefore, the outgoing arcs from state $(m',0)$ with probability $p_F$ are recursive to $m'$-th retry stage and retry stages $m'+1, m'+2, ..., m$ are not present in [6]. Our model being $m$-retry BEB, retry stage is reset irrespective of success and the outgoing arc from state $(m,0)$ neither is recursive, nor depends on success or failure of the last transmission. On the other hand, [5] modeled saturation traffic. Therefore, when the retry stage is reset, the Markov model always goes to the $0$-th retry stage and empty queue state $E$ is not present. In our model, if a transmission is successful or the retry limit
is exceeded, $E$ becomes the new state if the queue is empty. Although [8] investigated unsaturated load for $m$-retry BEB, the effects of imperfect channel and power capture are ignored.

We define the Markov model states as a bi-dimensional process $\{s(t), b(t)\}$ where $s(t)$ is the retry stage and $b(t)$ is the counter value at time $t$. The current backoff window $W_i$ at retry stage $i$ is defined as,

$$W_i = \begin{cases} 2^i W & \text{for } 0 \leq i \leq m', \\ 2^{m'} W & \text{for } i > m'. \end{cases}$$

### B. One Step Transition Probabilities

We denote the one step transition probability from state $(i, k)$ to $(i', k')$ with $P\{i', k' | i, k\}$ are defined as,

$$P\{i, k | i, k+1\} = 1 \text{ for } 0 \leq i < m \text{ and } 1 \leq k \leq W_i - 2,$$

$$P\{i, k | i-1, 0\} = \frac{W_i - k}{W_i} \text{ for } 1 \leq i \leq m \text{ and } 0 \leq k \leq W_i - 1,$$

$$P\{0, k | i, 0\} = \frac{p_{q}(1-p_{f})}{W} \text{ for } 0 \leq i \leq m - 1, 0 \leq k \leq W - 1,$$

$$P\{0, k | m, 0\} = \frac{p_{q}}{W} \text{ for } 0 \leq k < W - 1,$$

$$P\{E | 0\} = (1 - p_{q})(1 - p_{f}) \text{ for } 0 \leq i \leq m - 1,$$

$$P\{0, k | E\} = \frac{1}{W} \text{ for } 0 \leq k < W - 1.$$

### C. Stationary Probabilities of the Markov States

We define $b_{i,k}$ as the stationary probability distribution of any state $(i, k)$ which is given by,

$$b_{i,k} = \lim_{t \to \infty} P\{s(t) = i, c(t) = k\}, i \in [0, m], k \in [0, 2^i W].$$

Let the steady state probability of the system being in state $E$ be denoted by $b_{E}$. As long as the retry limit is not exceeded, the retry stage increments after each failed transmission and $b_{i,0} = b_{i-1,0} p_{f} = p_{b} b_{i,0}$ for $1 \leq i \leq m$.

Due to chain regularity,

$$b_{i,k} = \frac{W_i - k}{W_i} \left( p_{q}(1-p_{f}) \sum_{j=0}^{m-1} b_{j,0} + p_{q} b_{m,0} \right) \text{ for } i = 0,$$

$$b_{i,k} = \frac{p_{b} b_{i-1,0}}{p_{f}} \text{ for } 1 \leq i \leq m.$$

The steady state probability of entering and leaving any state is equal. Imposing this condition on state $E$ we obtain,

$$b_{E} = \frac{1}{p_{q}} \left( (1 - p_{q})(1 - p_{f}) \sum_{i=0}^{m-1} b_{i,0} + (1 - p_{q}) b_{m,0} \right).$$

Imposing the same condition on state $(0, 0)$ gives,

$$p_{b} b_{0,0} = (1 - p_{f}) \sum_{i=0}^{m} b_{i,0} + p_{f} b_{m,0}.$$  \tag{5}

Sum of all states, where a packet is transmitted, gives,

$$\sum_{i=0}^{m-1} b_{i,0} = b_{0,0} \sum_{i=0}^{m-1} p_{b} = \frac{1 - p_{m}}{1 - p_{f}} b_{0,0}. $$  \tag{6}

From (4) & (6), we derive

$$b_{E} = \frac{1 - p_{q}}{p_{q}} b_{0,0}.$$

The sum of probabilities of being in every state should be equal to 1 i.e., $\sum_{i=0}^{m} \sum_{k=0}^{W_i-1} b_{i,k} + b_{E} = 1$ which gives (8). Under saturation condition, a node will always have some packet to send. Therefore, $p_{Q} \to 1$ which gives $b_{0,0}$ for saturation model as shown in (5). A packet is transmitted only from the states $b_{i,0}$ where $i \in [0, m]$. Since $\tau$ denotes the probability of transmission by a random node at a random slot,

$$\tau = \sum_{i=0}^{m} b_{i,0} = b_{0,0} \sum_{i=0}^{m} p_{f}^{i} \frac{1 - p_{m}^{i+1}}{1 - p_{f}} b_{0,0}. $$  \tag{9}

Let $t_{\sigma}$, $t_{s_{IFS}}$, and $t_{d_{IFS}}$ be the length of an idle slot, SIFS, and DIFS period. $t_{\delta}$ is the propagation delay. Similarly, $t_{D}$, $t_{A}$, $t_{R}$, and $t_{C}$ are the transmission time of data frame, ACK frame, RTS frame, and CTS frame, respectively.

A successful transmission is detected by the reception of an ACK packet. We define $t_{S}$ as the time a sender has to wait before receiving an ACK for the last packet and remove it from the queue. Taking into account of all the delay elements, $t_{S}$ can be defined as

$$t_{S} = \begin{cases} \sum_{i=0}^{m} t_{d_{IFS}} + t_{D} + t_{\delta} + t_{s_{IFS}} + t_{A} + t_{\delta} & \text{for Basic,} \\
\sum_{i=0}^{m} t_{d_{IFS}} + t_{R} + t_{s_{IFS}} + t_{C} + t_{s_{IFS}} + t_{A} + t_{\delta} & \text{for RTS/CTS.} \end{cases}$$

When a transmission fails, the sender can detect neither collision nor erroneous reception. Therefore, it has to wait for a time duration $t_{E}$ before taking the transmission to be failed, which can be defined as

$$t_{E} = \begin{cases} t_{d_{IFS}} + t_{D} + t_{\delta} + t_{s_{IFS}} + t_{A} + t_{\delta} & \text{for Basic,} \\
t_{d_{IFS}} + t_{R} + t_{s_{IFS}} + t_{C} + t_{s_{IFS}} + t_{A} + t_{\delta} & \text{for RTS/CTS.} \end{cases}$$

Using $p_{f}$ and $\tau$ defined as in (9) and (10), respectively, we calculate the expected slot length $t_{slot}$ as,

$$t_{slot} = (1 - p_{B}) t_{\sigma} + p_{B} (1 - p_{S}) t_{D} + p_{B} p_{S} t_{E} + p_{B} p_{S} (1 - p_{E}) t_{S}.$$

### D. Performance Measures

Throughput $\psi$ is the number of payload bits that the MAC layer can transmit per second which is given by,

$$\psi = \frac{p_{B} p_{S} (l_{D} + l_{I} + l_{U})}{t_{slot}}.$$  \tag{11}

where $l_{D}$, $l_{I}$, and $l_{U}$ are length of application data, IP header, and UDP/TCP header.

A packet is dropped when it suffers from more than $m$ number of failed transmissions. Therefore, network packet loss $e_{N}$ is the probability that a packet suffers from at least $m+1$ failed transmissions and is given by (8),

$$e_{N} = p_{f}^{m+1}.$$  \tag{8}

If queue length is not very large, queuing loss $e_{Q}$ can play an important role for loss sensitive applications. For the $M/M/1/l_{Q}$ queue described earlier, $e_{Q}$ is given by,

$$e_{Q} = \begin{cases} \frac{(\lambda_{D} C)^{q} - (\lambda_{C})^{q+1}}{1 - (\lambda_{C})^{q+1}} & \text{when } \lambda_{D} C \neq 1, \\
\frac{1}{t_{Q}} & \text{otherwise.} \end{cases}$$

Channel access delay $d_{C}$ is defined to be the length of the period starting when a packet becomes the head of the queue (HoQ) and ending when an ACK frame confirms its successful reception. Delay faced by the dropped packets (because of exceeding retry limit) does not contribute to it. The probability that a packet is successfully transmitted is $1 - e_{N} = 1 - p_{f}^{m+1}$. A packet may face delay at each retry stage and the probability
that it reaches stage $i$ and is not dropped (i.e., it is successful) is given by $\frac{p_F^{m+1}}{1-p_F}$. At retry stage $i$, a packet can face $\frac{W_i}{2}$ number of slots on average. In total, the packet has to wait for $n_{\text{slot}}$ number of slots as HoQ which is given by,

$$n_{\text{slot}} = \sum_{i=0}^{m} \frac{W_{i+1} p_F^{m+1}}{2}\frac{p_F^{m+1}}{1-p_F}.$$

Using the above expression for number of slots, the channel access delay can be estimated as \([3]\).

III. SIMULATION AND RESULTS

We performed all mathematical analyses in Matlab 2007a and modeled the simulations in Network Simulator 2 (NS-2.28) which is widely used by network researchers. We found a very agreeing match, as shown later, between the simulations and our theoretical datasets which validates our model. NS-2’s original tracing mechanism being slow and inaccurate to trace a packet at each network layer, a separate tracing mechanism is developed and used in this work. To carefully study the effect of network load and packet arrival rate, a separate Poisson agent is also developed along with a corresponding AP (Null) agent. These classes were made open source and posted in NS-2 official mail group for comments and public use.

We tested the simulations for IEEE 802.11B with DSSS based physical layer but the method is applicable to any 802.11 variant. The parameters used in the simulation are shown in Table I. Fig. 2(a) shows throughput $\psi$ as a function of packet arrival rate $\lambda$ for different payload sizes. We used a wide range of payload sizes (100~5000 bytes). The simulation results match mathematical data closely. The throughput increases almost linearly with increasing arrival rate up to a certain point, beyond which it does not increase any further. The first region indicates unsaturated condition of the network where $\lambda \ll \frac{1}{d_C}$ while the second region indicates near saturation or saturation region where $\lambda \approx \frac{1}{d_C}$. The smooth transition between the two regions is where network switches into saturation and $\lambda \approx \frac{1}{d_C}$ (e.g., $\lambda \approx 20$~30 packets/sec for $l_D = 500$B). This transition is clearly a function of the payload and it is achieved at a lower $\lambda$ when the payload is larger.

For a larger payload, the transmission time becomes longer, making the transmissions more susceptible to collisions and errors. Thus the channel access time becomes longer saturating the network even with a smaller packet arrival rate. A longer channel access time incurs relatively high degradation in throughput since the channel is kept busy for each packet in the queue. However, the payload contributes even more to the total throughput and a greater payoff still gives a greater throughput although channel access time is longer in this case. This is the reason the rate of increase of throughput in the unsaturated region is higher for longer packets and the same level of throughput can be achieved by longer packets at a lower packet arrival rate. For the rest of this study, we use $l_D = 1000$ bytes as a representative data payload size.

Fig. 2(b) shows change in throughput for different packet error rates. These results show that although the impact of packet error rate is considerably high in saturated region, its impact is negligible in unsaturated region. In simulation, errors are introduced uniformly in the received packets. Since the number of transmissions in the unsaturated region is quite low, less number of retries is necessary and hence the impact is found insignificant. On the other hand, in the saturated region, a large number of transmissions and retransmissions take place; as a result the impact of erroneous packets on

![Figure 2(a)](image1)

![Figure 2(b)](image2)

![Figure 2(c)](image3)

**TABLE I: Simulation parameters**

| Parameter       | Value  | Parameter       | Value  |
|-----------------|--------|-----------------|--------|
| SIFS            | 10µs   | Idle slot       | 20µs   |
| DIFS            | 50µs   | Propagation delay | 1µs   |
| PLCP header     | 144b   | Preamble        | 48b    |
| Data rate       | 1 Mbps | Basic rate      | 1 Mbps |
| PLCP rate       | 1 Mbps | Run length      | 2000ns |

Fig. 2: Simulation and analytical throughput for different (a) payload, (b) packet error rate, and (c) number of stations.
saturation even with a much lower packet arrival rate. While ELSE being unchanged, a higher number of nodes achieved results demonstrated the same phenomenon where, everything in a higher channel access time and a lower throughput. The bytes, the saturation throughput is same in all cases which is lead to a greater number of collisions, which in turn will result Therefore, even if the number of generated packets in unit number of generated packets depends on them. The number of nodes also means the number of competitors for the channel. Therefore, even if the number of generated packets in unit time (and payload) is same, a greater number of nodes may lead to a greater number of collisions, which in turn will result in a higher channel access time and a lower throughput. The results demonstrated the same phenomenon where, everything else being unchanged, a higher number of nodes achieved saturation even with a much lower packet arrival rate. While a 4-node network did not achieve saturation until \( \lambda=60 \), a 15-node network is saturated even at \( \lambda=9 \). Therefore, network size plays an important role in attaining saturation level of a network, its impact being much higher than even packet arrival rate. Since our model matches very closely with simulation results (as shown above), in the rest of this paper we present analysis based on theoretical data derived from the model.

Fig. 4 presents results for different numbers of stations in the network. Since the payload is kept constant to 1000 bytes, the saturation throughput is same in all cases which is 0.9 Mbps. Below the saturation point, i.e., in the unsaturated region, the rate of increase in throughput is different for different network sizes. Both the packet arrival rate and the number of nodes define the load on the network since the number of generated packets depends on them. The number of nodes also means the number of competitors for the channel. Therefore, even if the number of generated packets in unit time (and payload) is same, a greater number of nodes may lead to a greater number of collisions, which in turn will result in a higher channel access time and a lower throughput. The results demonstrated the same phenomenon where, everything else being unchanged, a higher number of nodes achieved saturation even with a much lower packet arrival rate. While a 4-node network did not achieve saturation until \( \lambda=60 \), a 15-node network is saturated even at \( \lambda=9 \). Therefore, network size plays an important role in attaining saturation level of a network, its impact being much higher than even packet arrival rate. Since our model matches very closely with simulation results (as shown above), in the rest of this paper we present analysis based on theoretical data derived from the model.

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most common values of \( W \) are 16 and 32, a contention window range of \([1\ldots100]\) is investigated to demonstrate its effect. A low \( W \) reduces backoff time which in turn increases collisions and decreases throughput. However, decrease in the idle period reduces overall channel access time. Although number of stations and packet arrival rate are kept constant, \( d_C \) is initially low but grows quickly until \( W=20 \), e.g., \( d_C=0.054s \) and 0.065s at \( W=10 \) and 20 for \( p_E=0 \). Increase in channel access delay becomes minimal for \( W>70 \). Throughput shows a similar trend but apparently the effect of channel error on throughput is more aggressive compared to that of \( W \).

Fig. 3(b) shows throughput and channel access delay for a range of retry limits in presence of channel errors. Delay is found to be lower for a lower \( m \). When \( m=0 \), a packet will be dropped even after the first failure in its transmission. With higher \( m \), a higher number of failed transmissions attempts are tolerated. Therefore, the HoQ packet can hog the channel for a longer period and channel access delay becomes higher. On the contrary to infinite-limit BEB, where delay upper limit is high, \( m \)-retry BEB keeps the channel access delay within a defined margin. In delay sensitive applications (e.g., VoIP, video conference, etc.), a late packet is as good as a dropped packet. Playing a late packet will only increase jitter at the receiver end. Therefore, it is better to drop a late packet when it can help other packets to reach in time. For the parameters shown in the figure, \( d_C \) is found to be 66.9ms for \( m=0 \), which reaches 67.2ms for \( m=3 \) and remains unchanged for
higher values of \( m \). On the other hand, throughput is higher for lower \( m \) (e.g., 0.6279 mbps for \( m=1 \)), drops exponentially with increasing \( m \), and becomes nearly constant for \( m \geq 4 \).

Fig. 4(a) elaborates throughput and channel access delay as a function of the number of stations for two packet arrival rates. With \( \lambda=2 \), throughput is 260.027 kbps at \( n=1 \) which rises very quickly to 892.397 kbps at \( n=5 \) and remains unchanged for higher number of nodes. We omit discussion on higher number of stations since the network would then operate in saturated region while we mainly investigate the unsaturated condition in this paper. Interested readers may consult [5] and the references therein for more discussions on the saturated condition. With \( \lambda=6 \) the throughput is much higher initially (699.483 kbps at \( n=1 \)) and quickly reaches 891.681 kbps at \( n=3 \). As \( n \) increases, throughput remains the same irrespective of \( \lambda \). For \( \lambda=2 \), channel access delay is initially very low \((d_C=462\mu s)\) at \( n=1 \) which increases slowly at the beginning until \( n=3 \) \((d_C=200\mu s)\). Apparently, the offered network is too small to provide sufficient contention. After that, it increases almost linearly with \( n \).

A steady state system is defined by transmission probability and collision probability. In Fig. 4(a)-(c) collision probability \( p_C \), transmission probability \( \tau \), non-empty queue probability \( p_Q \), network loss \( e_N \), and queue loss \( e_Q \) are shown for varying packet arrival rate, initial collision window, and retry limit. With increase in packet arrival rate, as shown in Fig. 4(a), the probability that the queue is non-empty rises very quickly and becomes \( Q \) for \( \lambda>20 \). With the queue always non-empty, further increase in packet arrival rate increases the number of packets in the queue. When a newly generated packet finds the queue full, it will be dropped. Therefore, soon queue loss will overwhelm the system and loss becomes very high. That is why queuing loss \( e_Q \) starts growing very quickly from \( \lambda=20 \). But notably transmission probability \( \tau \) remains unchanged after some initial increase up to \( \lambda=20 \). Since the number of stations is constant, increase in \( \lambda \) does not have additional impact once the nodes become nearly saturated.

Fig. 4(b) on the other hand, elaborates the same performance measures for different values of contention window \( W \). A higher contention window gives greater spread of counter values over contenders ensuring less collision. At the same time, nodes will wait longer in idle state and channel access time increases. We find transmission probability to decrease with increasing collision window. Probability of a non-empty queue \( p_Q \) rises quickly and becomes \( 1 \) as before but queuing loss is much smaller in this case. Since the packet arrival rate and the number of stations are constant, queuing loss \( e_Q \) does not increase after reaching its highest value of 0.2931. Transmission probability, collision probability, and network loss decrease initially with increase in \( W \) since nodes tend to spend more time idly and frequency of transmissions become lower.

Fig. 4(c) presents the above mentioned parameters for variation of retry limit. At a lower \( m \), a packet will be dropped after a smaller number of failures. As a result, number of transmissions will be less and network loss will be higher. However, it also means that channel access delay will have a lower upper-limit. We find that transmission probability decreases initially and remains constant when the network configuration can transmit the offered load. Collision probability and queuing loss follow a similar pattern of different magnitudes. Since network loss is given by \( p_{m+1} \), \( e_N \) continues to decrease exponentially with increasing \( m \) in the constant \( p_C \) region. Probability of the queue being non-empty is 1 when packet arrival rate is greater than or equal to servicing rate of the queue i.e., \( \lambda \geq \frac{1}{d_C} \) or \( \lambda d_C \geq 1 \). Channel access delay decreases with increase in \( m \) as discussed. In particular, \( d_C=0.209 \) at \( m=1 \) and \( d_C=0.0707 \) at \( m=10 \). Therefore, albeit \( \lambda \) is kept constant, \( \lambda d_C \geq 1 \) for the whole region and \( p_Q=1 \) for this configuration.

IV. Conclusion

This paper introduces a Markov model analysis for IEEE 802.11 m-retry BEB based DCF under unsaturated condition. We investigated the most widely used network performance measures in an imperfect channel with provision for power capture. The performance measures are presented in terms of network parameters and will be of great assistance in designing and assessing IEEE 802.11 based wireless networks. WLANs tend to suffer from severe bottlenecks formed around the AP under high load which can cause call drop, call rejection [9] for voice/video calls, and degradation of VoIP voice quality. Therefore, these networks should be carefully designed considering expected load and network parameters. This model considers both network load and configuration, and is equally applicable for both saturated and unsaturated studies and, therefore, can play a significant role in network planning.

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