Performance Evaluation of Table Driven and Buffer Adaptive WLANs

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Abstract: - The random access control (MAC) technique of standard WLANs is called the distributed coordination function (DCF) [3]. DCF is a carrier sense multiple access based on collision avoidance (CSMA/CA) scheme with binary slotted exponential backoff. This exponential backoff makes the system more complex and fairness [3]-[13] among the stations is a major concern. This paper shows one possible evolution of WLANs where exponential backoff is not employed. In the new techniques herein users transmit randomly but adapt themselves to traffic conditions, thus improving throughput and delay while guarantying fairness. Users in the first technique are controlled by a table which is derived from traffic measurements (Table Driven). In the second, users transmission activities are function of their buffer contents.

Key-Words: - MAC, Table driven WLANs, Buffer adaptive WLANs

1 Introduction

Wireless local area networks (WLANs) have been widely deployed for the past decade. Their performance has been the subject of intensive research. In [1] improvement of throughput and fairness is shown by optimizing the backoff. [1] uses a measure called the average idle interval which does not consider the number of collisions. In [2], the authors proposed a MAC layer based WLAN technique in which they gave higher priority to access the channels so as to improve the throughput and the channel utilization. [1], [2] discuss the fairness problem of the exponential backoff. [3] Proposes a technique based on collision avoidance and fairness to improve the channel utilization. Few WLAN standards have been adopted e.g. IEEE 802.11 [4] which uses collision avoidance scheme with binary slotted backoff. [4] Expresses how the throughput deteriorates with increasing the number of nodes. [5] Uses an analytic model to study the channel capacity – i.e., maximum throughput – when using the basic access (two-way handshaking) method. [6] Considers three kinds of CSMA/CA protocols, which include Basic, Stop-and-Wait and 4-Way Handshake CSMA/CA, and introduce a theoretical analysis for them. Cali in [7] pointed out that depending on the network configuration, DCF may deliver a much lower throughput compared to the theoretical limit. In [8] a contention based MAC protocol named fast collision resolution is presented. [10] Proposes a model named DCF+ which shows the fairness performance. [11] Presents the performance evaluation of the decentralized nature of communication between nodes in IEEE 802.11, in presence of "hidden" nodes. [13] Shows the performance evaluation of Multihop Ad Hoc WLANs. Thus extensive research has been conducted on WLANs [1]-[13]. Fairness index was only discussed in [1], [2], [7] and [10].

This paper tries to investigate simultaneously the four performance indexes i.e. throughput, delay, delay variance and fairness which are not considered in previous studies [1]-[13]. In the table driven technique we consider both idle periods and number of collisions (Table driven technique) which shows the actual load on the network. In the second, we employ buffer adaptive technique which guaranties fairness and provide smaller delay variance.

2. The IEEE 802.11 MAC Protocol

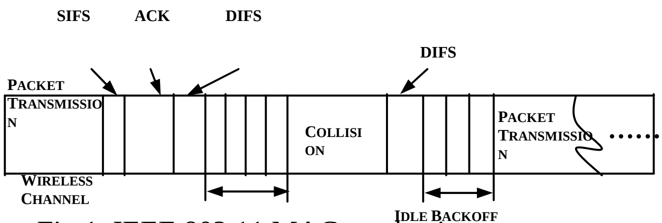

Fig 1. IEEE 802.11 MAC mechanism

Fig. 1 shows one of many transmission scenarios possible with the IEEE 802.11 DCF mode. In this mode a node with a packet to transmit initializes a backoff timer with a random value selected uniformly from the range [0, CW-1], where CW is the contention window in terms of time slots. After a node senses that the channel is idle for an interval called DIFS (DCF interframe space), it begins to decrease the backoff timer by one for each idle time slot observed on the channel. When the channel becomes busy due to other nodes transmission ativities the node freezes its backoff timer until the channel is sensed idle for another DIFS. When the backoff timer reaches zero, the node begins to transmit. If the transmission is successful, the receiver sends back an acknowledgement (ACK) after an interval called the SIFS. Then, the transmitter resets its CW to CW_{min}. In case of collisions the transmitter fails to receive the ACK from its intended receiver within the specified period, it doubles its CW subject to maximum value CW_{max}, chooses a new backoff timer, and starts the above processes again.

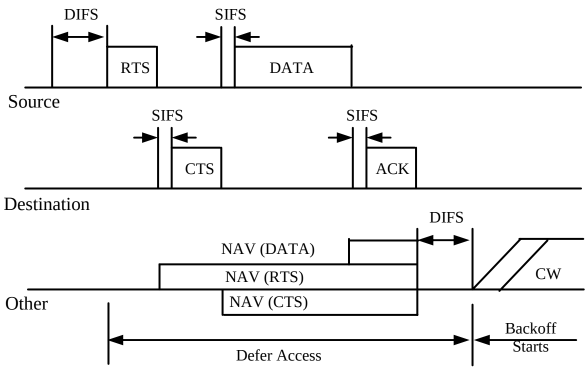

Fig 2. RTS/CTS access mechanism in DCF

In 802.11, DCF also provides a more efficient way of transmitting data frames that involve transmission of special short RTS and CTS frames prior to the transmission of actual data frame. As shown in fig.2, an RTS frame is transmitted by a node, which needs to transmit a packet. When the destination receives the RTS frame, it will transmit a CTS frame after SIFS interval immediately following the reception of the RTS frame. The source station is allowed to transmit its packet only if it receives the CTS correctly. Note that all the other stations are capable of updating their knowledge about other nodes transmission duration by receiving a certain field in RTS, CTS, ACK, and packets transmission called network vector allocation (NAV). This helps to combat the *hidden terminal* problems. In fact, a node that is able to receive the CTS frames correctly, can avoid collisions even when it is unable to sense the data transmissions directly from the source station. If a collision occurs with two or more RTS frames, much less bandwidth is wasted when compared with the situations where larger data frames in collision, thus justifying the case for RTS, CTS mode of operation.

According to the protocol new users typically get the access before existing users who may collide with each other leading to fairness problems [3] – [13].

3. System Analysis For The Ideal Standard Case Without Backoff

Let p be the transmission probability of each node and M be the number of active stations. Assuming no backoff and each user tries to transmit randomly in each slot following the DIFS period, only one user tries his RTS which is then followed by CTS and a successful packet, the probability of successful transmission, is thus given by the following

$$P_s = Mp(1-p)^{M-1} \quad \text{.................................(1)}$$

The probability of an idle slot is

$$P_o = (1-p)^M \quad \text{.......................................(2)}$$

and probability of unsuccessful transmission RTS (collision)is

$$P_c = 1 - P_s - P_o \quad \text{......................................(3)}$$

Let i be the number of idle periods (cycles) to success shown in fig 3 and j be the number of idle slots in each idle period lengths $(W_1, W_2, ...)$. So the efficiency (η_1) is given by equation (7).

It is easily seen that the average length of each idle period except the last one before packet success is given by

$$W_1 = W_2 = ...W_{I-1} = E(j) = \sum_{j=1}^{\infty} j(P_o)^j P_c$$

$$W_1 = W_2 = ...W_{I-1} = \frac{P_o P_c}{(1-P_o)^2} \text{ slots..............(4)}$$

The last idle period has an average of

$$W_I = \frac{P_o P_s}{(1-P_o)^2} \text{ slots...................................(5)}$$

The average number of cycles is given by,

$$I = \sum_{i=1}^{\infty} i P_i$$

$$I = \frac{(1-P_o)}{P_s} \quad \text{..(6)}$$

All cycles leading to no success (RTS heard but no CTS) will each have a cost of $W_i + T_{RTS} + T_{DIFS} + T_{Slot} + T_{SIFS}$ seconds.

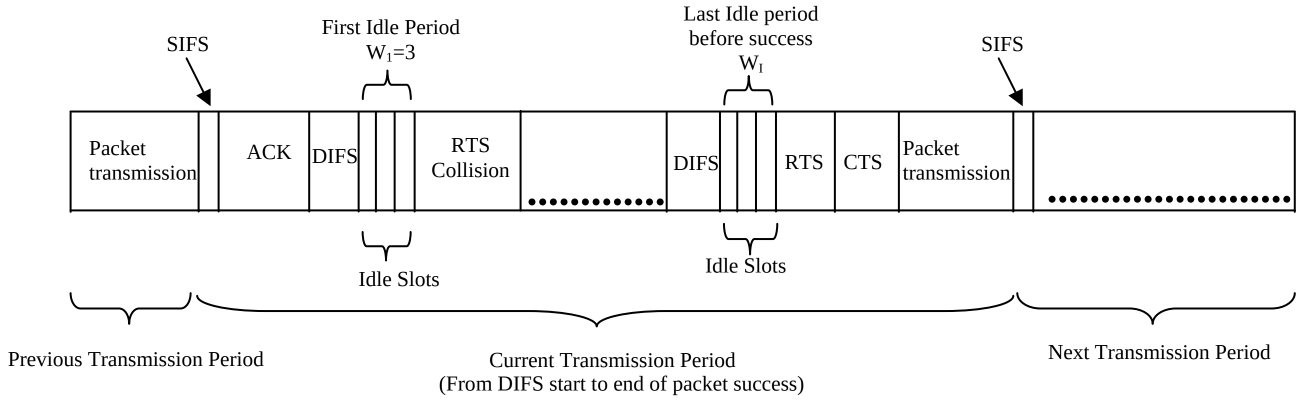

Fig 3. Transmission Activity on the Wireless Channel

$$\eta_1 = \frac{T_{Payload}}{(W_1 + W_2 + ..W_{I-1})T_{Slot} + (I-1)\{T_{RTS} + T_{DIFS} + T_{SIFS}\} + T_{DIFS} + T_{RTS} + T_{CTS} + T_{ACK} + 3T_{SIFS} + T_{Payload} + W_I T_{Slot}} \quad \ldots(7)$$

The number of collisions is $\overline{C} = I - 1$. This $\overline{C}$ and W_I are calculated from different values of M and p and stored in two **tables** (not shown for space consideration). So for particular values of M and p there is a particular value of $\overline{C}$ and W_I .

From equation (7) the efficiency η_1 can be calculated for different values of M and p as in fig. 4. Table 1 depicts the probabilities at which the maximum efficiency occurs for different values of M .

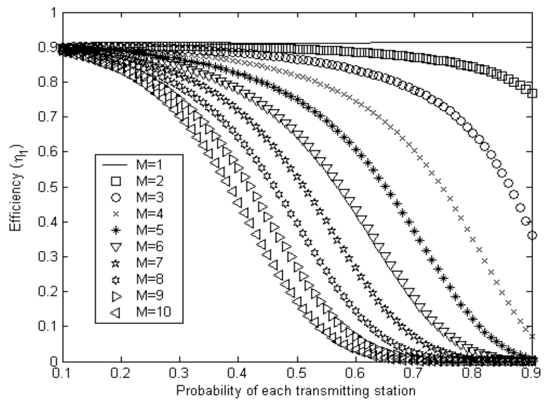

Fig 4. Efficiencies for different probabilities and different number of stations

Table 1. Optimum Efficiencies for different probabilities and different number of stations

No of Stations	Probability	Optimum Efficiency
1	0.9	0.914494552
2	0.25	0.903305479
3	0.15	0.901342654
4	0.11	0.900459291
5	0.1	0.89970777
6	0.1	0.898158644
7	0.1	0.895995929
8	0.1	0.893367296
9	0.1	0.890347837
10	0.1	0.886975732

4. Table Driven WLANs

In this new protocol, if the nodes sense that the channel is idle for an interval called DIFS (DCF interframe space), they try to send RTS of a packet with a probability p which is dependent on the traffic condition i.e. the number and activities of the nodes as follows.

The users continuously monitor the channel in each idle slot following the DIFS, idle period. If the previous slot is idle, it calls a uniform random generator (0,1). If the value of this generator is less than or equal to p , it tries to start its RTS transmission in the given next slot. If the value is larger than p , the users persist on listening and repeats RTS transmission trials as stated. However if the channel is sensed busy the user defers his transmission till the next DIFS idle period heard.

The nodes measure the number of collisions $\overline{C} = I - 1$ and the length W_I of the last idle period sliding, (by monitoring the channel over a large enough window) they can then use the tables formulated in section 3 to obtain the corresponding p and M .

Users having a non-empty queue start by monitoring the channel for the first n transmission periods. This active user will average the length of the idle period preceding the correct packet transmission over n transmission periods i.e. $\overline{W_I}$ and $\overline{C} = I - 1$ i.e. the average number of collision over the same period.

Aided with these values the users obtain the operating values of p and M and uses p to control their activities for the head of line packet in their queue. Active users continuously monitor the channel and use a sliding window technique to estimate W_I and I and hence obtain M , p . For example the first sliding window averages W_I and $\overline{C}$ of the first n transmission periods. The second window averages W_I and $\overline{C}$ of the $l = 2,3,.......n+1$ transmission periods. The third to (n+2) transmission periods. The sliding window averaging process reflects the changing traffic, so transmission activity of active users are dependent only on the current traffic and not on past history.

It is possible that the tables releting (M, p) to (W_I, I) yield more than one possibility for M , p for certain W_I, I measurement values from the sliding window. In this case, the user average the obtained values of M and use table 1 to find the optimum p at this averaged value of M . This table 1 is obtained from Fig. 4 in an evident manner. The operation of this table driven technique is similar to the DCF standard IEEE protocol [4] except for using this optimized transmission probability p and discarding the timers and backoff windows. The active users just estimate M , p from the traffic conditions (by sensing the channel) in a sliding window fashion transmission, one period after another.

We note that old and fresh users both measure the traffic and both adopts to same traffic condition fairly and obtain same p . However having same p does not mean all users will repeatedly collide in same slot. Since feeding a random number generator with p yields different slot number to start transmitting the RTS each time it is called.

5. Buffer Adaptive WLANs

In the buffer adaptive technique, each node's probability of transmission's trying is calculated based on the number of packets stored in the buffers. The probability of trying, p of each node is $\frac{Bu}{Bu_{\max}}$,

Where Bu is the number of packets stored in the buffers at each node and $Bu_{\max}$ is the user buffer capacity. The buffer adaptive technique is simple, it is similar to the IEEE 802.11 RTS/CTS, DCF mode except for the elimination of timers and exponential backoff. It is also similar to the table driven technique except for the elimination of table construction simply the users adjust their random transmission probability p fullowing the standard DIFS period, continuously based on the queue size. This technique does neither need traffic measurement nor table establishments.

6. Simulation Results

For numerical calculations the following parameters are used:

$T_{Payload}$=10msec;PHYheader=128bits;ACK=112bits +PHY header; RTS=160bits+PHY header; CTS=112bits+PHY header; Channel bit rate= 1 Mbits/s; Slot time (T_{Slot})= 50 µs; T_{SIFS}=28 µs; T_{DIFS} =128 µs.

In the table driven technique, as per the standards, following the observance of each DIFS, users try to transmit with probability p obtained from the above table which is obtained from the traffic measurements. So all users with non empty buffer try to transmit a packet with a probability obtained from the table. If two or more stations try to transmit at the same time, collisions occur. If no stations transmit (Fig 3), the number of idle slots will increase. If one station is successful after certain number of idle and collision period, the transmission period ends. As a result the total time for one successful packet transmission include T_{DIFS}, T_{SIFS}, T_{RTS}, T_{CTS}, T_{Idle}, $T_{Payload}$. The efficiency is calculated at the end of the simulation at certain values of M , λ , p i.e.,

$$\eta = \frac{T_{Payload} \times No\ of\ Transmission\ Periods\ in\ the\ whole\ simulation}{Time^{(n)}}$$

Where $Time^{(n)}$ is the total simulation time which depends on T_{DIFS}, T_{SIFS}, T_{RTS}, T_{CTS}, T_{Slot}, $T_{Payload}$.

Initially $Time^{(n)} = T_{DIFS}$ and subsequently increased based on the user's activity, e.g.

$$Time^{(n)} = Time^{(n)} + T_{Slot}\ ;\ for\ each\ idle\ Slot\ period$$

$$Time^{(n)} = Time^{(n)} + T_{RTS} + T_{DIFS} + T_{SIFS};\ for\ each\ collision$$

$$Time^{(n)} = Time^{(n)} + T_{RTS} + T_{CTS} + T_{DIFS} + T_{3SIFS} + T_{Payload};$$
$$for\ each\ successful\ packet$$

In the case of the buffer adaptive technique, the simulation time is calculated in the same way as per user's activity above. However, no table is constructed and sliding windows don't apply.

In the table driven technique described in section 4 the active stations estimate the value of M . At certain traffic the curve shown in fig. 5 is linear, that means the offered and the estimated values of the number of active stations are the same.

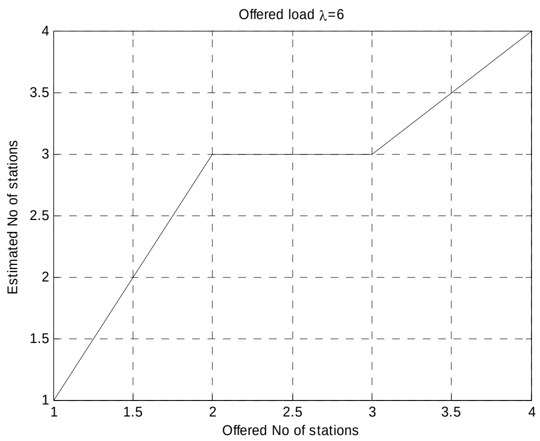

Fig 5. Estimated M at a certain Traffic

The table driven technique can be considered as a load adaptive system. That means it has the capability to adapt to the input traffic as quickly as possible. Figure 6 shows a case where the input traffic suddenly increase from 5 packets/sec to 10 packets/sec. In this case the throughput$(\eta \times Input\ traffic\ rate(\lambda))$ (Fig. 6) is shown to follow the offered traffic λ.

Fig. 7 shows the efficiency curve for different offered loads for the table driven technique for different window sizes. This shows that the efficiency rises and becomes saturated at higher values of the load. The window size has small effects on the efficiencies at different loads.

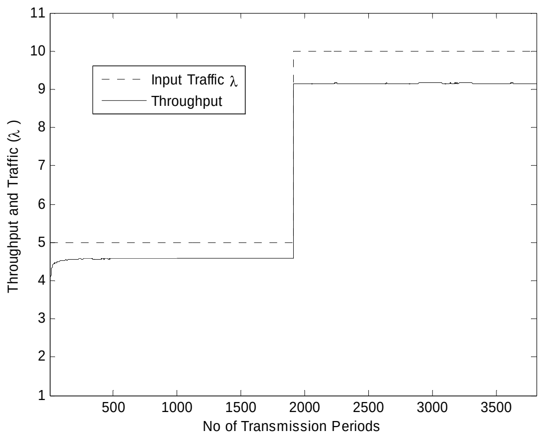

Fig 6. Throughput and Input Traffic corresponding to the number of Transmission periods.

Fig. 8 depicts the packet delay corresponding to different loads for the table driven technique for different window sizes. The window sizes have little effect on the packet delay corresponding to different loads.

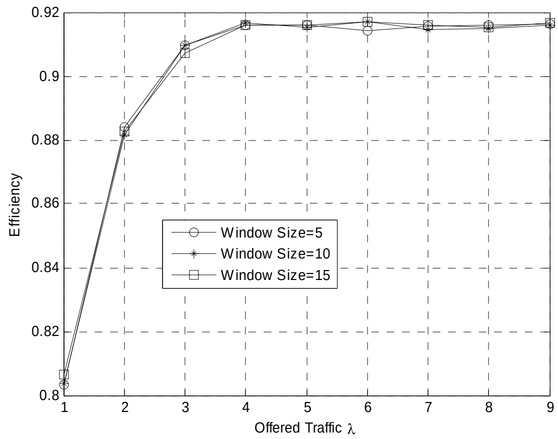

Fig 7. Efficiency corresponding to different offered traffic using different window size

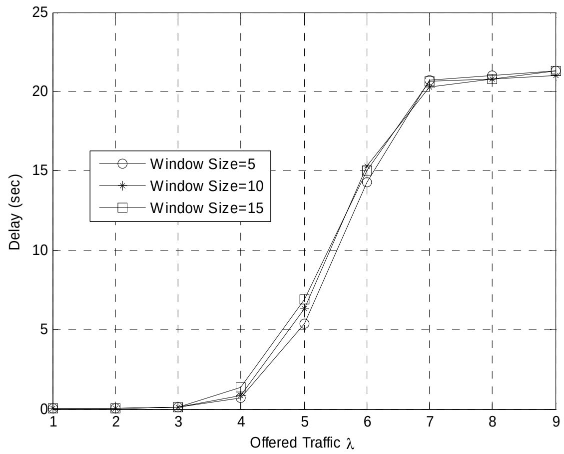

Fig 8. Delay corresponding to different offered traffic using different window size

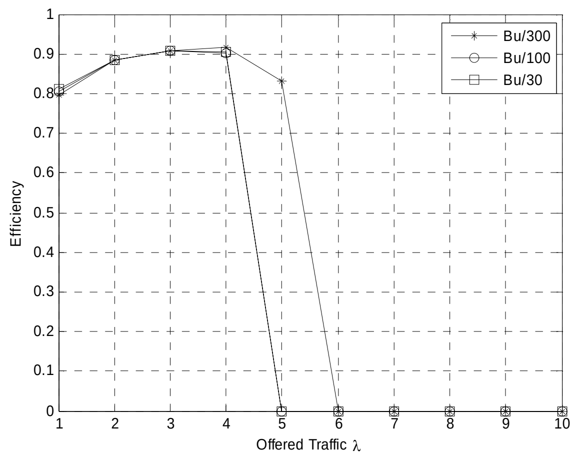

Fig 9. Efficiency at different loads for Buffer adaptive system for different capacities

As the efficiency rises with the increased load, excess numbers of packets are left in the buffers. This results in a large packet delay at higher loads.

Fig. 9 and fig 10 shows the efficiency and the delay performance at different loads for the buffer adaptive technique. From 2 packets/sec to 4 packets/s, the efficiency of the buffer adaptive technique and the table driven technique are more or less the same. Beyond 4 packets/s, in the buffer adaptive technique the efficiency becomes very small, because all stations transmit their packets with higher probability which results in high amount of collision and no success. In these figures efficiency and the delay are calculated for different buffer capacities, such as, 30, 100, 300.

Fig. 10 shows that the delay performance degrades as the capacity of the buffer increases. However the efficiency increases as the buffer capacity increases (fig. 9).

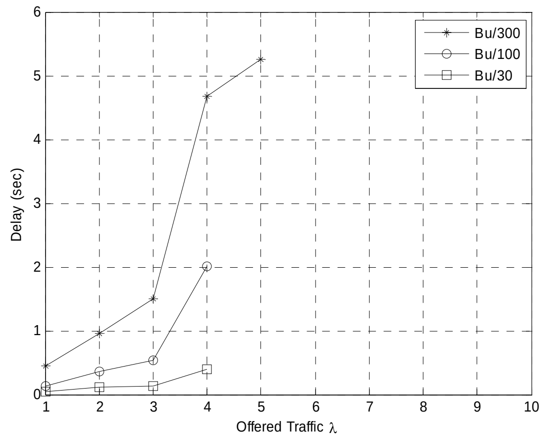

Fig 10. Delay curves at different loads for Buffer adaptive system for different capacities

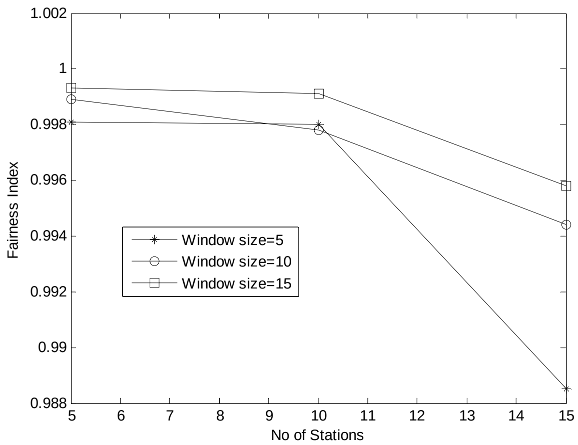

Fig 11. Fairness Index for the table driven technique for different window sizes

Fairness is another important issue. To express this, we take the fairness index defined in [2] to measure the fair packet capacity allocation. That is, $FI = \frac{\left(\sum_{i=1}^{n} x_i\right)^2}{n\left(\sum_{i=1}^{n} x_i^2\right)}$, where FI is the fairness index, n is the number of stations, x_i is the packets transmitted by the i^{th} active station during the simulation time (current traffic in which the offered traffic λ is same for all stations) .

Fig. 11 shows that the fairness index decreases as the window size is decreased for the case of table driven technique.

Fig. 12 shows the fairness index of the buffer adaptive technique. From this we can observe that the stations can be fairly operated when the buffer capacity is made high.

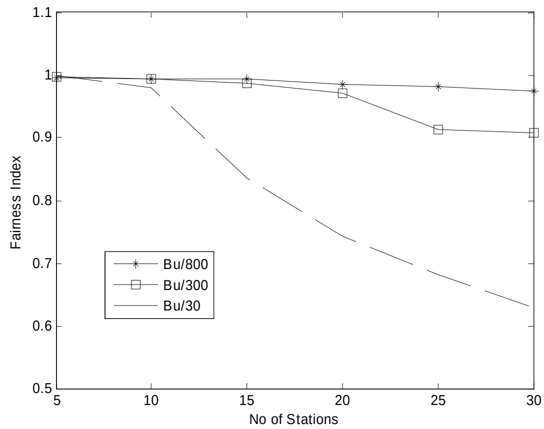

Fig 12. Fairness Index of the Buffer adaptive technique for different buffer capacities

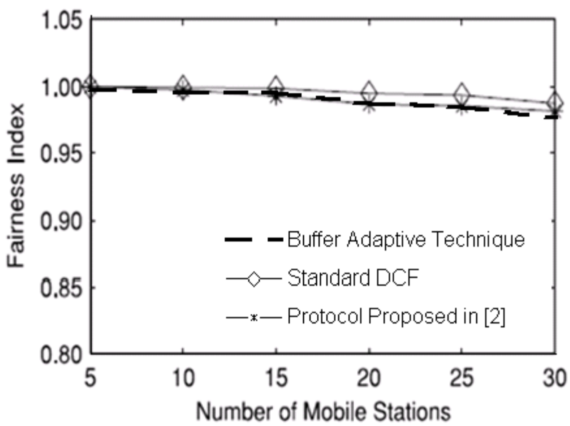

Fig 13. Fairness Index for different number of stations for different cases

Fig. 13 shows a comparison of the Fairness index between the buffer adaptive technique and the protocol proposed in [2]. We conclude that the buffer adaptive technique yields the same fairness index as in [2] which use modified exponential backoff.

Fig. 14 shows a comparison between table driven technique and buffer adaptive technique along with the protocol proposed in [2]. It can be observed that, for all the cases up to 20 active stations the performance is the same. Beyond that load, the fairness of the table driven technique degrades.

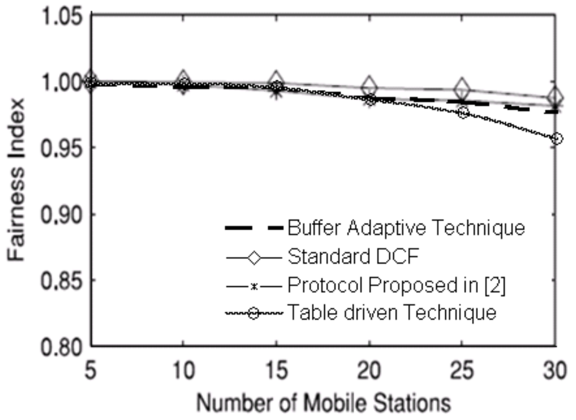

Fig 14. Fairness Index for different number of stations for different cases

Now let us introduce the Delay variance for the two different new techniques. Delay variance is calculated by $DV = \frac{\sum_{i=1}^{n}(D_i - D_{average})^2}{n}$, where DV is the delay variance, D_i is the average packet delay of the i^{th} station and $D_{average}$ is the average packet delay of all stations.

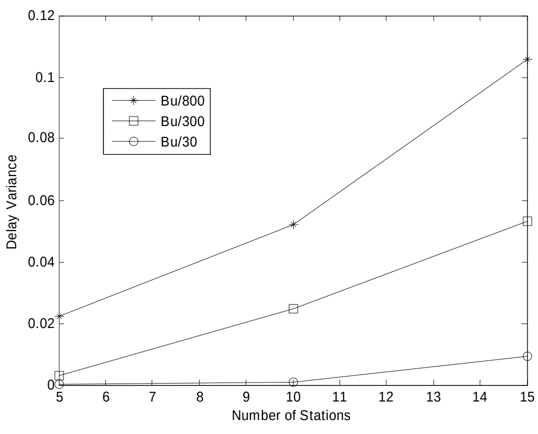

Fig 15. Delay Variance for the Buffer adaptive technique for different buffer capacities

Fig. 15 shows the delay variance for the buffer adaptive technique for different buffer capacities. It is observed that the delay variance increases in accordance with the number of stations as well as the buffer capacity.

Fig. 16 shows the delay variance of the table driven technique for different window sizes. The delay variance performance is worse than the case of buffer adaptive technique.

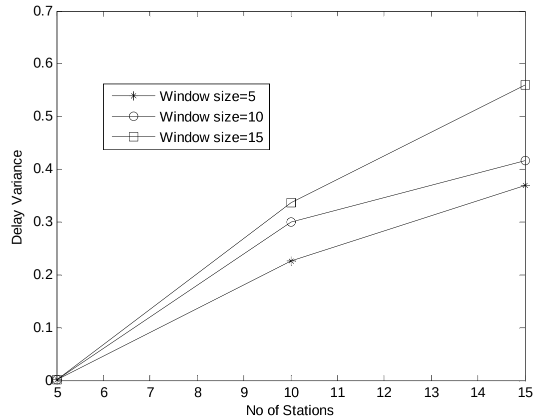

Fig 16. Delay Variance for the Table driven Technique for a different window sizes

7. Conclusion

In this paper two new techniques (Table driven and Buffer adaptive) were presented, modeled and compared.

Simulation results show that the table driven technique performs well for faster load variations, where as the buffer adaptive technique does not. But the buffer adaptive technique has FI (fairness index) and DV (Delay Variance) performances better than the table driven technique making it suitable for real time application (voice, video etc). The buffer adaptive technique does not scale well at higher loads as compared to the table driven technique. The throughput and delay performances are better in the case of table driven technique making it suitable for data application.

References:

[1] Xuejun TIAN, Xiang CHEN, Tetsuo IDEGUCHI and Yuguang FANG, **"**Improving Throughput and Fairness in WLANs through Dynamically Optimizing Backoff," *IEICE Transactions on Communications 2005* E88-B(11):4328-4338.

[2] Liang Zhang, Yantai Shu and Oliver Yang, "Performance Improvement for 802.11 Based Wireless local Area networks" *IEICE Trans. Commun.*,vol. E90-B, NO.4 April 2007.

[3] V. Bharghvan, "Performance evaluation of algorithms for wireless medium access," *IEEE International Computer Performance and Dependability Symposium IPDS'98*, pp.86–95, 1998.

[4] G. Bianchi, "Performance analysis of the IEEE 802.11 distributed coordination function," *IEEE J. Sel. Areas Commun*., vol.18, no.3, pp.535–547, March 2000.

[5] Y.C. Tay and K.C. Chua, "A capacity analysis for the IEEE 802.11 MAC protocol," *ACM/Baltzer Wireless Networks*, vol.7, no.2, pp.159–171, March 2001.

[6] J.H. Kim and J.K. Lee, "Performance of carrier sense multiple access with collision avoidance protocols in wireless LANs," *Wirel. Pers. Commun.*, vol.11, no.2, pp.161–183, 1999.

[7] F. Cali, M. Conti, and E. Gregori, "Dynamic tuning of the IEEE 802.11 protocol to achieve a theoretical throughput limit," *IEEE/ACM Trans. Netw.*, vol.8, no.6, pp.785–799, Dec. 2000.

[8] Y. Kwon, Y. Fang, and H. Latchman, "A novel MAC protocol with fast collision resolution for wireless LANs," IEEE INFOCOM'03, April 2003.

[9] J. Weinmiller, H. Woesner. J.P. Ebert, and A. Wolisz, "Analyzing and tuning the distributed coordination function in the IEEE 802.11 DFWMAC draft standard," *Proc. MASCOT, San Jose, CA, Feb. 1996.*

[10] H. Wu, Y. Peng, K. Long, S. Cheng, and J. Ma, "Performance of reliable transport protocol over IEEE 802.11 wireless LAN: Analysis and enhancement," *IEEE INFOCOM'02*, vol.2, pp.599–607, June 2002.

[11] H.S. Chhaya and S. Gupta, "Performance modeling of asynchronous data transfer methods of IEEE 802.11 MAC protocol," *ACM/Baltzer Wireless Networks*, vol.3, pp.217–234, 1997.

[12] G. Bianchi and I. Tinnirello, "Kalman filter estimation of the number of competing terminals in an IEEE 802.11 network," *IEEE INFOCOM'03*, vol.2, pp.844–852, April 2003.

[13] Farshad Eshaghi, Ahmed Elhakeem, Yousef R. Shayan, "Performance Evaluation of Multihop Ad Hoc WLANs", *IEEE Communications Magazine, March 2005*.