Performance analysis of Collision Alleviating DCF Protocol in congested wireless networks - A Markov chain analysis

<table>
<thead>
<tr>
<th>Journal:</th>
<th><em>IET Networks</em></th>
</tr>
</thead>
<tbody>
<tr>
<td>Manuscript ID:</td>
<td>NET-2012-0187.R2</td>
</tr>
<tr>
<td>Manuscript Type:</td>
<td>Research Paper</td>
</tr>
<tr>
<td>Date Submitted by the Author:</td>
<td>18-Feb-2013</td>
</tr>
<tr>
<td>Complete List of Authors:</td>
<td>Tatineni, Madhavi G, Sasi Bhushana Rao MNVSS, Kumar</td>
</tr>
<tr>
<td>Keyword:</td>
<td>NETWORK PROTOCOLS, WLAN, WIFI</td>
</tr>
</tbody>
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Performance analysis of Collision Alleviating DCF Protocol in Congested Wireless Networks - A Markov chain analysis

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Abstract

IEEE 802.11 is the most popular Wireless Local Area Network (WLAN) standard in use. WLANs support broadband multimedia communication and hence providing Quality of Service (QoS) requirements like good throughput and minimum end-to-end delay are the two main challenging issues in designing of WLAN protocols for supporting real time applications. Until now, several Markov chain models have been developed to evaluate and to enhance the performance of the IEEE 802.11 Distributed Coordination Function (DCF) protocol. However, these models cannot accurately predict the performance of the network. Also, the existing models suffer with high packet collisions resulting in degradation of throughput and end-to-end delay particularly under congested environments. This paper proposes an exact Markov chain model to accurately predict the performance of the wireless networks. To alleviate the collisions and to avoid channel capture effect, we introduce a post backoff stage to provide Inter Packet Backoff (IPB) delay between successive packet transmissions. The analysis is carried out by considering the non-saturated traffic and the impact of channel errors due to Rayleigh fading. Results show significant improvement in throughput and reduction in delay using the proposed model when compared to the existing models.
Keywords – IEEE 802.11, DCF, BEB, Throughput, End-to-End delay

I Introduction

In all IEEE 802.11 Wireless Local Area Networks (WLANs), the basic Medium Access Control (MAC) technique used is distributed coordination function (DCF). DCF employs Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) scheme using binary exponential backoff (BEB) algorithm. DCF describes two techniques for packet transmission: the two-way handshaking technique called basic access mechanism and an optional four way handshaking technique known as request-to-send/clear-to-send (RTS/CTS) mechanism. Readers can refer [1] for data transmission procedures using these mechanisms. The IEEE 802.11 standard also uses an optional channel access method called point coordination function (PCF) that can be implemented in an infrastructure network. PCF uses a point coordinator to determine which node currently has the right to transmit [1]. In this paper we concentrate on DCF scheme for analyzing the performance of IEEE 802.11 networks.

Several scientific publications have analyzed the performance of IEEE 802.11 DCF by developing the two-state Markov chain models. However many of these models are based on simplifying assumptions and are not suitable for real time environments. Giuseppe Bianchi developed a simple analytical model to compute the IEEE 802.11 DCF saturation throughput assuming ideal channel conditions and finite number of terminals [2-3]. The model developed by Bianchi is the basis for further research that is going on for years. In [4], authors follow the same Markov chain model developed by Bianchi and considers frame retry limits to avoid overestimation of the throughput of 802.11. To increase the accuracy of the results, busy medium conditions are taken into account in [5]. Some of the authors have addressed the finite load
performance of IEEE 802.11 DCF with queuing models [6]. In [7], authors extended the Bianchi’s model to a non-saturated environment but errors due to imperfect channels are not considered. Yong Shyang Liaw et al. added a new idle state to the Bianchi’s model to represent the node with empty queue [8]. In Bianchi’s model, the node remains in the $m^{th}$ backoff stage until the packet gets transmitted successfully; however, the packet is discarded after $m$ backoff stages in [8]. Chatzimisios et al. analyze the end-to-end delay performance using basic and RTS/CTS access mechanisms [9]. The effects of non-ideal wireless channels are introduced in [10]. Yu Zheng et al. considered the impact of channel errors and incoming traffic loads on the performance of IEEE 802.11 DCF [11]. A post backoff stage is introduced and the equation for optimal constant window that maximizes the network throughput in saturation conditions is derived in [13]. In [14], authors proposed a new collision alleviation scheme to improve the performance of vehicular Ad hoc Networks (VANETs) in which the initial size of the contention window is adjusted based on run time estimation of channel status. Daneshgaran et al. presented a Markov model to analyze the throughput considering transmission errors and capture effects over Rayleigh fading channels [15]. Their model is very accurate compared to the previous research when the contention level of the network is high. Some recent papers analyze the performance of DCF in error-prone channel and in non-saturated conditions [17]. Jin Woo Yang et al. [16] and Fu-Yi Hung et al. [18] considered the effect of hidden nodes to evaluate the performance of IEEE 802.11 DCF. David Malone et al. extended the Bianchi’s model to evaluate the throughput in non-saturated conditions considering different traffic arrival rates [20]. Literature survey reveals that there is no analytical model that considers the backoff freezing, packet collision errors and channel error conditions with post backoff stage in one model. In this
paper, a collision alleviating DCF (CAD) protocol is developed to accurately predict and to enhance the performance of the network.

The paper is organized as follows. The description of CAD protocol is given in section II. In section III, we present the Markov chain model for the CAD protocol and derived the equation for transmission probability. Throughput of the proposed model under non-saturated conditions in Rayleigh fading channel is analyzed in section IV and the delay performance is carried out in section V. Finally in section VI, we discuss the results obtained with the proposed model and conclude the paper.

II Description of CAD protocol

On studying several research publications, in this paper, we propose an exact Markov chain model to accurately predict the performance of the IEEE 802.11 DCF protocol. In the original DCF protocol, when a node has a packet for transmission, it follows the procedure according to basic access or RTS/CTS mechanism and transmits that packet. This procedure has mainly two drawbacks. Firstly, after successful packet transmission, the node again senses the channel for a period of distributed inter frame space (DIFS) to transmit the next packet. The other nodes which are in the same coverage area overhear the successful transmission of the current node and try to transmit their own packets by sensing the channel and may find the channel idle. In this case, more than one node transmits the packets and they collide and hence the performance of the network degrades. Secondly, the node which transmits the packet more recently resets its contention window to minimum and may always get the access to the channel which is called channel capture effect. To overcome these problems there is a need to modify the original DCF protocol. In this paper, to alleviate the collisions and to avoid channel capture a
post backoff stage is introduced to provide inter packet backoff (IPB) delay between successive transmissions. The modified protocol is termed as CAD protocol and its performance is evaluated in terms of throughput and end-to-end delay. The CAD protocol also uses the CSMA/CA scheme with BEB algorithm similar to the legacy DCF and compatible with the existing protocol. The BEB algorithm is well explained in [2]. A two dimensional Markov chain model is used to analyze the performance of CAD protocol.

In real networks, traffic is mostly unsaturated, so this model is developed for practical network operations. The post-backoff stage is used to provide IPB delay i.e. each packet must be transmitted only after the post-backoff interval. Otherwise the node which has been transmitted the packet more recently and the new contending nodes only will get the access to the channel and other nodes may not get the immediate access. This leads to more collisions and channel capture effect. The parameters such as backoff freezing, packet collision errors and channel error conditions are taken into account for the proposed CAD protocol.

### A. Modifications made to original DCF

(i) To reduce the contention among nodes, after the successful transmission of a packet at any backoff stage, the node waits for a random backoff interval to access the channel again. The advantage of this is to avoid the channel capture. Under saturated conditions, the node selects this interval between \((0, W_0-1)\) at post backoff stage where \(W_0\) is the initial contention window (\(CW\)) size.

(ii) In unsaturated conditions, where the packet arrival follows the Poisson’s process, the node waits in idle (\(-1,0\)) state until the next packet arrives in its queue.

(iii) The packet can be transmitted at any backoff stage when its backoff counter is zero. When the packet arrives at node’s buffer and when the channel is idle, it
goes from state (-1,0) to state (0,0) and transmits the packet. When the channel is busy, it selects the CW between (0, W₀⁻¹) at (0,0) state.

III Development of Markov chain model for CAD protocol

Consider the number of contending nodes as fixed, defined as \( n \). Let \( b(t) \) and \( s(t) \) are the stochastic process representing the backoff counter and the backoff stage \((0, \ldots, m)\) respectively for a given node at slot time \( t \), where \( m \) is the maximum backoff stage. In order to consider the non-saturated traffic, we define \( q \) as the probability of having at least one packet in the node’s buffer to transmit. Similar to [3], the key approximation in the proposed model is that, at each transmission attempt, and regardless of the number of retransmissions suffered, each packet collides with constant and independent probability \( P_{\text{col}} \). Also it is assumed that transmission errors due to the imperfect channel can occur with probability \( P_e \) and the channel is busy with probability \( P_b \). The collision and transmission error probabilities are assumed to be statistically independent [15]. Here, the state of each node is described by \( \{i, k\} \), where \( i \) indicates the backoff stage \((0, \ldots, m)\) and \( k \) indicates the backoff delay. The backoff delay takes the values \((0,1,\ldots,W_i-1)\) where \( W_i = 2^i \text{CW}_{\text{min}} \). \( W, W_0 \) and \( \text{CW}_{\text{min}} \) can be interchangeable. The contention window will be increased either due to packet collisions or due to transmission errors since a node cannot distinguish a packet collision from a transmission error. The probability of successful transmission is therefore equal to \((1 - P_e)(1 - P_{\text{col}})\), from which the equivalent probability of failed transmission \( P_{\text{eq}} \) can be expressed as

\[
P_{\text{eq}} = P_e + P_{\text{col}} - P_e P_{\text{col}}
\]  

(1)

The discrete-time Markov model developed by Ziouva et al. is extended here by introducing a post-backoff stage to provide a delay between successive packet transmissions.
(a) Two state Markov chain model for CAD protocol

The discrete-time Markov chain model that gives unsaturated traffic conditions is shown in the Figure 1. The procedure for packet transmission is similar to BEB algorithm except that the delay is introduced between successive packet transmissions. So, a post-backoff stage \((-1,k)\), \(k \in \{0..W_0-1\}\) is added in the proposed Markov model.

**Backoff state transitions**

1. The backoff counter decrements when the node senses the channel idle.

\[
P\{(i,k) | (i,k+1)\} = 1 - P_b \quad k \in (0,W_i-2), \; i \in (0,m)
\]

(2)

2. The backoff counter freezes when the node senses that the channel is busy.

\[
P\{(i,k) | (i,k)\} = P_b \quad k \in (1,W_i-1), \; i \in (0,m)
\]

(3)

3. After each successful transmission, the node with a packet in queue goes to post-backoff stage.

\[
P\{(-1,k) | (i,0)\} = \frac{(1-P_{eq})k}{W_0} \quad k \in (0,W_0-1)
\]

(4)

4. After unsuccessful transmission at stage \(i\)-1, the node reschedules a backoff delay in the next stage.

\[
P\{(i,k) | (i-1,0)\} = \frac{P_{eq}}{W_i} \quad k \in (0,W_i-1), \; i \in (1,m)
\]

(5)

5. When the transmission is unsuccessful in all the stages, the node reaches the last stage of backoff procedure and remains in that stage until the packet transmission is successful.

\[
P\{(m,k) | (m,0)\} = \frac{P_{eq}}{W_m} \quad k \in (0,W_m-1)
\]

(6)
Post-backoff state transitions

1. After each successful transmission, the node goes to idle state \((-1, 0)\) when the queue is empty and waits in that state until the new packet arrives in the queue.

\[
P\{(−1,0) \mid (i,0)\} = (1 − P_{eq})(1 − q) \quad i \in (0,m)
\]

\[
P\{(−1,0) \mid (−1,0)\} = 1 − q
\]

2. The node with a packet for transmission, goes to \((0, 0)\) state when the channel is free and then transmits the packet.

\[
P\{(0,0) \mid (−1,0)\} = q(1 − P_b)
\]

3. The node with a packet for transmission, selects a backoff stage when the channel is busy.

\[
P\{(0,k) \mid (−1,0)\} = \frac{qP_b}{W_0} \quad k \in (0,W_0−1)
\]

4. In the post-backoff stage, the backoff counter decrements when the node senses the channel idle and freezes when the channel is busy.

\[
P\{(−1,k) \mid (−1,k+1)\} = (1 − P_b) \quad k \in (0,W_0−2)
\]

\[
P\{(−1,k) \mid (−1,k)\} = P_b \quad k \in (1,W_0−1)
\]

The probability that a node occupies a given state \(\{i, k\}\) at any discrete time slot is

\[
b_{i,k} = \lim_{t \to \infty} P\{s(t) = i, b(t) = k\} \quad \text{where } i, k \text{ are integers and } -1 \leq i \leq m, 0 \leq k \leq W_i − 1.
\]

In steady state, the following relations are valid:

\[
b_{i,k} = \frac{W_i − k}{W_i} \cdot \frac{P_{eq}^i}{1 − P_b} \cdot b_{0,0} \quad k \in (1,W_i−1), i \in (1,m−1)
\]
\[ b_{m,0} = \frac{P_{eq}^m}{1 - P_{eq}} b_{0,0} \]  

(12)

\[ b_{m,k} = \frac{W_i - k}{W_i} \frac{P_{eq}^m}{(1 - P_{eq})(1 - P_b)} b_{0,0} \]  

(13)

In the above Markov chain model, after successful transmission at any backoff stage, the node directly enters state \((-1, 0)\) when there are no packets to be transmitted and keeps iterating in that state until the arrival of new packet. The stationary probability to be in state \(b_{-1,0}\) can be evaluated as

\[ b_{-1,0} = \frac{1 - q}{q} b_{0,0} \]  

(14)

After successful transmission at any backoff stage, when the node is ready to transmit the next packet, the node enters state \((-1, k)\) to provide some backoff delay between these packets to avoid channel capture. So the stationary probability to be in state \(b_{-1,k}\) obtained is

\[ b_{-1,k} = q \left( \frac{W_0 - k}{W_0} \right) \frac{1}{1 - P_b} b_{0,0} \quad k \in \{1, W_0 - 1\} \]  

(15)

and the stationary probability that the node to be in state \(b_{0,k}\) is

\[ b_{0,k} = \frac{P_b (1 - q) W_0 - k}{1 - P_b} b_{0,0} \]  

(16)

According to probability conservation relation, total probability is equal to one. Therefore

\[ \sum_{i=-1}^{m} b_{i,0} + \sum_{i=-1}^{m} \sum_{k=1}^{W_i-1} b_{i,k} = 1 \]  

(17)

On solving the above equations, we get the following equation for \(b_{0,0}\).
\[ b_{0,0} = \left\{ \frac{1-q}{q} + \frac{1}{1-P_{eq}} + q \left( \frac{W-1}{2} \right) \frac{1}{1-P_b} + \left( \frac{2^m W - 1}{2} \right) \frac{P_{eq}^m}{(1-P_{eq})(1-P_b)} \right\}^{-1} \]  

(18)

**b) Transmission Probability, \( \tau \)**

Let \( \tau \) be the probability with which a node transmits a packet in a randomly chosen slot time. The node transmits the packet when the backoff counter reaches the value of zero. Then the equation for \( \tau \) becomes:

\[ \tau = \sum_{i=0}^{m} b_{i,0} = \frac{1}{1-P_{eq}} b_{0,0} \]  

(19)

By substituting equation (18) into equation (19), \( \tau \) becomes

\[ \tau = \frac{2q(1-P_b)(1-2P_{eq})}{2(1-P_b)(1-2P_{eq})(1-q)(1-P_{eq}) + q(1-2P_{eq})(q(W-1)(1-P_{eq}) + (2^m W - 1)P_{eq}^m) + \ldots} \]  

(20)

In the above equation, the transmission probability \( \tau \) depends on \( P_{col} \) and \( P_b \). In equation (20), under saturated traffic conditions \( q \rightarrow 1 \) when \( m = 0 \), i.e., when no exponential backoff is considered and assuming the packet transmission errors are only due to collisions and \( P_b = 0 \), \( \tau \) reduces to

\[ \tau = \frac{2}{W + 1} \]  

(21)

This is similar to the equation for the constant backoff window problem which shows that the transmission probability is independent of the collision probability \( P_{col} \).
(c) Frame error probability, $P_e$

The performance analysis of the proposed model is done using the network parameters of IEEE 802.11b protocol since it is easier to compare with the existing models, even though the model suits for any IEEE 802.11 family. The IEEE 802.11b supports data rates of 1, 2, 5.5 and 11 Mbps. In this, the physical layer convergence procedure (PLCP) preamble and header are transmitted using differential binary phase shift keying (DBPSK) modulation at a transmission rate of 1 Mbps. The standard specifies a long preamble and a short preamble formats. The long preamble can be used with low data rate mode of operation (1 Mbps) for better synchronization and short preamble can be used with higher data rate modes of operation for better performance. In this paper, the long preamble format is used for performance analysis. The MAC protocol data unit (MPDU) is transmitted with the rate depending on the modulation used. Due to imperfect channels, the transmitted frame will get corrupted and the frame error probability is affected by the bit error probability and the size of the frame.

The frame error probability, $P_e$ is defined as

$$P_e = 1 - (1 - P_{\text{PHY\_error}})(1 - P_{\text{MAC\_error}})$$

(22)

Where $P_{\text{PHY\_error}}$ is the physical layer (PHY) overhead error probability and $P_{\text{MAC\_error}}$ is the MPDU error probability. The PHY and MAC layer overhead probabilities depend on the bit error probabilities as given below.

$$P_{\text{PHY\_error}} = 1 - (1 - P_{b1})^{24 \times 8}$$

(23)

Where $P_{b1}$ is the bit error probability or probability of error of the PLCP preamble and header (24 bytes) occurs during the transmission of physical layer overhead.

$$P_{\text{MAC\_error}} = 1 - (1 - P_{b2})^{(28 + \text{MSDU}) \times 8}$$

(24)
Where $P_{b2}$ is the bit error probability or probability of error of the MPDU occurs during the transmission of MAC header (28 bytes) and MAC service data unit (MSDU). The equations of $P_{b1}$ and $P_{b2}$ can be found in [12].

IV Throughput Analysis

The performance of the wireless communication network can be evaluated in terms of system throughput, end-to-end delay, probability of collision, etc. The average throughput is the ratio between the total data received and the total delay incurred. In this paper, the throughput analysis is carried out for the proposed CAD protocol and compared with the existing models using basic and RTS/CTS access mechanisms. The analysis is done considering non-saturated traffic conditions and impact of channel errors during packet transmission through Rayleigh fading channel.

(a) Busy and Collision probabilities

Busy and Collision probabilities are important parameters in evaluating the performance of IEEE 802.11 system. The channel is detected busy when at least one node transmits the packet in the considered slot time. For a WLAN of $n$ contending nodes on the channel, each transmits with probability $\tau$, the probability $P_b$ or $P_{tr}$ that the channel is busy is given by

$$P_{tr} = P_b = 1 - (1 - \tau)^n$$  (25)

A packet is transmitted successfully when the packet encounters no collisions and no channel errors are introduced during transmission. For each node, the probability of colliding with other nodes $P_{col}$ is given as

$$P_{col} = 1 - (1 - \tau)^{n-1}$$  (26)
Equations (20), (25) and (26) represent a nonlinear system with the three unknowns \( \tau, P_{col} \) and \( P_b \) which can be solved by numerical methods and has a unique solution. Note that \( P_{col} \in [0,1] \), \( P_b \in [0,1] \) and \( \tau \in [0,1] \).

Figure 2 depicts the variation in collision probability, \( P_{col} \) with respect to the number of nodes. The proposed model is compared against five models which are represented with the last name of the first author. These are Bianchi’s [3], Ziouva’s [5], Chatzimisios’s [9], Daneshgaran’s [15] and Haitao’s [17] models. When the number of nodes increase, more packet collisions occur which result in increase in collision probability. The collision probability obtained by using Haitao’s model and Chatzimisios’s model is same. Similarly the performance of Daneshgaran’s and Bianchi’s models is same in terms of collision probability. As shown in the figure, the proposed model outperforms the existing models particularly when the system contains large number of competing nodes. When the number of nodes is 25, the reduction in collision probability is more than 10% using the proposed model.

**(b) Throughput analysis considering transmission errors under finite load conditions**

When the data passes through the communication channel, it is corrupted by the noise. Unsuccessful transmission occurs when more than one user simultaneously transmit the packets that collides with each other or the data packets may be corrupted at the receiver due to erroneous channels. In both the cases, the acknowledgment (ACK) will not be received by the transmitting node and it reschedules the backoff procedure.

Let the normalized system throughput \( S \), defined as the fraction of time the channel is used to transmit the payload bits successfully. The expected time per slot \( E[S]\) can be calculated by taking the successful transmission slot time with the probability \( P_s(1-P_s) \), unsuccessful transmission slot time due to collision with the probability \( (1-P_s) \), unsuccessful transmission slot
time due to channel errors with probability $P_s T_e$ and idle slot time with probability $(1-P_{tr})$. Now, the equation for $E[S_t]$ can be written as

$$E[S_t] = P_{tr} P_s (1 - P_e) T_{\text{success}} + P_{tr} (1 - P_s) T_{\text{collision}} + (1 - P_{tr}) \cdot \text{slottime} + P_{tr} P_s P_e T_e$$

(27)

The equation for throughput when the channel errors are considered becomes

$$S = \frac{P_{tr} P_s (1 - P_e) E[P]}{P_{tr} P_s (1 - P_e) T_{\text{success}} + P_{tr} (1 - P_s) T_{\text{collision}} + (1 - P_{tr}) \cdot \text{slottime} + P_{tr} P_s P_e T_e}$$

(28)

In the above equation,

- $P_s$ is the probability that a transmission occurring on the channel is successful is given by the probability that exactly one node transmits on the channel, conditioned on the fact that at least one node transmits.

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

(29)

- $\text{Slottime}$ is the idle slot time, $T_{\text{success}}$, $T_{\text{collision}}$ and $T_e$ are the average times that the channel is sensed busy because of a successful transmission, due to collision and due to transmission errors. Here, $T_{\text{collision}}$ and $T_e$ are assumed to be same.

- $E[P]$ is the average packet payload size.

The equations for $T_{\text{success}}$ and $T_{\text{collision}}$ using basic and RTS/CTS access mechanisms are given below:

$$T_{\text{success}}(\text{basic}) = DIFS + T_{\text{PACKET}} + \delta + SIFS + T_{\text{ACK}} + \delta$$

(30)

$$T_{\text{collision}}(\text{basic}) = T_e = T_{\text{PACKET}} + \delta + \text{ACK _ Timeout}$$

(31)

$$T_{\text{success}} = DIFS + T_{\text{RTS}} + \delta + SIFS + T_{\text{CTS}} + \delta + SIFS + T_{\text{PACKET}} + \delta + SIFS + T_{\text{ACK}} + \delta$$

(32)
\[ T_{\text{collision}} = T_e = DIFS + T_{\text{RTS}} + \delta + SIFS + T_{\text{CTS}} + \delta + SIFS \]  \hspace{1cm} (33)

DIFS is the distributed interframe space period, SIFS is the short interframe space period, \( T_{\text{PACKET}} \) is the time taken to transmit the data including PHY and MAC headers, \( T_{\text{RTS}}, T_{\text{CTS}} \) and \( T_{\text{ACK}} \) are the timings required to transmit RTS, CTS and ACK frames respectively and \( \delta \) is the propagation delay. The \( \text{ACK\_Timeout} = SIFS + T_{\text{ACK}} + DIFS \).

In the literature, many researchers have analyzed the performance of the wireless networks under saturated traffic conditions. But the network does not perform best when it is saturated and extensive research has been undertaken to prevent the network from saturation [19]. So the effect of \( q \) must be considered and the analysis is done to an arbitrary input rate, which constitutes a more challenging problem. To analyze the performance of unsaturated wireless networks, a parameter \( \lambda \) is used which represents the rate at which packets arrive at the node’s buffer from the upper layers and measured in packets per second (Pkts/sec). If the traffic arrives in a Poisson distribution with small buffer size, the probability \( q \) can be well approximated as [15].

\[ q = 1 - e^{-\lambda E[S_i]} \]  \hspace{1cm} (34)

A more accurate model can be derived upon considering different values of \( q \) for each backoff state. However, a reasonable solution consists in using a mean probability valid for the whole Markov model derived from \( E[S_i] \). Now, \( E[S_i] \) can be used to calculate the probability \( q \). The probability for \( k \) packet arrivals in a generic time \( T \) is given by

\[ P\{a(T) = k\} = e^{-\lambda T} \frac{(\lambda T)^k}{k!} \]  \hspace{1cm} (35)

From the above equation, the relation of \( E[S_i] \) and \( q \) can be written as

\[ q = 1 - P\{a(E[S_i]) = 0\} = 1 - e^{-\lambda E[S_i]} \]  \hspace{1cm} (36)
The notations used in analytical analysis are listed in Table 1.

<table>
<thead>
<tr>
<th>Notation</th>
<th>Description</th>
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<tbody>
<tr>
<td>$P_{b}, P_{tr}$</td>
<td>Probability that the node at the backoff stage senses the channel busy</td>
</tr>
<tr>
<td>$P_{s}$</td>
<td>Probability of successful transmission</td>
</tr>
<tr>
<td>$P_{col}$</td>
<td>Probability of transmission collision</td>
</tr>
<tr>
<td>$P_{e}$</td>
<td>Probability of transmission failure due to imperfect channel conditions</td>
</tr>
<tr>
<td>$P_{eq}$</td>
<td>Equivalent probability of failed transmission</td>
</tr>
<tr>
<td>$W_0, W$</td>
<td>Initial Contention window size</td>
</tr>
<tr>
<td>$\tau$</td>
<td>The probability that a station transmits a packet in a randomly chosen slot time</td>
</tr>
<tr>
<td>$m$</td>
<td>Maximum backoff stage</td>
</tr>
<tr>
<td>$n$</td>
<td>Number of contending nodes</td>
</tr>
<tr>
<td>$q$</td>
<td>The probability of having at least one packet in the stations buffer to transmit.</td>
</tr>
<tr>
<td>$E[S_t]$</td>
<td>Expected time per slot</td>
</tr>
<tr>
<td>$DIFS$</td>
<td>Distributed interframe space period</td>
</tr>
<tr>
<td>$SIFS$</td>
<td>Short interframe space period</td>
</tr>
<tr>
<td>$\delta$</td>
<td>Propagation delay</td>
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<tr>
<td>$T_{ACK}$, $T_{RTS}$ and $T_{CTS}$</td>
<td>Transmission time of ACK, RTS and CTS frames</td>
</tr>
<tr>
<td>$T_{PACKET}$</td>
<td>Transmission time of data including PHY and MAC headers</td>
</tr>
<tr>
<td>Symbol</td>
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<td>--------</td>
<td>-------------</td>
</tr>
<tr>
<td>$\lambda$</td>
<td>Packet arrival rate</td>
</tr>
<tr>
<td>$E[P]$</td>
<td>Average packet payload size</td>
</tr>
<tr>
<td>$T_{success}$</td>
<td>Average time that the channel is sensed busy because of successful transmission</td>
</tr>
<tr>
<td>$T_{collision}$</td>
<td>Average time that the channel is sensed busy due to collisions</td>
</tr>
<tr>
<td>$T_e$</td>
<td>Average time that the channel is sensed busy due to transmission errors</td>
</tr>
</tbody>
</table>

Table 1 Notations used in analytical analysis

Ziouva et al. [5], Chatzimisios et al [9] and Bianchi [3] analyses the throughput under saturated traffic conditions only. The models proposed by Daneshgaran et al. [15] and Haitao et al. [17] works well for unsaturated traffic conditions. So the performance of the proposed model under non saturated traffic conditions is compared with these models. The throughput as a function of packet rate using basic and RTS/CTS access mechanisms is plotted in Figure 3. Using basic access mechanism, the proposed model improves the throughput by 5.9% and 10.4% compared to Daneshgaran’s and Haitao’s models respectively. As shown in the figure 3(b), the performance of the proposed model is found to have higher throughput even using RTS/CTS mechanism.

Figure 4 evaluates the throughput as a function of $n$ under basic and RTS/CTS access mechanisms. The number of packet collisions increases with increase in number of nodes which results in decrease in throughput. When the number of contending nodes is 50, the proposed CAD protocol gives the throughput of 6.6% and 10.5% higher when compared to Daneshgaran’s and Haitao’s models respectively under basic access scheme. When the number of active nodes
increases, the increase in throughput is even higher. When the number of contending nodes is less than 15, the throughput obtained using the proposed model is somewhat less compared to the reference models. Similar is the case using RTS/CTS mechanism.

V End-to-End delay analysis

In most wired/wireless applications end-to-end delay is also a key metric of network performance. The end-to-end delay refers to the delay for a successfully transmitted packet. This is defined as the time interval from the time the packet is at the head of its MAC queue ready for transmission, until an acknowledgement for this packet is received. Some researchers have calculated the average packet delay considering the retry limits but transmission errors are ignored. Therefore, these models do not predict the frame delay in an accurate way. In this paper, the accurate delay analysis is made considering transmission errors and self loop probability on every backoff state.

A successful transmission may occur at one of the several backoff stages. The equation for average packet delay for successful transmission can be found in [5]. Let $E[N_c]$ be the number of collisions of a frame until its successful reception. This is given by the following equation:

$$E[N_c] = \frac{1}{P_s} - 1$$

(37)

Each node spends some time before accessing the channel under busy channel conditions. This is called backoff time and is represented with $E[BD]$. When the transmitted frame collides, each node has to wait for some time before sensing the channel again. Let this time be $T_o$. Therefore mean frame delay is the sum of average time spent by the node during collisions and during successful transmission and can be calculated as
\[ E[D] = E[N_c] (E[BD] + T_{\text{collision}} + T_0) + (E[BD] + T_{\text{success}}) \]  \hspace{1cm} (38)

\[ T_0 \text{ depends on the access method and equals to} \]

\[ T_0 = \begin{cases} 
SIFS + ACK \text{ _timeout} \\
SIFS + CTS \text{ _timeout} 
\end{cases} \]  \hspace{1cm} (39)

**Calculation of** \( E[BD] \):

The average backoff delay depends on the value of its counter and the duration for which the counter freezes when the node detects transmissions from other nodes. Channel is busy when other nodes are transmitting and also during the time when the node \( A \) is transmitting.

Now, number of times that the node is idle is given by

\[ E[\psi] = \frac{1}{P_b} - 1 \]  \hspace{1cm} (40)

When the counter of a node is at state \( b_{i,k} \), the average number of backoff slots \( E[X] \) required to reach state 0 without taking into account the time the counter is stopped is given by

\[ E[X] = \left[ \sum_{i=0}^{m} \sum_{k=1}^{W_i-1} k b_{i,k} \right] + \frac{W_0 + 1}{2} \]  \hspace{1cm} (41)

The term \( \frac{W_0 + 1}{2} \) is added in the above equation because a post backoff stage is added in the proposed model. Based on equations of \( b_{0,0} \), \( b_{i,k} \) and \( b_{m,k} \), the equation for \( E[X] \) becomes

\[ E[X] = \frac{W_0 + 1}{2} + \frac{b_{0,0}}{6(1-P_b)} \left[ P_b (1-q)(W_0^2 - 1) + \frac{4 P_{eq} W_0^2 (1-P_{eq}) (1-4 P_{eq}) - 3 P_{eq} W_0^2 (1-4 P_{eq})}{(1-4 P_{eq}) (1-P_{eq})} \right] \]  \hspace{1cm} (42)

Let \( E[N_{\text{fon}}] \) be the average number of times that the node detects transmissions from other nodes and is given by

\[ E[N_{\text{fon}}] = \frac{E[X]}{\max(E[\psi],1)} - 1 \]  \hspace{1cm} (43)
Therefore the average time $E[S]$ that the counter stopped while listening to other nodes transmissions is

$$E[S] = E[N_{jou}] \left( p_s T_{success} + (1 - p_s) T_{collision} \right)$$  \hspace{1cm} (44)

Now, the average backoff delay $E[BD]$ is

$$E[BD] = E[X] + E[N_{jou}] \left( p_s T_{success} + (1 - p_s) T_{collision} \right)$$  \hspace{1cm} (46)

By substituting the above equations into the equation of $E[D]$, the mean frame delay can be calculated.

(a) End-to-End delay analysis under saturated load conditions

In saturated load conditions, each node has always a packet for transmission. So all the nodes contend the channel and hence it is easy to find the end-to-end delay. Figure 5 shows the end-to-end delay versus the number of nodes with two different retries. When more number of nodes attempts to access the medium, more collisions occur. This leads to increase in number of retransmissions and hence the packet delay increases. As a result, for the proposed model, when the number of nodes, $n = 5$, the average packet delay is 0.03 sec and it increases to 1.58 sec when $n = 50$. Another reason for the increase in packet delay is that the node has to wait for a long time due to successful transmissions of other nodes. Similarly when collisions occurs, the contention window size get doubled and hence the delay increases with increase in number of retries. When the contending nodes are 25, the packet delay is 1.248 sec when $m = 5$. This is for the Markov chain model without post backoff stage. But in the proposed model the packet delay reduces to 0.7241 sec. in other words the delay is reduced by 41.98%. Similarly when $m = 3$, the delay is reduced by 38.52%. 
The dependency of the mean packet delay on the number of nodes and data rates under saturated conditions is examined in Figure 6 and Figure 7. The delay of both the models decreases as the data rate increases. The reason is that the duration of interframe space timings and slot time are constant for a particular PHY and is independent of the data rate. So, the time spent for successful transmissions and collisions diminishes which results in reduced packet delay.

The post backoff stage introduced in the proposed Markov model reduces the number of collisions and avoids capture effect which in turn results in reduced packet delay compared to the model without post backoff stage. When the data rate is 11 Mbps and the number of nodes are 25, the delay obtained is 0.1747sec in Ziouva’s model and it reduces to 0.1019 Sec (41.67%) when inter packet backoff delay is used. Similar is the case using RTS/CTS mechanism.

(b) End-to-End delay analysis in unsaturated load conditions

A saturated load condition is not valid for real time applications because some of the contending nodes may have empty queues. Nodes with empty queues will not contend the channel. Since the number of contending nodes varies according to their packet arrival rates, it is very difficult to analyze the delay more accurately. The end-to-end delay required for transmitting the packet using the proposed model under basic and RTS/CTS mechanisms in non-saturated conditions is plotted in Figure 8. It is assumed that each node produces packets of 1000 Bytes. The end-to-end delay increases with increase in packet rate and number of nodes. Increase in number of active nodes increases the probability of collision and hence the packet delay becomes longer. The delay performance achievable by the RTS/CTS mechanism is always higher compared to basic access mechanism. This is due to shorter packet collision duration.
The network simulator ns-2 (version 2.29) is used for evaluating the performance of the CAD protocol and each simulation runs for at least 50 simulation seconds. The parameters listed in Table 2 have been used for simulation.

<table>
<thead>
<tr>
<th>Channel bit rate</th>
<th>1 Mbps, 5.5 Mbps and 11 Mbps</th>
</tr>
</thead>
<tbody>
<tr>
<td>PHY header</td>
<td>24 bytes</td>
</tr>
<tr>
<td>MAC header</td>
<td>28 bytes</td>
</tr>
<tr>
<td>RTS</td>
<td>20 bytes + PHY header</td>
</tr>
<tr>
<td>CTS</td>
<td>14 bytes + PHY header</td>
</tr>
<tr>
<td>ACK</td>
<td>14 bytes + PHY header</td>
</tr>
<tr>
<td>DIFS</td>
<td>50 µs</td>
</tr>
<tr>
<td>SIFS</td>
<td>10 µs</td>
</tr>
<tr>
<td>Slot Time</td>
<td>20 µs</td>
</tr>
<tr>
<td>Propagation delay, δ</td>
<td>1 µs</td>
</tr>
<tr>
<td>CW_{min} (slots)</td>
<td>31</td>
</tr>
</tbody>
</table>

Table 2 Simulation parameters

VI Results and Conclusion

In this paper, a new Markov chain model is proposed to enhance the performance of IEEE 802.11 DCF. IPB delay is considered between successive packet transmissions. The expression for transmission probability is derived under non saturated and erroneous channel conditions for the developed Markov model. From the results it is observed that the collision probability in the proposed model is reduced by more than 10% when compared to the existing
models. The throughput analysis is carried out under finite load conditions and erroneous channel conditions. The throughput of the CAD protocol is found to be improved when compared to the existing models.

The expression for end-to-end delay is derived for the developed Markov model and compared with Ziouva’s model [5]. When the contending nodes are 25, the End-to-End delay reduces by 41.98% using the proposed model when $m = 5$ and this reduction is 38.52% when $m=3$. The delay decreases with increase in data rate. Even at higher data rates, the proposed model gives better performance. The delay analysis is also carried out under limited load. It is observed that the end-to-end delay increases with increase in packet arrival rate and number of nodes. Increase in number of active nodes increases the probability of collision and hence the delay becomes longer. Large number of backoff stages improves the system throughput at the expense of higher end-to-end delays.

Finally, it is worth noticing that the procedure for data transmission explained in this paper enhances the QoS parameters (probability of collision, throughput and end-to-end delay) compared to legacy DCF protocol used in wireless local area networks.

References


Figure 1 State transition diagram of the Markov chain model for the proposed CAD protocol
Figure 2 Nodes versus collision probability
Figure 3 Throughput as a function of $\lambda$ when $P_e = 10^{-1}$

(a) Basic access

(b) RTS/CTS
Figure 4 Throughput as a function of nodes when $P_e = 10^{-3}$ and $\lambda = 50$ Pkts/sec

(a) Basic access

(b) RTS/CTS
Figure 5 Nodes versus End-to-End delay for various retries

Figure 6 Nodes versus End-to-End delay for various data rates using basic access mechanism
Figure 7 Nodes versus End-to-End delay for various data rates using RTS/CTS mechanism.

Figure 8 Nodes versus End-to-End delay for different packet arrival rates.