Medium Access Control with Physical Layer Assisted Loss Differentiation

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Abstract— **The binary exponential backoff (BEB) algorithm used in IEEE 802.11 DCF suffers from the unfairness problem and yields low throughput under heavy load. With physical layer assisted loss differentiation, this paper proposes a new distributed queuing medium access control (MAC) protocol (PALD-DQMP). In this protocol, utilizing the user detection module in the physical layer, losses due to collisions are distinguished from those due to link errors, and such information is made available to the MAC layer. Based on different users' channel states, PALD-DQMP schedules their transmissions. Simulation results show that the proposed scheme outperforms the standard MAC protocol in terms of network throughput and fairness. This improvement is mainly due to the availability of cross-layer channel information, and the elimination of collisions and backoff periods.**

I. INTRODUCTION

The IEEE 802.11 standard [1] is one of the most popular wireless local area network (WLAN) technologies. For medium access control (MAC), the fundamental mechanism is the distributed coordination function (DCF). Retransmission of collided packets is managed according to the binary exponential backoff (BEB) algorithm. DCF uses two techniques: the default basic access mechanism and the request-to-send/clear-to-send (RTS/CTS) mechanism. The advantages of RTS/CTS are well recognized. The notable one is its merit in large network scenarios. With the capability to cope with hidden terminals, the RTS/CTS access scheme is widely used. Our work mainly focuses on the RTS/CTS mechanism in DCF. However, the BEB scheme suffers from the unfairness problem and yields low throughput under heavy load [3]. Analyses in [2] show that, for the DCF scheme, the throughput decreases due to the presence of collisions and backoff periods, especially in heavy load networks with many mobile users. Therefore, the elimination of such wastage should be able to improve the network throughput. Based on these ideas, we propose a queuing scheduled MAC protocol which takes advantage of the loss differentiation information from the physical layer to enhance the radio channel utilization and network fairness.

In the current standard MAC protocol, the BEB algorithm assumes that all losses are due to collisions. If a frame is lost because of a random link error instead of a collision, doubling the contention window is inappropriate and will seriously degrade performance. When a collision is detected, the BEB algorithm retransmits a packet after a period of random backoff time within an increasing contention window, therefore when different users experience different channel variations, lack of fairness is a serious issue for the BEB algorithm. In [4], the authors proposed a method to distinguish collisions and link errors in the MAC layer. For a frame loss, if the sender receives a CTS but no ACK, it deduces the loss is caused by wireless errors; if the sender receives neither a CTS nor an ACK, it deduces the loss is caused by collisions. In this paper, an effective user detection module in the physical layer is designed to feed back both the error type information and the detailed channel condition information. In [5], A MAC protocol based on the Distributed Queuing Random Access Protocol (DQRAP) is originally proposed for a Code Division Multiple Access (CDMA) mobile communication system. The method has also been applied to WLAN networks [6]. This mechanism totally changes the frame structure and the contention mechanism. It is not easy to implement in current WLAN systems and the reserved m contention periods in each frame result in additional transmission overhead. However, the queuing method gives us inspiration on transmission scheduling. Based on the above ideas, this paper proposes a PALD-DQMP protocol which has the following features:

1. It maintains the current MAC frame format, without any extra transmission overhead.
2. The physical layer assisted user detection module can provide effective loss differentiation information and users' channel state information.
3. Based on cross-layer information, PALD-DQMP utilizes a queuing system to handle the random access and data transmission, thus eliminating the bandwidth wastage caused by the BEB algorithm.
4. PALD-DQMP can effectively enhance the network performance and fairness, particularly for WLANs with heavy load.

II. USER DETECTION FOR LOSS DIFFERENTIATION

In the IEEE 802.11 standard, there are two types of WLAN topologies: the independent basic service set (IBSS), i.e., as

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an ad hoc network, and the infrastructure network. In an infrastructure network, an access point (AP) acts as a hub and connects the basic service set (BSS) network to an extended service set (ESS) network, where BSS is a set of stations controlled by a single coordination function and ESS is a set of one or more interconnected BSSs and integrated local area networks (LANs) that appears as a single BSS to the logical link control layer at any station associated with one of those BSSs. In an IBSS network, we assume that any node can be chosen as a coordinating node (CN). Even with this CN, the IBSS still works in a distributed manner because the CN only acts as a detector of the access requirement/acknowledgment and it behaves simply as a repeater in order to broadcast its information to other nodes. According to the statement of logical service interfaces in [1], the necessary information is exchanged between the AP/CN and other nodes by association related services and authentication related services at network initialization. Of this information, we are mostly interested in the ability of APs/CNs to easily gather the transmitter addresses of the active nodes associated with it.

Multiuser detection is an effective technique in CDMA systems. It can successfully separate users at the receiver based on user specific spreading codes [7]. In WLANs, there exists similar user dependent information – transmitter address (TA). Assuming users can synchronously or asynchronously send RTS frames, some multiuser detection methods can be adapted to WLAN systems, and provide further channel condition information. Based on this information, the MAC protocol can better schedule different users' transmissions, accounting for users' QoS or fairness requirements. In terms of throughput, by eliminating the collision errors and the consequent bandwidth wastes, user detection techniques can effectively improve MAC protocols. In this section, a synchronous user detection method for WLANs will be presented.

It has been mentioned above that TAs are the information available to an AP/CN after network initialization. From the definition of the RTS frame format, TA is also part of the useful information obtained from senders' RTS frames. Compared with spread spectrum codes in CDMA systems, in our detection scheme, TAs will be used to identify active users, similar to user identification in CDMA systems [7]. In this section, we firstly assume that the RTSs are transmitted synchronously, which will be presented in the next section. Let us consider a BSS system where K mobile nodes are associated with an AP/CN, so the AP/CN keeps a transmission address list of the active nodes in its BSS. A transmission address is an N-bit ($N = 48$) bit vector, so the list of TA can be recorded as the following matrix:

$$\mathbf{C} = \left[\begin{array}{ccccc} \mathbf{c}_1, & \cdots, & \mathbf{c}_k, & \cdots, & \mathbf{c}_K \end{array} \right],$$

where $\mathbf{c}_k = \left[\begin{array}{ccccc} c_{k,1}, & \cdots, & c_{k,n}, & \cdots, & c_{K,N} \end{array} \right]^T$. For these K users, let the vector $\mathbf{h}$ represent the channel gains experienced in their individual wireless channels:

$$\mathbf{h} = \left[\begin{array}{ccccc} h_1, & \cdots & , & h_k, & \cdots, & h_K \end{array} \right]^T.$$

At time t, assume that l ($0 \leq l \leq K$) users have data to transmit and send RTSs simultaneously to contend the channel. For users who have not sent RTSs, their corresponding channel gains in vector $\mathbf{h}$ will be zero. After the RTS contention procedure, the received signal is given by

$$\mathbf{r} = \mathbf{C}\mathbf{h} + \mathbf{n}, \tag{1}$$

where $\mathbf{n}$ is the noise. Consequently, according to the least-squares criterion, the users' channel gain vector $\hat{\mathbf{h}}$ is estimated at the receiver by the following formula:

$$\hat{\mathbf{h}} = (\mathbf{C}^T\mathbf{C})^{-1}\mathbf{C}^T\mathbf{r} = \mathbf{\Pi}\mathbf{r}, \tag{2}$$

where $\mathbf{\Pi} = (\mathbf{C}^T\mathbf{C})^{-1}\mathbf{C}^T$. Basically, each calculated element $\hat{h}_k$ represents the detected channel gain of each active user. It is necessary to distinguish these results into three cases:

1) silence state: the active node has no RTS requirement in the current contention round.

2) link error state: the active user takes part in the channel contention; however, its channel quality is deduced to be inadequate for transmission.

3) collision state: the active user takes part in the current contention and obtains a good channel to transmit its next frame.

In order to implement the above differentiation function, our user detection module utilizes the power value of h_k, namely $|\hat{h}_k|^2$, as the decision criterion, and then constructs two thresholds to discriminate the above three cases. For those users who have not participated in the RTS contention, their $|\hat{h}_k|^2$ values should be zero. However in the network scenario with noise present, the value of $|\hat{h}_k|^2$ could be nonzero. So, it is required to set a few thresholds for different users to distinguish the first case from the other two cases. By substituting (1) in (2) , we obtain:

$$\hat{\mathbf{h}} = \mathbf{h} + \mathbf{\Pi}\mathbf{n}. \tag{3}$$

Clearly, the power of the noise signal in the vector $\mathbf{\Pi}\mathbf{n}$ should be the silence threshold. We assume $\mathbf{n}$ is an additive White Gaussian noise vector whose element has zero mean and variance σ^2. Denote the calculated inverse matrix by $\mathbf{\Pi} = [a_{i,j}]_{K\times N}$, then we obtain the silence threshold $H_{silence-thresh}$:

$$\begin{aligned} H_{silence-thresh} &= E\{|\mathbf{\Pi}\mathbf{n}|^2\} \\ &= \sigma^2 \times \Big[\textstyle\sum_{j=1}^{N}(a_{1,j})^2, \cdots, \\ &\qquad \textstyle\sum_{j=1}^{N}(a_{k,j})^2, \cdots, \sum_{j=1}^{N}(a_{K,j})^2\Big]^T_{K\times 1}, \end{aligned}$$

where $|\cdot|$ is an operator to take the absolute value of each element in a vector. In terms of the second threshold, we denote it as $H_{collision-thresh}$ to distinguish the second case from the third case. The configuration of this parameter is related to the specified system performance, the user's QoS requirement, etc., and then plus the value of $H_{silence-thresh}$.

It should be mentioned that since $N = 48$, the maximum number of detectable users is 48. If there are more than

48 active users, we can consider adding additional address information in the RTS frame, but this will cause more overhead. Alternatively, we can distribute additional users to other APs/CNs.

III. PALD-DQMP DESCRIPTION

Based on the realization of the physical layer assisted loss differentiation scheme, the proposed MAC protocol should resolve two issues: 1) how to synchronize the required RTSs at the beginning of a contention period, but keep the random access characteristic for the nodes; 2) if the co-existing synchronized RTS signals are separated, and the channel state information is detected, how to make use of this information to schedule the data transmissions effectively? In this section, the PALD-DQMP protocol utilizes a queuing system mechanism to solve these two problems.

A logical distributed queue is constructed in PALD-DQMP. This queue is simply represented by two integers at each node, denoted pDQ and DQ. pDQ is the position of a given node in the data transmission queue. Its value ranges from 0 to DQ. The node does not have any position in the queue if it is 0 and occupies the first position of the queue if it is 1. DQ represents the total frame number of the data transmission queue. It has the same value for all active nodes, while pDQ has a specific value for each node. We assume the queue is FIFO. Both values are initially set to zero and must be updated by broadcasted RTS/ACK frames from the AP/CN. A set of algorithm rules are required for updating the two queue parameters and scheduling the subsequent frame transmissions. With this procedure, collision free data transmission can be achieved.

The following is the description of our algorithm that each node executes. It consists of two phases:

A. Channel Contention Phase

1) In the initial state, both DQ and pDQ are zero. For users who have data to transmit, their RTSs are easily transmitted synchronously after a period of DIFS. Along with this procedure, the network allocation vectors (NAVs) of the other active users are set to the longest time recorded in the “Duration/ID” field in RTS frames. This 16-bit field lies in the MAC header and contains a duration value for each frame type defined in the IEEE standard.

2) After the receiver obtains the co-existing signals, the user detection module of the AP/CN will be triggered to detect users’ channel conditions and categorize them to distinguish the silence, collision, and link error states. Then, our PALD-DQMP has two ways to feed back CTSs.

i) Just send CTSs to the users whose channels are in collision states, namely RTSs have been successfully detected and the corresponding wireless channels are good enough for frame transmission. In this step, PALD-DQMP arranges the broadcast of CTSs by the priority of their deduced power values of the channel. For AP/CN, the DQ value can be easily achieved. For other active nodes, their DQ values are updated by counting the number of received CTSs. To update pDQ, a user compares the values of the receiver address (RA) field in CTSs to its own address. If the RA value in a certain CTS is identical with its own address, its pDQ value is updated to the sequence of this CTS in the CTS list; if not, the pDQ value remains zero. In the frame control field of a MAC header, there is a one-bit “More Fragment” field, which denotes whether there is another fragment of the current MAC service data unit (MSDU) or not. PALD-DQMP utilizes this bit in a series of CTSs to inform all active users whether there is another CTS following the current CTS. If the last CTS is received, the current channel contention phase terminated, and all active users know the data transmission phase will start after a period of SIFS. In addition, the NAV will be set to the time consumed by all users to finish their frame transmissions.

ii) Let us consider link error state users. Actually they have data transmission requirements, but the wireless channel is not good enough. This is also useful information when the sender has some mechanism to enhance the channel condition, such as implementing power control. Apart from broadcasting CTSs to the collided nodes, AP/CN should also send CTSs to the link error nodes and gives them the chance to transmit within the updated channel environment. In this case, better performance should be produced.

3) In general, during the channel contention phase or data transmission phase, if some nodes have data to transmit, it will check both the values of DQ and NAV. If one or both of them are nonzero, the RTS will be deferred to the next contention period.

B. Data Transmission Phase

At the end of the channel contention phase, each node obtains its updated DQ and pDQ values. After a period of SIFS, data transmission will be scheduled by the following rules:

1) A user whose pDQ value equals 1 has the highest priority to be granted the channel for its best channel gain power. After receiving the ACK of this frame broadcasted from the AP/CN, all active nodes reduce their DQ and pDQ values by 1. Consequently the node whose pDQ value changes to 1 wins the next data transmission chance.

2) Once the last ACK which causes the values of DQ, pDQ, and NAV in all mobile nodes change to zero is received, the new round of contention phase will be triggered. After a period of DIFS, nodes which have RTS requirements will contend the channel again, then the PALD-DQMP is repeated from the channel contention phase.

IV. PERFORMANCE ANALYSIS

In this section, we use 802.11b as an example. The notations and parameter values used in this paper are given in Table I.

A. Throughput Bound Analysis

Throughput bounds of the standard MAC protocol and our PALD-DQMP protocol are compared in WLANs where the wireless channels are perfect, without fading and noise. We assume each MSDU only includes one frame.

TABLE I

NOTATIONS AND PARAMETERS VALUES.

Notations	Descriptions	Values
MAC	MAC overhead in standard	224 bits
DIFS	the period of DIFS	50 μs
SIFS	the period of DIFS	10 μs
RTS	RTS frame size	160 bits
CTS	CTS frame size	112 bits
Payload	data size	1000 bytes
ACK	ACM frame size	112 bits
CWmin	minimum value of CW	31
CWmax	maximum value of CW	1023
SoltTime	backoff slot time	20 μs
PHY	overhead at physical layer	192 bits
DataRate	physical rate for data frame	2 M
BasicRate	physical rate for control frame	1 M
t_p	propagation delay	2 μs

1) For the standard MAC protocol

With the increase of the number of mobile nodes, in the BEB algorithm, the high collision probability causes frequent startups of backoff procedures. Obviously, this will seriously waste the bandwidth and cause the unfairness problem. Therefore the throughput bound will be reached with only a pair of mobile nodes transmitting data. In this case, there is no backoff procedures, and no bandwidth wastage. So, we have:

$$T_{std} = \frac{Payload}{\frac{RTS+CTS+ACK+4PHY}{BasicRate} + \frac{MAC+Payload}{DataRate} + DIFS + 3SIFS + 4t_p}.$$

2) For the PALD-DQMP protocol

However, the throughput bound for the standard MAC protocol corresponds to the worst case for the PALD-DQMP protocol. The reason is that multiple users share the same RTS transmitting time, and their transmissions are scheduled by a logical queue, without bandwidth wastage. So, the overhead of the RTS frame will be ignored, If a large number of nodes are contending for the channel, then we have:

$$T_{new} = \frac{Payload}{\frac{CTS+ACK+4PHY}{BasicRate} + \frac{MAC+Payload}{DataRate} + DIFS + 3SIFS + 4t_p}.$$

From the fairness view point, the BEB scheme resets the contention window of a successful sender to the minimum value (CWmin), while other nodes continue to maintain larger contention windows, which reduces their chances of seizing the channel and results in channel domination by successful nodes. However in PALD-DQMP, the users' transmission requirements are fairly scheduled according to their channel conditions, as be demonstrated by our simulation results.

B. Throughput Analysis with Fading Channel

In a practical environment, packet losses are due to the low quality of wireless channels. In the standard MAC protocol, failed users will firstly generate a period of backoff time which may increase with the increase of the retry time, and then retransmit an RTS frame, until its retry time reaches the maximum, at which point this frame will be dropped. Therefore more backoff procedures will cause more bandwidth wastage.

For PALD-DQMP, even though some users' current channels are weak, the user detection module can still detect their data transmission requirements. If a user cannot combat the bad channel, it will be deferred to the next contention period, just wasting its shared RTS overhead. If in the system users have ways to combat their channels, such as increasing transmission power, their frames will be transmitted without deferring. Therefore the throughput will be close to the above analytical bound.

C. Throughput Analysis with Noise

In a noisy channel with noise, in PALD-DQMP, the user detection module relies on $H_{silence-thresh}$ to identify the active users who have RTS requirements. If the effect of the noise is small, the deduced threshold is effective. When the noise is big, we need to consider two types of errors caused by the improper threshold. One type happens to a user who has sent an RTS but perhaps through a weak channel, so the detected power of $\hat{h}_k$ is worse than the corresponding threshold. In this case the user will wait till timeout, and then attend the next channel contention round. However if the estimated power is lower than the corresponding elements in $H_{collision-thresh}$, this type of error will not affect the throughput. The other type happens to a user who has not sent an RTS, but the detection module mistakenly sends a CTS to it. In this case, the user just sends a short frame without data payload to inform other nodes to update their DQ and pDQ values. Apart from taking these actions, a coefficient can be used to adjust the value of $H_{silence-thresh}$. This is proved to be effective in our simulation.

V. SIMULATION RESULTS

This section compares the performance of the standard MAC protocol, our proposed PALD-DQMP protocol, and the enhanced PALD-DQMP (E-PALD-DQMP), in which system users are supposed to have capabilities to combat the weak channel condition by power control. For comparison, in E-PALD-DQMP, we assume that once the channel condition is better, the success of the frame transmission is guaranteed. We consider WLANs where K nodes contend to transmit with an AP/CN. Simulation parameter values are configured as in Table I. In the physical layer, we chose the Raleigh fading model with channel gain power of 1 is chosen. Channel noise is assumed to be White Gaussian with SNR equal to 20dB. In addition, we use the value of $H_{collision-shresh}$ to control the network packet loss rate (PLR). Under each PLR chosen from the set {0.01, 0.05, 0.10, 0.15, 0.20}, we test the network performance where the number of nodes is selected to be 1, 20, and 40, which is arrange on the top of PLR in the horizontal labels of each figure.

A. Throughput

Fig. 1 shows the throughput comparison of the three protocols. At a certain PLR value, the throughputs of the protocols are similar when the node number is 1. As the number of the mobile nodes increases, the performance of PALD-DQMP

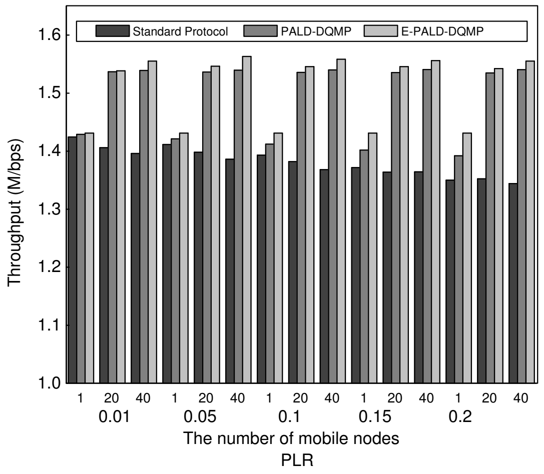

Fig. 1. Throughput comparison.

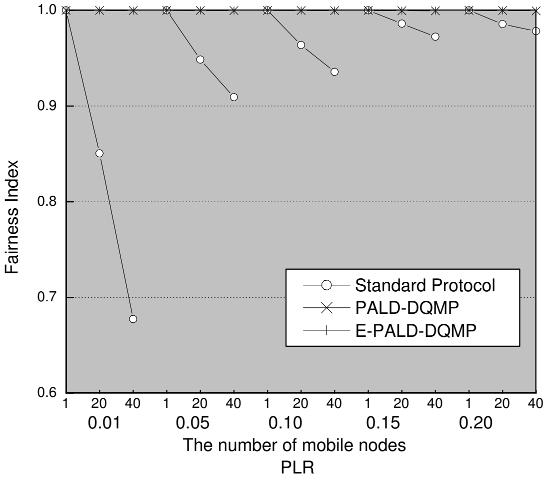

Fig. 2. Fairness comparison.

becomes better, but the performance of the standard protocol becomes worse. As the PLR increases, significant throughput improvement can be observed.

B. Fairness

The fairness index f is defined as $f = \frac{(\sum_{k=1}^{K} x_k)^2}{K(\sum_{k=1}^{K} x_k^2)}$ [8], where K is the number of active user in the network and x_k is the throughput achieved by user $k, 1 \leq k \leq K$. Fig. 2 shows the fairness index comparison. An index value of one signified fairness is achieved. It can be found that our PALD-DQMP has effectively solved the fairness problem.

C. Average Delay

The average round trip time of all users is also investigated in this paper. For the standard protocol, when multiple users contend for the channel, some users may even have no chance to seize the channel. In this case, their delay values are infinity. Actually, Fig. 3 only shows the average results of the finite delay values. The exact average values for all active users are much worse for the standard protocol.

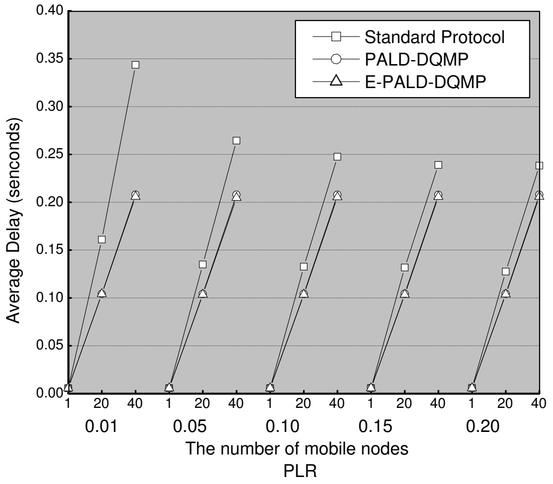

Fig. 3. Average delay comparison.

VI. CONCLUSION

A cross-layer based MAC protocol named PALD-DQMP has been proposed in this paper. Based on the synchronized RTS frames and the awareness of TAs in the MAC layer, physical layer assisted user detection can effectively detect the active users and their channel state, namly silence, collision, or link error state. Using the loss differentiation information, the queuing scheduled mechanism in the MAC layer focuses on arranging the different users' transmissions by a series of rules. This protocol eliminates the collisions and the bandwidth wastage caused by the BEB algorithm. It maintains the frame format in the standard MAC protocol, without additional overhead. Simulation results show that throughput has been enhanced, particularly as the number of active nodes increases. In addition, the unfairness problem has also been mitigated.

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