Improved MAC Protocols for DCF and PCF Modes over Fading Channels in Wireless LANs

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***Abstract*-Wireless LAN has become increasingly popular as a medium to connect portable devices to the network. Most of the existing wireless LAN protocols based on IEEE standard 802.11 assume that the wireless channel is time invariant and thus assign radio resources accordingly. In this paper, we devise a modified MAC level channel allocation method by carefully scheduling radio resource according to the temporal channel characteristics, assuming the influence of channel fading. Based on this method we propose new protocols for the Distributed Coordination Function (DCF) and the Point Coordination Function (PCF), the two modes of communication defined in the IEEE standard for WLAN. The proposed protocols introduce minimum computational overhead. Our simulations show that the proposed PCF protocol improves the channel capacity up to 14% and the DCF protocol improves the channel capacity up to 90%, depending on the loss rate and temporal characteristics of the wireless channel. We also show that the proposed DCF protocol can lower the SNR requirements thus potentially extending the battery life of portable devices that use WLAN.**

Keywords-Wireless LAN; PCF; DCF;

I. INTRODUCTION

Recently Wireless LAN has gained popularity for its capability to connect portable or mobile devices to the Internet. In 1997 IEEE published 802.11 standard for the physical and MAC layer protocols of the wireless LAN. The MAC layer protocol of 802.11 based on Carrier Sense Multiple Access (CSMA) technology is much like IEEE 802.3 Ethernet protocol. The original physical layer specifications supported traffic rate up to 2Mbps with two subsequent extensions 802.11b and 802.11a that can support traffic up to 11Mbps and 50Mbps, respectively. However, the MAC layer protocol remained unchanged. The protocol designers introduced a collision avoidance procedure to avoid collision with hidden and exposed station, but the channel itself was treated as time-invariant. However, in the existing literature wireless channel has been modeled as a multipath, time-variant fading channel [1]. Let's look at Figure 1 to see how such a model contradicts the existing CSMA wireless LAN protocols.

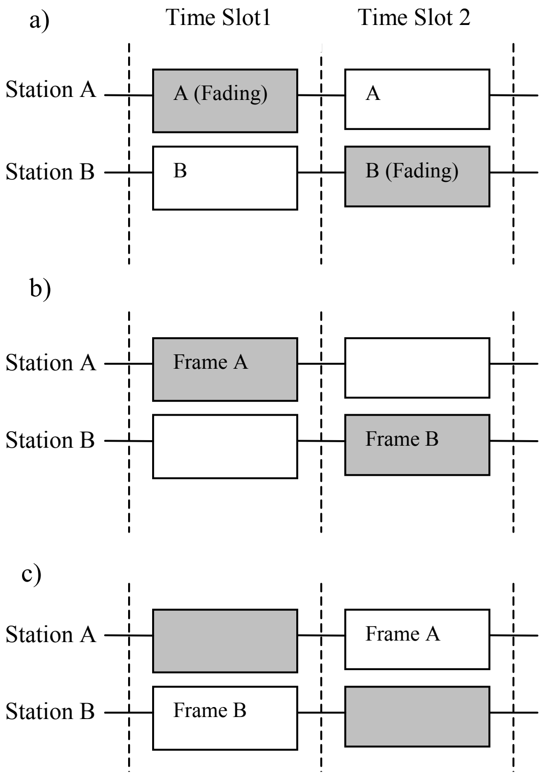


Figure 1: The relationship between channel status and packet transmission, the gray box stands for time slots suffering deep fading so packets transferred in those time slots could not be received

a) Channel status: gray slot denote that the channel is in deep fading
b) Both frames are lost due to channel fading
c) Both frames are successfully received

Let's suppose two stations, station A and station B, both have one frame to transmit to a destination, for example an access point (AP). In Figure 1a, Channel A denotes the wireless channel between station A and AP; Channel B denotes the channel between station B and AP. The two wireless channels may have different fading patterns. Suppose in Time Slot 1, channel A is in deep fading so that frame sent from A to AP will probably be lost, but channel B does not suffer deep fading at that time. Similarly, in Time Slot 2 channel B is in deep fading but channel A is okay. According to the CSMA protocol, frames may be chosen randomly for transmission in each slot. So if the transmission order is as shown in Figure 1b, both frames will be lost due to deep fading, although no contention happened. However, if we can predict the channel status in advance and then send the frames in the order as shown in Figure 1c, both frames will reach the destination. So if we have the knowledge of the channel fading status, we may assign the radio resource (i.e. time slots) to the users in a more efficient way such that data frames have a better chance of reaching the destination.

In this paper, we study the temporal characteristics of the fading wireless channel and propose new protocols for DCF and PCF modes that assign radio resources based on the channel status, thus optimizing the channel capacity. Our simulations show that, depending on the channel characteristics, the proposed protocols can improve the average system capacity up to 14% for PCF mode and up to 90% for DCF mode.

The organization of the rest of the paper is as follows. In the next section, we analyze the channel characteristics and present a method to predict the fading envelop. In Section III we introduce the modified MAC level protocols for DCF and PCF modes and analyze their fairness. In Section IV we present the simulation results. Finally we draw some conclusions in Section V.

II. THE TIME VARIANT WIRELESS CHANNEL MODEL

The most frequently used approach to model the signal received over wireless fading channels is Rayleigh or Rician Distribution model since they are both simple in math and close to the reality [4]. The envelop, *r(t)*, of the received signal in Rayleigh model can be expressed as:

$$r(t) = \sqrt{x_I(t)^2 + x_Q(t)^2}$$

Where $x_I(t)$ and $x_Q(t)$ represent the two statistically independent channel path loss behaviors, and are assumed to have Gaussian distribution. Both $x_I(t)$ and $x_Q(t)$ have zero mean and same variance σ^2. Where the variance σ^2 is the time-average power of the received signal before envelope detection. To be fair, we need to normalize the variance while calculating the resource assignment. It could be achieved by averaging the receiving power level of recently received packets. The temporal relationship of the signals could be formulated as [2]

$$\rho = < x(t)x(t+\tau) > = \sigma^2 J_0(2\pi f_d \tau)$$

Where f_d is the maximum Doppler frequency shift, τ is the time difference between two samples and J_0 is the zero-order Bessel function of first kind.

In [1] it was verified that it is accurate enough to model the Rayleigh fading envelop as a first order Markov process. That is, given the observation of the fading envelope at time *t*, *t-τ*, *t-2τ*,..., the probability distribution at time t+1 only depends on the observation at time *t*. Based on this conclusion, we can formulate the conditional distribution function of the receiving signal as:

$$P(r(t+\tau)\,|\,r(t)) = \sum_{\substack{x_Q(t+\tau)=\pm\sqrt{r^2(t+\tau)-x_I^2(t+\tau))} \\ ,x_Q(t)=\pm\sqrt{r^2(t)-x_I^2(t)}}} \int_{-r(t)}^{r(t)} P(x_I(t)\,|\,r(t)) \int_{-r(t+\tau)}^{r(t+\tau)} P(x_I(t+\tau)\,|\,x_I(t)) P(x_Q(t+\tau)\,|\,x_Q(t))\,dx_I(t+\tau)\,dx_I(t)$$

Since it is hard to derive a close form for this conditional probability distribution, it is convenient to use simulation to approximate the conditional distribution. After deriving this distribution, we can establish a relationship between the receiving envelope and the frame error rate. In [3] it is shown that a threshold model could be used. That is, if the SNR is greater than a given threshold, the frame is received successfully with a probability of 1; otherwise the frame is lost with a probability of 0. i.e. the error probability $P_e(t)$ can be formulated as

$$P_e(t) = \begin{cases} 0, r(t) > Th \\ 1, r(t) \le Th \end{cases}$$

So given the receiving power level at time t, we can anticipate the probability of successfully receiving the next frame transmitted at time t+τ, that is:

$$P_s(t+\tau) = 1 - P_e(t+\tau) = \int_{\sqrt{ThN}}^{\infty} P(r(t+\tau)\,|\,r(t))\,dr(t+\tau)$$

Using the above formula, we can build the probability table showing the receiving probability of each station at different time slots for independently fading channels.

Suppose there're four users trying to send packets to an access point (suppose all the packets have the same length), then an example probability table may look like as in Table 1. Note that the entries in the example table are arbitrary. In this table, rows are the receiving probability of each time slot for a certain station, calculated based on the average receiving power of the most recently received packet (even a damage packet). Columns of the table show the probability of all stations in a certain time slot.

We need to find a permutation from stations to time slots, such that.

$$\arg\max_{f(i)} \sum_i p_{i,f(i)}$$

Where $p_{i,f(i)}$ is the probability of successfully receiving a frame from station *i* transmitted at time slot *f(i)*. The idea is to maximize the average number of successfully received frames. This problem can be formulated as an integer linear programming problem [8]. The optimal assignment for our

	T1	**T2**	**T3**	**T4**
STA A	0.4753	0.4849	0.4906	0.4997
STA B	0.8411	0.7306	0.6511	0.6005
STA C	0.5019	0.4954	0.5024	0.5025
STA D	0.5957	0.5553	0.5354	0.5221

Table 1: An example probability table with 4 stations and 4 time slots.

example is A-4, B-1, C-3 and D-2 with a total receiving probability of 2.3985. That is station B should transmit first, followed by D, then C, and then A.

III. THE PROPOSED PROTOCOLS

Based on the analytical analysis in the previous section, we develop efficient channel resource allocation algorithms. Let's consider a WLAN Basic Service Set (BSS) consisting of an Access Point (AP) and several stations associated to it. The stations may or may not be hidden to each other, which means they may not notice the existence of others, as shown in Figure 2.

We propose two different protocols for the DCF and PCF modes. Essentially, the main idea governing both the protocols is based on utilizing the temporal characteristics of the wireless channel.

3.1 The protocol for DCF

The Distribution Coordination Function (DCF) of IEEE 802.11 is a contention-based protocol, like the Ethernet protocol IEEE 802.3. In the wireless environment, remote stations may not hear from each other due to path loss. So a ground-acquire procedure known as Collision Avoidance (CA) has been introduced to let the others know that the communication pair wants to use the radio resource. This procedure starts with the transmitter sending a ready to send frame (RTS) to the destination, then the destination replies by sending a clear to send (CTS) frame.

Instead of let RTS frames collide in the air and stations perform exponential backoff individually, as defined in the existing DCF protocol, our proposed protocol works as follows:

1) The destination listens to the channel for the whole contention window instead of just replying the first valid RTS request. Then it may receive several RTS frames, one from each user.
2) The destination measures the receiving power level from each user. (This could be acquired from the physical layer to MAC layer primitive PHY_RXSTART.ind) and calculates the fading status of each station.
3) The destination chooses the station with the best channel status and sends a CTS frame to that station. (The best here means the one with the highest receiving probability in the following time slots).
4) After receiving the CTS frame, the chosen station sends its data to the destination.
5) Once the media is clear for time longer than DCF's inter-frame space (DIFS), new communication rounds starts.

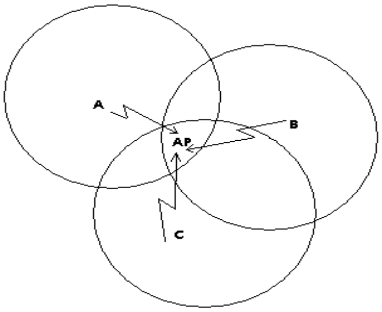
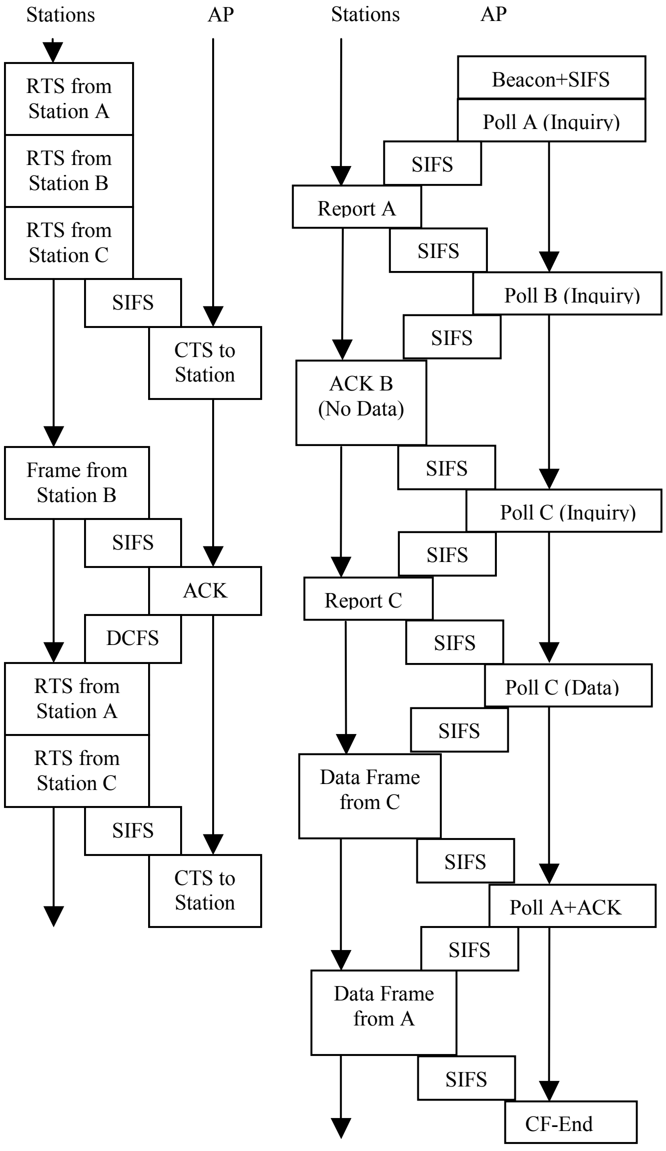


Figure 2: An example 802.11 WLAN scenario with 3 users associated to an AP and each user is invisible to

Figure 3a elaborates the flow of control and data frames for an example scenario. In this example, the BSS consists of three stations, namely A, B and C. At the first round all three stations send RTS frames to AP since they have data to



(a) (b)

Figure 3: PCF and DCF scenarios
(a) DCF scenario with 3 stations in the BSS
(b) PCF scenario with 3 stations in the BSS

transfer to AP. After receiving these packets, AP calculates the channel status for each stations and finds out that the channel status of station A is the best, so it sends back a CTS frame to A. All three stations hear that CTS frame and know the following time slot is assigned to A. So after the short inter frame space (SIFS) duration, the station A starts transmitting its data. If the data frame is received correctly, AP replies with an ACK frame. Once the media is idle for longer than DIFS duration a new round of ground acquisition and data transmission starts.

This proposed protocol is fully compatible with the existing 802.11 MAC layer protocols. No frame format and procedure change is required. The only thing we need to modify is the parameter of CTSTimeout. We need to increase the value of this parameter to prevent stations retransmit RTS frames before they get CTS.

3.2 The protocol for PCF

The IEEE 802.11 standard for MAC layer protocol has also defined an optional access method called Point Coordination Function (PCF). When BSS works under PCF, AP controls the access to the radio resource. Only one station is allowed to communicate at a given time and AP determines the order of this assignment. Most of the existing implementations of the PCF statically determine this assignment.

In the protocol proposed for PCF, we let AP dynamically decide the service order based on the channel status. The steps of the proposed protocol are as follow:

1) After the Beacon, AP serially sends CF-poll to each active station in the BSS.
2) Instead of transmitting data fragments immediately, each station first reports how much data it has to send.
3) After receiving reports from all the stations, AP calculates the assignment based on the temporal characteristics of the wireless channel for each station and the history of the assignment in the previous PCF slot.
4) AP polls the stations again in the order of the assignment. When the station is polled, it sends one data fragment to the station.

An example of PCF scenario is shown in Fig. 3b. In this example, there’re 3 stations associated with the AP. After the beacon, AP polls all the three stations to see whether they have data to transmit. Stations A and C have data in the send buffers, so they report how much data they have to transmit. Station B has no data to send, so it just replies back with an ACK frame. After receiving responses from all the stations, AP calculates the optimal assignment and polls stations accordingly for data.

In PCF we need to perform the linear programming algorithm to decide the data polling sequence for each round since there are multiple assignments and we need to maximize the overall probability of success. The calculation of the optimal assignments is based both on channel status and previous assignment history. Timer should be set to guarantee that no stations wait too long.

The PCF protocol has a higher time complexity due to linear programming. It is shown in [8] that the average time complexity of a linear programming algorithm is 3*m*, where *m* is the number of unknowns in the problem. In our problem, *m* is equal to the product of number of users in the BSS and number of time slices in a single CF duration. Thus the worst case time complexity is $O(N^2)$ for the decision algorithm, where N is the number of users. For 802.11b, one access point can support from three 2Mbps users (for high quality streaming video) to sixty 100kbps users [10]. Since the number of users in one BSS is limited, therefore, it is possible for AP to complete the calculation in real time.

3.3 Fairness of the Proposed DCF and PCF Protocols

In this section we discuss the fairness of the proposed protocols since they are dependent on the channel behavior.

For the PCF protocol, it is obvious that if the receiving power from each station is identical, then the algorithm will lead to an unbiased assignment since the receiving power levels of stations have same distribution. Power controlled multiple access protocols have been suggested either for power saving purposes [6] or for capacity improvements [5]. By using such protocols, the transmitter and receiver negotiate the transmitting power before transmitting the data frames. Therefore, the average signal power reaching the destination will be same. Our proposed protocol is fair even if we assume that the average receiving power is different. Every station has a bounded waiting time to transmit its data, no matter how good or bad its channel status is. The bound on the waiting time is given as follows:

$$\left\lceil \frac{(N-1)\times MaxMPDUTime}{CFPMaxduration} \right\rceil \times \frac{1}{CFPRate}$$

For the DCF protocol, however, if the threshold is different for each station, the algorithm will be biased towards stations transmitting with higher power. We can eliminate the bias by using the current receiving power level/average receiving power level as the measurement of the channel status and select the user with best channel status. In this way each station has an equal probability of being chosen for transmission.

IV. THE SIMULATION RESULT

We have simulated the performance of the proposed DCF and PCF protocols using MATLAB communication blockset. The frame length is fixed at 2300 bytes, which is the maximum length allowed by the 802.11 MAC protocol. Under this scenario it requires about 1.8 milliseconds to transmit the data frame and receive the confirmation ACK frame. We assume a retransmission error recovery protocol. If one frame is lost during transmission, it needs to be retransmitted. We also assume that all the channels fade independently. The channel correlation factor ρ is defined as in section 2. The average channel loss rate is the mean frame loss probability if we continuously transmit data frames on that channel. The performances are measured as the number of frames transmitted successfully per second.

4.1 DCF protocol

In this section we compare the performance of the proposed DCF protocol with the standard DCF protocol. We assume 8 users are associated with the AP at any given point in time. In the simulation, we also consider the protocol overhead. In both the standard and the proposed DCF protocols, station will wait for a DIFS after media is idle, then wait for a random backoff time, then exchange RTS and CTS. So the protocol overhead is:

$$T_o = DIFS + T_{Backoff} + T_{RTS} + 2\times SIFS + T_{CTS}$$

The difference is that in the standard DCF AP will reply a RTS as soon as it receives a valid RTS, so the mean value of backoff time is ½ of the backoff window; while in the proposed protocol, AP needs to wait for the entire backoff window. Since there are 8 stations associated, so a backoff window of 8 time slots is enough. T_{RTS} and T_{CTS} are the times required to transmit RTS and CTS frames, respectively. These two items could be ignored since both frames are very short. By using the 802.11b physical layer protocol parameters, such as Slot Time = 20 microseconds, SIFS = 10 microseconds, and DIFS = 50 microseconds, the overhead for the standard protocol (T_{os}) is 150 microseconds and overhead for the proposed protocol (T_{op}) is 230 microseconds.

We have simulated the performance with different values of ρ, i.e. channel correlation, and different traffic loads. We simulate the traffic model as Poisson processes with a parameter λ. The traffic load is calculated as the mean number of frames to be transmitted from all the stations. For example, a 100% traffic load implies λ = 0.125 for each station. We assume all stations has unlimited buffers and data generation processes at each station are independent. The simulation results for DCF are shown in Fig. 5 and 6. The solid lines show the performances of the proposed DCF protocol and dashed lines show the performances of the standard DCF protocol. The number 50%, 100%, 200% is the traffic load.

The simulation results show that the proposed DCF protocol performs extremely well when the traffic load is high. The performance decreases when ρ decreases, i.e., when channel is less correlated. However, even with ρ=0.8, we have an improvement of 70% for an average channel loss rate of 0.5. When the traffic load is low, the proposed protocol performs a little bit worse (no worse than 10%) than the standard protocol. This is due to the higher overhead involved in the proposed protocol.

Regardless of the channel correlation and traffic loads, our proposed protocol utilizes the channel capacity to the fullest. This is evident from the flatness of the curves of the proposed protocols in all the three figures. In other words, we show that the number of frames lost can be bounded in a narrow range regardless of the channel loss rate. This provides another interesting observation related to power usage.

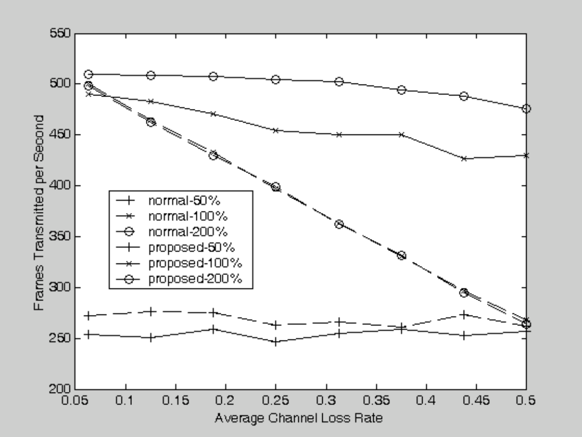


Figure 4: The DCF protocol performance, ρ=0.9

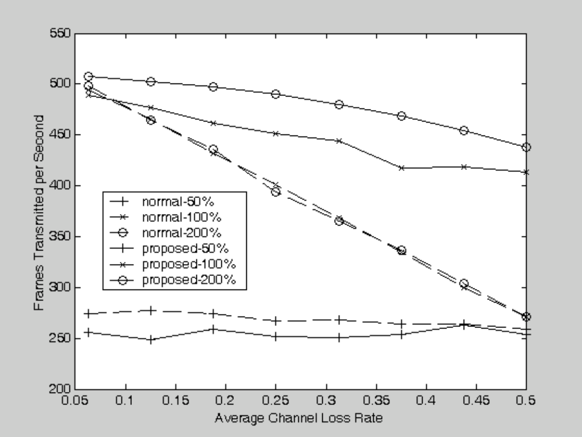


Figure 5: The DCF protocol performance, ρ=0.8

Since channel loss rate is a function of SNR, as shown in equation 3, this means we can decrease the transmitting power without significant performance loss. Using the threshold model of equation 2, in order to guarantee an average channel lost rate Pe = 0.1, we have

$$1 - P_e = P\left(\frac{r^2(t)}{N} > Th\right) = \int_{\sqrt{ThN}}^{\infty} P(r)dr = 0.9$$

Assuming Rayleigh fading envelop with unit variance, we can compute *ThN* = 0.21. On the other hand if we transmit with power 1/α and drop the average channel lost rate =0.4, we have

$$1 - P_e = P\left(\frac{\frac{1}{\alpha} r^2(t)}{N} > Th\right) = \int_{\sqrt{\alpha ThN}}^{\infty} P(r)dr = 0.6$$

This yields αThN =1.02. Thus α=1.02/0.21=4.8.

This means we only need to transmit data frames with a power level of 1/4.8 (-6.8dB) of the original power to stay in the same frame loss range, which means we can extend the battery life to 4.8 times! This is very attractive for battery-powered devices.

4.2 PCF protocol

In this section we compare the performance of our proposed PCF protocol with the standard PCF protocol. In the simulations, the overhead associated with the start and end of a PCF duration is ignored since it is relatively short compared to the whole polling period. Also we assume that a fixed number of stations are associated with one BSS (in our experiments it is 8) and all the stations are fully loaded with PCF data requests. We evaluate the performance of the proposed protocol assuming different divisions of the contention free (CF) period, different channel characteristics and different channel loss rates. The simulation results are shown in Figures 7 and 8.

In the equal division case we divide the CF period into N number of slots of equal length, where N is number of stations that have requested to use CF. In the 4-division and 2-division case, the CF duration is divided into 4 and 2 equal length slots, respectively, regardless of the number of stations in the BSS.

Solid lines show the performances of proposed protocol and dashed line show the performances of standard protocol.

From these figures we can easily conclude that the equal division protocol has the best performance for all the cases, with a capacity improvement from 2.4% (for low error rates) to 14% (for high error rates). This is due to the fact that the number of choices to assign radio resource is higher in the equal case (i.e. 8) compared to the other schemes. This yields better assignment and thus higher success probability.

We also find that the performance is better for higher values of ρ. This is because with high value of ρ, we can predict the receiving probability in a more accurate way.

Though in the simulation we assume all the stations have same traffic load, however, it is not a requirement of the protocol. In order to handle a station with a higher data rate, we only need to treat it as several virtual stations running on the same physical machine.

V. CONCLUSION

In this paper we have analyzed the temporal characteristics of the fading channel. We have shown that by using the knowledge of the channel's fading status, we can efficiently schedule the time for each station to transmit their data. The proposed protocols introduce minimum computational overhead. Our simulations show that the proposed PCF protocol improves the channel capacity up to 14% and the DCF protocol improves the channel capacity up to 90%, depending on the loss rate and temporal characteristics of the wireless channel. We also show that the proposed DCF protocol can lower the SNR requirements thus potentially extending the battery life of portable devices that use WLAN.

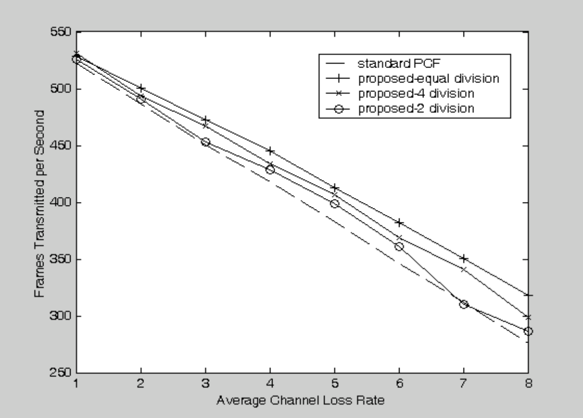


Figure 6: The PCF protocol performance, ρ=0.9

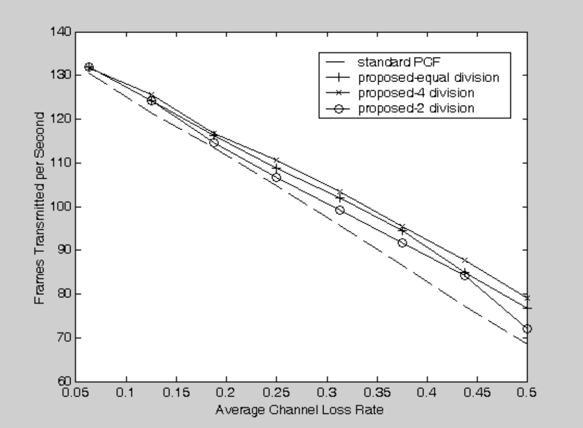


Figure 7: The PCF protocol performance, ρ=0.8

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