Contention-Based Airtime Usage Control in Multirate IEEE 802.11 Wireless LANs

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***Abstract*—In a multirate wireless LAN, wireless/mobile stations usually adapt their transmission rates to the channel condition. It is difficult to control each station's usage of network resources since the shared channel can be overused by low transmission-rate stations. To solve this problem, we propose a distributed control of stations' airtime usage which 1) always guarantees each station to receive a specified share of airtime, and 2) keeps service for individual stations unaffected by other stations' transmission rates. Such airtime control enables service differentiation or quality of service (QoS) support. Moreover, it can achieve a higher overall system throughput.**

The proposed airtime usage control exploits the Enhanced Distributed Channel Access (EDCA) of the IEEE 802.11e standard [1]. Two control mechanisms are proposed: one based on controlling the station's arbitration inter-frame space (AIFS) and the other based on the contention window size. We show how the stations' airtime usage is related to the AIFS and contention window size parameters. Using this relation, two analytical models are developed to determine the optimal control parameters. Unlike the other heuristic controls or analytical models, our model provides handles or parameters for quantitative control of stations' airtime usage. Our evaluation results show that a precise airtime usage control can be achieved in a multirate wireless LAN.

***Index Terms*—Medium access control, resource allocation, wireless LAN.**

I. INTRODUCTION

RESOURCE allocation in wired networks has long been the subject of research due mainly to its importance to network performance. In general, the objective of resource allocation is to allocate contending users certain system resources so as to meet their service requirements while maximizing network utilization. With effective resource allocation, one can prevent greedy or misbehaving users from exhausting network resources, and provide service differentiation or QoS support to preferred users/applications. Resource allocation can be realized by means of a scheduling algorithm with adequately assigned weights. Numerous scheduling algorithms have been proposed [3]–[5]. These scheduling algorithms, originally designed for wired networks, have been adapted to rapidly-growing wireless networks and applications. However, new issues as described below need to be resolved in order to efficiently allocate resources in wireless networks.

A. Distributed Environment

Consider an IEEE 802.11 wireless LAN [2] operating in this mode as shown in Fig. 1. In the infrastructure mode, a station can only send packets[1] to an access point (AP) which is the only station that can send packets to all other stations. For downlink (from the AP to a station) traffic, it is easy for the AP to schedule the transmission since the AP knows packet size and packet arrival time of individual traffic flows. Therefore, any desired resource allocation can be achieved by adjusting the weights of individual flows via existing scheduling algorithms. However, for uplink (from a station to the AP) traffic the AP does not have such information. Therefore, the stations need to provide the AP some queueing status or traffic information so that the AP can schedule the uplink transmission accordingly. For example, the AP at least needs to know whether or not a station has packets to send.

Most existing scheduling algorithms require detailed traffic information from stations for determination of correct transmission order [4], [5]. The self-clocked fair queueing (SCFQ) [4], originally designed for wired networks, is modified for this purpose. Although it can work in a distributed environment [6], the information required for SCFQ, such as the service tag (or the arrival time) of each packet and a "system-wide" virtual clock, is still needed and could incur extra scheduling overheads. Moreover, the delay in relaying such information could make it obsolete by the time when this information is to be used by the AP. Although more frequent transmission of scheduling information could alleviate this problem, it will incur substantial overheads. The problem becomes even worse for ad hoc networks, in which there is no central scheduler. In this type of network, each wireless station usually uses a distributed mechanism, such as carrier sense multiple access with collision avoidance (CSMA/CA), to access the channel. Therefore, it is very difficult to control a station's actual resource usage. Some distributed scheduling algorithms have been proposed to maintain fair resource usage by simplifying the random backoff mechanism of the IEEE 802.11 standard [7], [8]. As we will show later, the random backoff process in the IEEE 802.11 standard is complex, and existing

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[1]Although the terms "packet" and "frame" are used interchangeably here, one should note that a packet is usually referred to as the transmission unit at the IP layer.

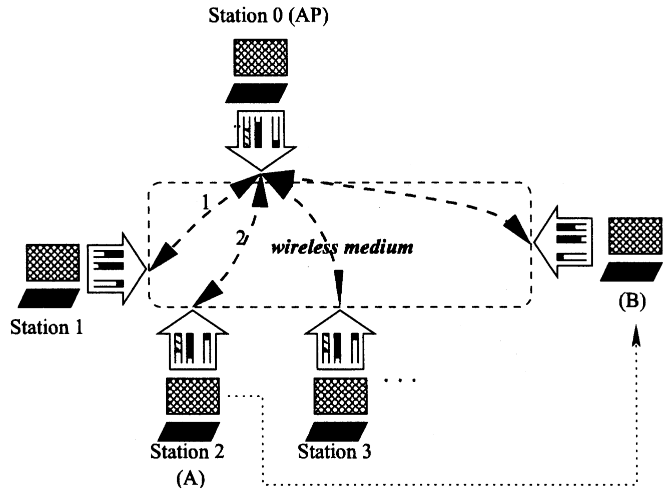

Fig. 1. Generic wireless/mobile network.

simple controls over the backoff process usually cannot achieve the desired resource allocation.

B. Location-Dependent Errors (LDE)

Another well-known property in wireless networks is the high probability of transmission errors, which is intrinsically location-dependent. That is, different wireless stations usually experience different transmission errors due to multipath fading or interference determined by their locations. For example, traffic flow 1 between stations 0 and 1 in Fig. 1 may experience much more transmission errors than traffic flow 2. The error-prone flow 1 will then end up with receiving a smaller throughput than flow 2, simply because more packets of flow 1 are corrupted during transmission. To maintain the error-prone flow's throughput, the AP has to provide more transmission opportunities in order to retransmit the corrupted packets of the error-prone flow. This results in the error-prone flow's overuse of airtime and reduces the overall system throughput. The most common solution for this problem is to defer the transmission of error-prone flows and compensate them after the channel condition improves. WPS [9], [10], CIF-Q [11], and CSDPS [12], are examples of this approach. A long-term fairness server was proposed so that the impact of a compensation mechanism on error-free flows can be reduced [13]. The concept of adaptive weights was also used such that a scheduler can dynamically adjust the weights of error-prone flows to compensate for their throughput loss. The *power factor* [14] and *compensation index* [15] are the main control parameters to adjust the weights so that compensation can be made without degrading error-free flows too much.

C. Location-Dependent Transmission Rate (LDR)

Many existing/emerging wireless networks, such as the General Packet Radio Service (GPRS) and the IEEE 802.11 standard family, support more than one physical transmission rate. Consider again the IEEE 802.11 standard as an example. An 802.11b wireless LAN can support 11, 5.5, 2, and 1 Mb/s, while an 802.11a wireless LAN can support up to eight different rates. Depending on their channel conditions, stations may choose different transmission rates (i.e., so-called "link adaptation" [16]) in order to increase the probability of successful transmission. As shown in Fig. 1, for example, station 2 moving from point A to point B may decrease its transmission rate from 11 Mb/s to 1 Mb/s in order to maintain reliable transmission. Because of the egalitarian channel access mechanism in the IEEE 802.11 a, b, or g standard, station 2 still has the same chance as others to access the channel regardless of its transmission rate. However, due to its low transmission rate, station 2 will use more airtime to transmit a single packet than other stations during each possession of the wireless channel. That is, the existing Distributed Coordinate Function (DCF) in 802.11 does not have an effective control over a station's airtime usage. For the purpose of resource allocation, we should directly control the stations' airtime usage in order to: 1) prevent low-transmission rate stations from exhausting system resources and 2) maintain the QoS support for stations that maintain their transmission rates. In [22], an opportunistic auto rate (OAR) protocol is proposed to increase the system throughput while maintaining an equal share of system airtime in a multirate IEEE 802.11b wireless LAN.

Even though the causes of LDE and LDR are correlated, the resulting problems and the corresponding solutions are quite different. The problems caused by LDE results from the fact that some stations experience significantly more transmission errors than others. Since a station in a unirate wireless network cannot adjust its physical transmission rate to improve the transmission, a "defer-and-compensate" strategy seems to be a good approach to the LDE-induced unfairness problem. In a multirate network, LDE can be eliminated, or at least alleviated, if the stations adopt link adaption as mentioned above. This way, the effects of LDE on stations' airtime usage become less significant but the resource misuse caused by LDR becomes a problem. If this problem can be solved with an appropriate airtime usage control, one can then solve the LDE-induced problem by combining airtime control and link adaption.

In this paper, we first justify the need to control a station's airtime usage in a multirate wireless LAN, and discuss its importance to service differentiation and QoS support. A contention-based control mechanism that exploits the Enhanced Distributed Channel Access (EDCA) of the IEEE 802.11e standard, is proposed to allocate airtime to stations. It has been shown in [17], [18] that service differentiation is achievable by manipulating the EDCA parameters but algorithms of how to determine these parameters were not provided. The existing analytical models of the DCF mode may be used for this purpose. A Markovian model that takes into account both AIFS and contention window sizes of the backoff process can be found in [19]. The problem with this model is the scalability of the three-dimensional Markovian chain. Even though it is possible to obtain the station throughput by an iterative algorithm from the given EDCA parameters, the inverse problem is much more difficult and computationally expensive to solve. The authors of [20] proposed a lightweight Markovian model based on [21], but neither of them considered the reset mechanism of contention window size as mandated in the IEEE 802.11 DCF and 802.11e EDCA. OAR protocol was also proposed to maintain egalitarian airtime usage in a multirate wireless network [22]. Instead of controlling the stations' backoff process, OAR controls the "More Fragment" bit in the IEEE 802.11 MAC header as was done in [17]. Some weighted temporal fairness schemes were also proposed for cellular wireless networks [23]. Nevertheless, their algorithm relies on a *centralized* scheduler for every transmission attempt according to stations' channel conditions. It cannot be used in the DCF/EDCA mode or ad hoc mode of the IEEE 802.11 wireless LAN. Moreover, their scheme requires estimation of stations' status in order to schedule a station which has the best channel condition. In the scheme proposed here, the only information needed—the physical transmission rates of individual stations—can be found in the preamble of the frame header. It will not incur additional control overhead or introduce control errors due to inaccurate estimation.

The rest of this paper is organized as follows. In Section II, we justify the use of airtime control in a multirate wireless LAN. Section III presents two analytical models for controlling interframe space and backoff parameters to allocate proportional airtime to stations in a distributed manner. Section IV discusses the numerical and simulation results. Finally, conclusions are presented in Section V.

II. Airtime Control in Multirate Wireless Networks

Let $S_i(t_1,t_2)$ be the amount of resources station i receives in a time interval (t_1,t_2), and ϕ_i be its assigned share. The objective of resource allocation is to ensure that stations acquire their assigned shares of resources

$$\frac{S_i(t_1,t_2)}{S_j(t_1,t_2)} \geq \frac{\phi_i}{\phi_j} \tag{1}$$

if station i is continuously backlogged during (t_1,t_2). We assume that ϕ_i is determined by the admission control in order to comply with station i's QoS demand. Therefore, it is a given parameter to our control algorithm.

There are different metrics one can use to measure a station' resource usage S_i in (1). For example, if it is measured by the amount of transmitted traffic, (1) will be the weighted throughput or bandwidth allocation problem since two continuously backlogged stations will have the throughputs proportional to their shares. If it is measured by the amount of airtime each station uses, (1) will be the weighted *temporal fairness* problem or referred to as the *airtime allocation* problem in this paper. Let $B_i(t_1,t_2)$ be station i's average throughput over the time interval (t_1,t_2) and $T_i(t_1,t_2)$ the airtime station i obtains during the same time interval. We will have

$$\frac{B_i(t_1,t_2)}{B_j(t_1,t_2)} = \frac{r_i \cdot T_i(t_1,t_2)}{r_j \cdot T_j(t_1,t_2)} \tag{2}$$

where r_i is the physical transmission rate of station i. If all stations use the same transmission rate, stations with the same throughput will use the same amount of airtime. That is, there is no difference between controlling stations' throughput and controlling their airtime usage in a unirate network, from the perspective of resource allocation. On the other hand, these two measures will yield different results if the station uses different transmission rates such as in a multirate IEEE 802.11 wireless LAN.

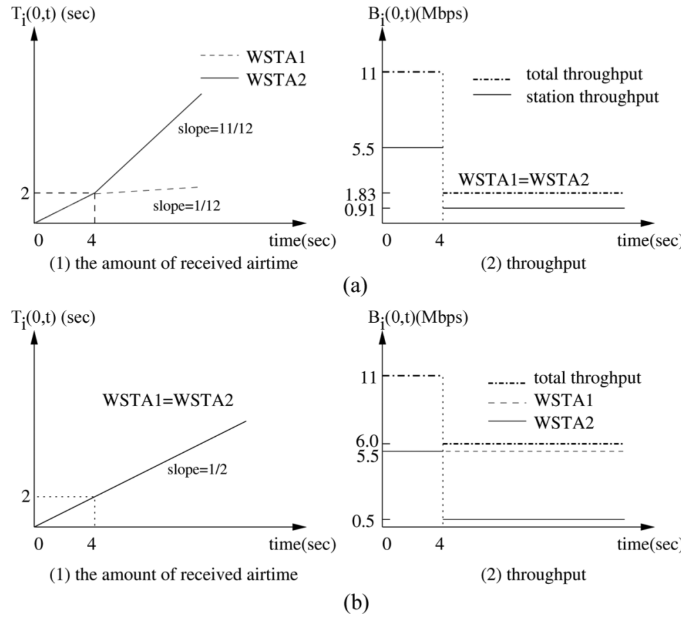

Fig. 2. Throughput fairness versus airtime fairness. (a) No airtime usage control. (b) With airtime usage control.

Let us consider a simple case of two stations, STA 1 and STA 2, exchanging frames with the AP at 11, 5.5, 2 and 1 Mb/s in a multirate IEEE 802.11b wireless LAN. Based on the 802.11 DCF, each station statistically has an equal opportunity to access the channel. Whenever a station acquires the channel, it is allowed to transmit a frame before releasing the channel for re-contention. If both stations' frame sizes are the same, both stations will have an equal share of the system throughput, regardless of their transmission rate. Now, assume that both STA 1 and STA 2 transmit at 11 Mb/s before $t = 4$. After $t = 4$, STA 2 moves from point A to point B as shown in Fig. 1, and adjusts its transmission rate to 1 Mb/s via link adaptation. As shown in Fig. 2(a)-2, the throughputs of STA 1 and STA 2 always remain the same because of the DCF operation. Nevertheless, STA 2 uses much more airtime after $t \geq 4$ as shown in Fig. 2(a)-1 because it takes much longer to transmit a frame at a lower transmission rate. Since the total system airtime is limited, using more airtime by STA 2 means degradation of STA 1's throughput and the overall system throughput. If STA 1 has ongoing bandwidth-demanding traffic, the QoS for this traffic may no longer be guaranteed. It is evident that a direct control over stations' airtime usage is crucial in a multirate wireless network.

Fig. 2(b) shows the case where the stations' airtime usage is controlled. Irrespective of the transmission rate used by STA 2, the airtime each station receives is kept the same as others', as shown in Fig. 2(b)-1. For $t < 4$, the resulting throughputs for both stations are identical since both stations use the same transmission rate. For $t > 4$, the results are quite different:

1) Without the airtime usage control, the amount of traffic transmitted by STA 1 varies with the transmission rate of STA 2, and so does STA 1's throughput. However, as shown in Fig. 2(b)-2, the throughput of STA 1 remains unaffected in the case when the airtime usage can be controlled. Because of this fixed throughput, the QoS for STA 1's traffic can be maintained regardless of STA 2's

TABLE I
COMPARISON OF THROUGHPUT AND AIRTIME FAIRNESS

	resource measure	
	throughput	airtime
Throughput fairness	YES	NO
Airtime fairness	NO	YES
Differentiated Service	YES	YES
Isolated QoS support	NO	YES
System throughput	Lower	Higher

transmission rate. Of course, STA 2's throughput is lower than that in the case without airtime control after $t = 4$. This is unavoidable for the "damage control." Instead of compromising all stations' performance, maintaining airtime fairness makes individual stations' service independent of each other, and forces the station which lowers its transmission rate to absorb the throughput reduction.

2) Because of (1), the overall system throughput achievable by controlling airtime usage, 6 Mb/s, will be much higher than in the case without airtime control, which is 1.83 Mb/s.[2] This is an additional benefit of controlling airtime usage in a multirate wireless LAN.

Controlling stations' airtime usage is very important to service differentiation and QoS support in a multirate wireless LAN. For service differentiation, we can allocate more airtime to preferred stations by assigning larger weights. For QoS support, we can guarantee that the QoS for stations that maintain their transmission rates are not affected by other stations' channel condition. That is, we can provide independent or isolated QoS support via airtime usage control. Of course, the station that lowers its transmission rate may suffer a throughput reduction due to airtime usage control, thus compromising the QoS of its applications. This problem can be solved in two ways: (1) the station may dynamically adjust the weights of its traffic flows so that the reduction of its throughput will not affect the traffic with strict bandwidth requirements but will affect only best-effort traffic [14]; (2) graceful degradation can be applied to some applications so that the resultant quality is still within their acceptable range, if these applications are QoS-degradable [24]. The comparison of performance with and without airtime usage control is summarized in Table I.

Airtime usage control would not be an interesting problem if a centralized scheduler were available to schedule transmissions. In fact, any existing scheduling algorithm can be adapted to control stations' airtime usage as long as there is a centralized scheduler in the wireless network. Let F be such a scheduling algorithm which controls the stations' access to the channel. In order to apply F for airtime usage control, the only necessary modification needed by the scheduler is to adapt a station's weight to its transmission rate as

$$\phi_i' = \frac{r_i}{\min(r_j)} \cdot \phi_i. \tag{3}$$

[2]Here, we do not consider any overhead or collision for an illustrative purpose.

That is, the scheduler should reduce the weight of a station (i.e., polling that station less frequently) proportionally if the station lowers its transmission rate. Even though that station will use more airtime to transmit a frame for each channel access, the scheduler will reduce its access to the channel because of the reduced weight. This way, the station's actual airtime usage can be maintained regardless of its transmission rate. Such adaptation can be applied to the Point Coordinate Function (PCF) or Hybrid Coordination Function (HCF) mode in the IEEE 802.11 wireless LANs since the AP takes charge of scheduling both uplink and downlink transmissions. In the DCF or EDCA mode of an infrastructure IEEE 802.11 wireless LAN or an ad hoc wireless LAN, such centralized scheduling is infeasible since each station relies on contention to access the channel. Therefore, we focus on the contention-based mechanism to control a station's airtime usage.

III. CONTENTION-BASED AIRTIME USAGE CONTROL

As mentioned earlier, stations in an IEEE 802.11 wireless LAN contend for the wireless channel using CSMA/CA and random backoff. Because all stations use the same mechanism to access the channel, they acquire the wireless channel with an equal probability. In order to control a station's airtime usage, we can either 1) control the transmission time a station can use during its acquisition of the channel, or 2) control the contention process so that stations can access the channel with different probabilities. To achieve 1), we can set the "More Fragment" bit in the frame header so that more than one frame can be transmitted. By controlling the number of frames to be transmitted, the station's actual airtime usage can be controlled [17], [22]. To achieve 2), stations must use different parameters for contention as described below. In what follows, we use 2) to control the stations' airtime usage since the IEEE 802.11e standard allows stations to use different contention parameters. We will focus on determining the parameters that each station should use in order to obtain its target share of the airtime. For completeness, we first give a brief review of the contention mode in IEEE 802.11e standard.

A. The Contention Mode IEEE 802.11(e) Standards

The contention mode in IEEE 802.11e standard is based on the "listen-before-talk" or CSMA/CA with exponential random backoff. Based on this mechanism, a station desiring to initiate the transmission of a frame listens to the channel to determine whether the channel is idle or busy. If the channel is determined to be idle,[3] the station will decrement its backoff time for every slot time. If the channel becomes busy during a backoff process, the backoff is suspended. When the channel becomes idle again, and stays idle for an extra DIFS/AIFS time interval, the backoff process resumes from the point where it was suspended. A station can access the channel when its backoff time is decremented to zero.

In an IEEE 802.11e wireless LAN, stations may have different contention parameters, denoted as $CW_{\min}[i]$ and

[3]This is assessed if there is no transmission from any other station for a duration called as Distributed Inter Frame Space (DIFS) time in the IEEE 802.11 standard and Arbitration Inter Frame space (AIFS) time in the IEEE 802.11e standard.

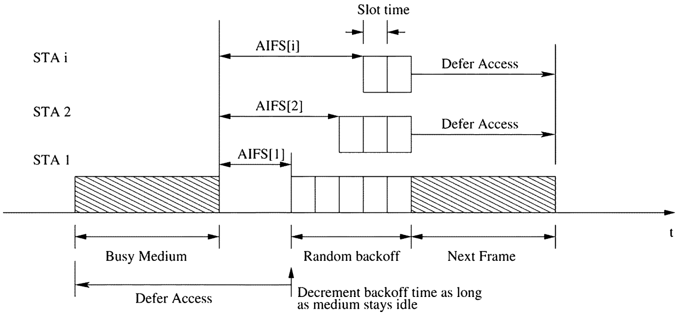

Fig. 3. Distributed channel access in an IEEE 802.11 wireless LAN.

$CW_{\max}[i]$, to determine the contention window size $CW[i]$ (which is initially set as $CW_{\min}[i]$). Based on its $CW[i]$ value and transmission results, a station can then determine the backoff time. The backoff time is chosen as a random integer drawn from a uniform distribution over the interval $[1, CW[i]]$. For each successful reception of a frame, the receiving station immediately acknowledges the frame reception by sending an acknowledgment (ACK) frame. If an ACK frame is not received after the data transmission, the transmission is considered failed and the frame is retransmitted after another random backoff. The new backoff time is obtained using a new $CW[i]$ value computed by

$$CW[i] \Leftarrow 2 \cdot (CW[i] + 1) - 1$$

until $CW[i]$ reaches its maximum, $CW_{\max}[i]$ The exponential increase of $CW[i]$ reduces the collision probability in case there are multiple stations attempting to access the wireless channel.

After each successful transmission, $CW[i]$ is reset to $CW_{\min}[i]$. The station completing its transmission then performs a new random backoff even if there are no other pending frames in the queue. This is often referred to as "post" backoff, as this backoff is done after, not before, a successful transmission. This post backoff ensures there is at least one backoff interval between two consecutive transmissions. The $CW[i]$ value is also reset when the retransmission limit is reached. All of the parameters including AIFS, slot time, $CW_{\min}$, and $CW_{\max}$ are dependent on the underlying physical layer (PHY). Fig. 3 shows the timing relation of stations' AIFSs and backoff times in the EDCA of the IEEE 802.11e standard.

B. Basic Control Equation

By controlling the values of AIFS$[i]$, $CW_{\min}[i]$ and $CW_{\max}[i]$ used by each station, stations may have different opportunities to access the wireless channel, and thus obtain different amounts of airtime. However, how to choose these parameters in order to satisfy stations' requirements is never an easy task. Let us consider two stations, STA 1 and STA 2, each using a different AIFS, $CW_{\min}$ and $CW_{\max}$. As shown in Fig. 4, we assume that both STA 1 and STA 2 choose an initial backoff time of $5 \cdot aSlotTimes$ (i.e., $BT_1^{(i)} = 5$ for $i = 1$ and 2). Since STA 1 uses a smaller AIFS, STA 1 decrements its backoff time to zero before STA 2 does. Therefore, STA 1 is allowed to transmit while STA 2 suspends its backoff process during STA 1's transmission. At the end of STA 1's

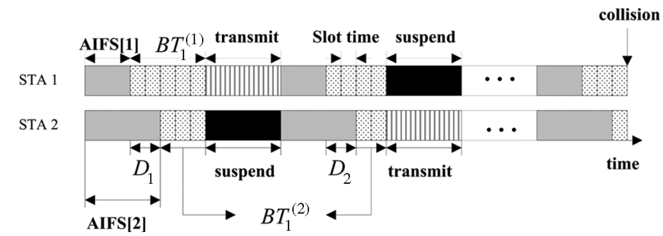

Fig. 4. Stations' random backoff times between collisions.

transmission, both stations wait for $AIFS[1]$ and $AIFS[2]$, respectively, before starting the backoff process again. Because the remaining backoff time of STA 2 (i.e., 2) is smaller than the second backoff time chosen by STA 1, STA 2 will win the contention and transmit a frame while STA 1 will suspend its backoff process. By excluding the transmission and suspension time shown in Fig. 4, a relation between stations' backoff times can be obtained as

$$\sum_{i=1}^{n_1} BT_i^{(1)} = \sum_{j=1}^{n_2} BT_j^{(2)} + \sum_{h=1}^{n_1+n_2-1} D_h \tag{4}$$

if we consider any time interval between collisions. Here, n_j is the total number of times that STA j draws a new backoff time during the observed interval, and $BT_i^{(j)}$ is the ith backoff time $(i \le n_j)$ chosen by STA j. D_h is referred to as the hth "decrementing lag" of STA 2 because STA 2 waits longer than STA 1 each time before starting its backoff.

Since a station draws a new backoff time only at the end of each transmission,[4] n_i also represents the number of a station's accesses to the wireless channel. Therefore, n_i also represents the airtime usage of each station. The value of random backoff time BT_i in (4) is determined by the station's contention window parameters, $CW_{\min}$, $CW_{\max}$ and $retry\ limit$, while the decrementing lag is mainly decided by the AIFS value. Thus, (4) provides the relation between airtime usage and control parameters $CW_{\min}$, $CW_{\max}$ and AIFS. Based on this relation, we can choose appropriate parameters in order to control a station's airtime usage.

C. Controlling AIFS Time

According to (4), one way to control stations' airtime usage is to control the decrementing lag. Unfortunately, we do not have direct control over the decrementing lag since it is determined by the AIFS values used by the stations. Let us consider the case that STA 1 has a smaller AIFS than STA 2 and $AIFS[2] - AIFS[1] = 2 \cdot aSlotTimes$. Every time STA 2 starts to decrement its backoff time, STA 1 has already decremented its backoff time by $2 \cdot aSlotTimes$. One may mistakenly think that $D_h = AIFS[2] - AIFS[1]$ and conclude that D_h is then a constant. In fact, D_h is a random variable and could be any integer values between 1 and $AIFS[2] - AIFS[1]$, depending on the backoff time chosen by the stations. For example, if STA 1 chooses a backoff time of $1 \cdot aSlotTime$, STA 2 will not have any chance to start its backoff given $AIFS[2] - AIFS[1] = 2 \cdot aSlotTimes$. In this case, the decrementing lag

[4]It can be a collision.

at this round of contention is 1. Therefore, we will need a relation between stations' AIFS values and the decrementing lag in order to use AIFS for airtime usage control.

Let us revisit the previous example but assume $AIFS[2] - AIFS[1] = d \cdot aSlotTimes$. We have to consider two cases as shown in Fig. 5.

1) **Case (a)**: STA 1 chooses a backoff time $BT_1^{(i)}$ that falls between 1 and $d-1$ as shown in Fig. 5(a). Therefore, the current decrementing lag of STA 2, D, will be equal to $BT_1^{(i)}$ because STA 1 has decremented its backoff time by $BT_1^{(i)} = D \cdot aSlotTimes$ (to zero) before STA 2 starts its backoff. In fact, in this case STA 2 has no chance at all to start decrementing its backoff until the next round of contention. The conditional mean of D given that this case occurs can be calculated as

$$E[D|Case\ (a)] = \frac{\sum_{i=1}^{d-1} i}{d-1}. \tag{5}$$

2) **Case (b)**: STA 1 chooses a backoff time $BT_1^{(i)} \geq d$ as illustrated in Fig. 5(b). In this case, both STAs will perform their backoffs and either STA 1 or STA 2 will win the contention depending on their backoff times. The computation of decrementing lag becomes more complicated, and one has to consider the following two subcases:
   - Case (b)-1: If $BT_1^{(i)} - d \leq BT_2^{(j)}$, STA 1 will win the current round of contention. In this case, we have $D = d$.
   - Case (b)-2: If $BT_1^{(i)} - d > BT_2^{(j)}$, STA 2 will win the current round of contention. Given that STA 1 has a nonzero backoff time, it is possible that this remaining backoff time results in another decrementing lag, denoted as D', in the next round of contention (from STA 2's perspective). If the remaining backoff time of STA 1 $BT_1^{(i)} - d - BT_2^{(j)}$ falls between 1 and $d-1$, we will have $D' < d$ in the next round of contention.

   By combining Case (b)-1 and Case (b)-2, we can obtain the conditional mean of D (given that Case (b) occurs) by

$$E[D|Case\ (b)] = \frac{CW-(d-1)}{CW} \cdot d + \frac{\sum_{i=1}^{(d-1)} i}{CW} \tag{6}$$

   since the stations choose their own backoff times uniformly within their contention windows. The first term represents Case b-(1) which occurs with a probability of $CW - (d - 1)/CW$. The second term represents Case (b)-2 where the remaining backoff time of STA 1 could be any integer between 1 and $d-1$, each with a probability of $1/CW$. Here, we assume that if STA 1 "loses" the current round of channel contention (i.e., $BT_1^{(i)} > BT_2^{(j)} + d$), it will "win" the next round (i.e., $BT_1^{(i)} < BT_2^{(j)} + BT_2^{(j+1)} + 2d$), given that the probability that a station wins in consecutive contention rounds is relatively small.

Finally, we can obtain the mean of D by combining Case (a) and Case (b) as

$$E[D] = \frac{d-1}{CW} E[D|Case\ (a)] + \frac{CW-(d-1)}{CW} E[D|Case\ (b)] \tag{7}$$

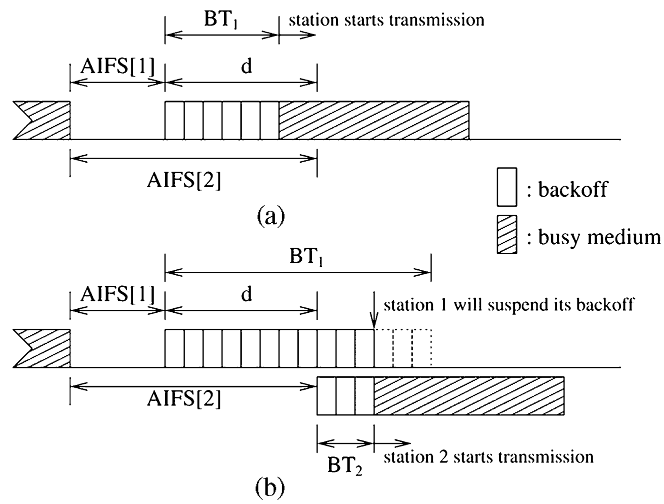

Fig. 5. Decrementing lag of STA 2.

where $E[D|Case\ (a)]$ and $E[D|Case\ (b)]$ are given in (5) and (6). The equation can be further simplified as

$$E[D] \approx d - \left[\frac{d(d-1)}{CW_{\min}} - \frac{d(d-1)^2}{2CW_{\min}^2}\right] \tag{8}$$

if we assume that both STAs use the same values of $CW_{\min}$ and $CW_{\max}$, and the collision probability is small. It should be noted that more precise calculation can be done by considering other events, and introducing more higher order terms in (8). Our simulation results in the next section show that a very good estimate of D can be obtained without considering those higher order terms.

Equation (8) shows some interesting relation between contention window size and the decrementing lag. First, if we choose a very large $CW_{\min}$, STA 2's decrementing lag should be very close to $AIFS[2] - AIFS[1] = d$ since it is very unlikely for STA 1 to choose a backoff time less than d. This can be observed in (8). Second, the term inside the square brackets of (8) represents the contribution of STA 1's backoff process to STA 2's decrementing lag. If there are more than one station with a smaller $AIFS(= AIFS[1])$, STA 2's decrementing lag should be smaller because it is more likely for at least one of those stations to choose a random backoff time smaller than d. In fact, STA 2's average decrementing lag can be calculated as above by using the concept of union bound [25]

$$E[D] \approx d - \left[\frac{d(d-1)}{CW_{\min}} - \frac{d(d-1)^2}{2CW_{\min}^2}\right] * K_1 \tag{9}$$

where K_1 is the number of stations using $AIFS = AIFS[1]$.

Finally, it should be noted that the number of stations using the same or larger AIFSs will not affect a station's decrementing lag. This is because stations with larger AIFSs can only start decrementing their backoff times after stations with smaller AIFSs. With this property, we can generalize (9) for a station using $AIFS = AIFS[k]$ as

$$E[D^{(k)}] = d_1^{(k)} - \sum_{i=1}^{k-1} \left[\frac{d_i^{(k)}\left(d_i^{(k)}-1\right)}{CW_{\min}} - \frac{d_i^{(k)}\left(d_i^{(k)}-1\right)^2}{2CW_{\min}^2}\right] * K_i \tag{10}$$

```
step 1 Let d_i^(k) = d_1^(k) - d_1^(i)
step 2 Calculate E[n_k]/E[n_1] from system equations of Eq. (12) using Eq. (10)
       if |E[n_1]/E[n_k] - φ'_1/φ'_k| ≤ error
          AIFS[k] = AIFS[1] + d_1^(k);
          go to step 3
       else
          if E[n_1]/E[n_k] < φ'_1/φ'_k
             d_1^(k) = d_1^(k) + 1;
          else
             d_1^(k) = d_1^(k) - 1;
          go to step 1
step 3 Stop
```

Fig. 6. Pseudo-code for calculating AIFS values.

where K_i is the number of stations using $AIFS = AIFS[k]$, and $d_i^{(k)} = AIFS[k] - AIFS[i]$ for $i = 1$ to $k-1$. Even though the derivation of (8) introduces an estimation error to (10), due to the use of union bound, we will show later that it matches the simulation results very well.

With the decrementing lag being derived, we now show how to control stations' airtime usage via manipulating stations' AIFS values. Let K_i be the number of stations that require an airtime usage ratio ϕ'_i. Obviously, stations requiring the same airtime usage should use the same parameters. We assume that $\phi'_i > \phi'_j$ if $i < j$ so that the station with a ratio of ϕ'_1 use the smallest AIFS value (i.e., AIFS[1]), the station with a ratio of ϕ'_2 has the second smallest AIFS value (i.e., AIFS[2]), and so on. In this general case, (4) can be rewritten as

$$E[n_1]E[BT^{(1)}] = \left(\sum_{j=1}^{N} K_j E[n_j] - E[N_{col}]\right) \times E\left[D^{(k)}\right] + E[n_k]E\left[BT^{(k)}\right] \quad (11)$$

for $k = 2$ to N in the steady state. Here, $E[D^{(k)}]$ is the average "decrementing lag" of stations with an airtime usage ratio of ϕ_k, with respect to the stations with the smallest AIFS, and $E[N_{col}]$ is the average number of collisions within any observed time interval. In order to emphasize the effects of AIFS values on the stations' airtime usage, we further assume that all stations use the same $CW_{\min}$ and $CW_{\max}$. The effect of these two parameters on stations' airtime usage will be thoroughly investigated in the next subsection. Under this assumption, (11) can be further rewritten as

$$E[n_1]\frac{CW_{\min}}{2} \approx \sum_{j=1}^{N} K_j E[n_j] E\left[D^{(k)}\right] + E[n_k]\frac{CW_{\min}^2}{2}. \quad (12)$$

Here, we simply substitute $CW_{\min}/2$ for $E[BT^{(i)}]$ and assume $\sum_{j=1}^{N} K_j E[n_j] \gg E[N_{col}]$. This is reasonable because the random backoff process is designed to minimize (especially, consecutive) collisions. The probability that a station collides with others more than twice in a row is very small. Later, we will show how to calculate $E[BT]$ when consecutive collisions occur.

By replacing $E[D^{(k)}]$ in (12) with (10) and solving the system of $N-1$ equations given by (12), we can obtain $d_i^{(N)}$ (for $i = 1$ to $N-1$) for any given $\phi'_1/\phi'_k = E[n_1]/E[n_k]$.[5] Once $d_1^{(N)}$ is obtained, the required AIFS values can be calculated using $AIFS[i] = AIFS[N] - d_1^{(N)}$. The only problem here is that the resulting equations is a system of discrete nonlinear equations[6] and solving such a system equation is never an easy task. Therefore, we propose an alternative algorithm shown in Fig. 6 to calculate AIFSs more efficiently.

The initial value for $d_1^{(k)}$ in step 1 is obtained by solving the system equations given by (12), using $E[D^{(k)}] = d_1^{(k)}$ and $E[n_1]/E[n_k] = \phi'_1/\phi'_k$. Given that the value of $CW_{\min}$ is much larger than $d_1^{(k)}$ and K_i is a not-too-much-larger number in typical cases, using $E[D^{(k)}] = d_1^{(k)}$ is a good approximation and gives a good initial solution of $d_1^{(k)}$'s. Note that the resulting equations are linear equations of $d_1^{(k)}$, in contrast to nonlinear equations if (10) were used directly. Therefore, the computation complexity is reduced. With the help of this initial solution and the proposed algorithm, we can obtain more accurate solutions of $d_1^{(k)}$'s and calculate the AIFS values much faster.

D. Controlling $CW_{\min}$ and $CW_{\max}$

As suggested in (4), one can also control stations' airtime usage by manipulating backoff parameters $CW_{\min}$ and $CW_{\max}$. In this subsection, we assume that all stations use the same AIFS value but different backoff parameters to contend for channel accesses. Equation (4) can then be rewritten as

$$\sum_{i}^{n_1} BT_i^{(1)} = \sum_{j}^{n_2} BT_j^{(2)}. \quad (13)$$

By taking the expected values of both sides in this equation, we have

$$\frac{E[n_1]}{E[n_2]} \approx \frac{CW_{\min}[2]}{CW_{\min}[1]}. \quad (14)$$

Again, we use $CW_{\min}^{(1)}/2$ as STA 1's expected backoff time as we did in the previous subsection. Equation (14) shows that the airtime usage ($\propto n_i$) of a station is approximately inversely proportional to its minimum contention window size. This property provides us an easy way to control each station's share of airtime in a distributed manner. A similar relation can also be found in [8], but the exponential increment of contention window and the reset mechanism of contention window were not considered. In fact, the random backoff process is far more complicated because the contention window size needs to be adjusted, depending on the outcome of each transmission attempt. Even though one can expect that the mean value of a station's random backoff time is close to $CW_{\min}/2$ because of the small collision probability, precise control over station's airtime usage cannot be achieved without including the exponential increment and reset mechanism of stations' contention window sizes, especially when the number of stations in a wireless LAN is large.

[5]Note that there are only $N-1$ independent variables in this system of equations since any $d_n^{(m)}$ in (10) can be represented by $d_m^{(N)} - d_n^{(N)}$.

[6]The AIFS values must be a multiple of $aSlotTimes$.

In order to accurately analyze each station's share of airtime, we propose an enhanced model based on a previous DCF model [20], [21]. The station's backoff process is observed whenever at least one station's backoff time changes in the wireless LAN, i.e., at the end of an idle slot or at the end of a transmission/collision. Each station's backoff process is represented by a state vector, $(w_i(t), b_i(t))$, at these particular observation points. Here, $w_i(t)$ represents the contention window index of station i and $b_i(t)$ is station i's backoff time, in units of $aSlotTimes$. The resulting process, $\{(w_i(t), b_i(t)) : t = t_1, t_2, \cdots\}$ for station i can then be modeled as a two-dimensional discrete-time Markov chain as suggested in [21]. However, our Markovian model differs from their models as follows.

1) When a station has a nonzero backoff time, it should not decrement its backoff time until the channel has been idle for one slot time. If some stations decrement their backoff times to zero at one observation point, the backoff time of the other stations should remain unchanged at the next observation point which is the end of current transmission. Therefore, the transition probability from $(w(t_j) = w, b(t_j) = i)$ to $(w(t_{j+1}) = w, b(t_{j+1}) = i-1)$ for a station with a nonzero backoff time is less than 1. The probability should be computed as

$$\begin{aligned} P\left[w(t_{j+1}) = w, b(t_{j+1}) = i-1 | w(t_j) = w, b(t_j) = i\right] \\ = P\left[b_k(t_j) > 0 \text{ for all } k \neq i\right]. \quad (15) \end{aligned}$$

This key property in the random backoff has been overlooked in [21], and the above transition probability was assumed to be 1 there.

2) The reset mechanism of the contention window size, after the number of retransmissions reaches the retry limit, is also included. The retry limit plays an important role in the contention but was not considered in [20] and [21]. A small retry limit can effectively reduce a station's average backoff time because the contention window size gets reset after a few retransmission attempts. On the other hand, a large retry limit determines how many times a station can use $CW_{\max}$ before resetting it. When the channel condition is poor, a station with a smaller retry limit is then more likely to acquire the access to the channel than a station with a larger retry limit.

3) Each station may use different backoff parameters: $CW_{\min}[i]$, $CW_{\max}[i]$, and $retry\ limit$. In the IEEE 802.11e standard, these values depend on the priority level of a station/application, and our model can handle this general case.

4) The transmission error is included in our model. Moreover, each station may have different transmission-error probabilities (i.e., location-dependent error).

We now consider a tagged station i with the backoff parameters, $CW_{\min}[i] = W_0^{(i)} - 1$, $CW_{\max}[i] = 2^{m^{(i)}} \cdot W_0^{(i)} - 1$, and $retry\ limit = n^{(i)}$. $W_0^{(i)}$ is usually chosen as a power of 2, and $m^{(i)}$ determines how much a contention window can be increased. Obviously, $1 \leq w_i(t) \leq m^{(i)}$. If $n^{(i)} \geq m^{(i)}$, the contention window index may remain unchanged for $(n^{(i)} - m^{(i)})$ times after it reaches $m^{(i)}$, because station i will use $CW_{\max}[i]$ as the contention window to retransmit a frame up to $(n^{(i)} - m^{(i)})$ times. Otherwise, the contention window of station i will never reach $CW_{\max}[i]$; instead, the maximum value it can reach is $2^{n^{(i)}} \cdot W_0^{(i)} - 1$, and will be reset thereafter regardless whether the transmission succeeds or fails. The transition probabilities of the Markov chain can then be computed as follows.

1) After a successful transmission, station i will reset its contention window, and choose a new backoff time:

$$P[0,k|j,0] = \frac{1-p_f^{(i)}}{W_0^{(i)}},\ \forall\ j \neq M^{(i)}$$

for $0 \leq k \leq W_0^{(i)} - 1$. Here, $p_f^{(i)} = 1 - (1-p_c^{(i)})(1-p_e^{(i)})$ is the probability of transmission error, $p_e^{(i)}$ the frame} error rate due to the wireless medium, and $p_c^{(i)}$ the collision probability of station i, and $M^{(i)} = \min(n^{(i)}, m^{(i)})$ is the maximum contention window index.

2) After an unsuccessful transmission attempt, station i will use the next contention window $W_{j+1}^{(i)} = 2 \cdot W_j^{(i)}$, and choose a new backoff time:

$$P[j+1,k|j,0] = \frac{p_f^{(i)}}{W_{j+1}^{(i)}},\ \forall\ j \neq M^{(i)}$$

for $0 \leq k \leq W_{j+1}^{(i)} - 1$.

3) Station i will reset its contention window after the number of retransmissions for a frame reaches n_i, and will randomly choose a new backoff time:

$$P[0,k|M^{(i)},0] = \frac{1}{W_0^{(i)}},\ 0 \leq k \leq W_0^{(i)} - 1.$$

4) Station i decrements its backoff time only when all the other stations have nonzero backoff times:

$$P[j,k-1|j,k] = p_0^{(i)},\ \forall\ j,\ 1 \leq k \leq W_j^{(i)} - 1$$

where $p_0^{(i)}$ is the probability perceived by station i that the channel is idle.

Let $p_{m,n}^{(i)} = \lim_{t\to\infty} P[w_i(t) = m, b_i(t) = n]$ represent the stationary distribution of the Markov chain for station i, where $m = 0, 1, \cdots, M^{(i)}$ and $n = 0, 1, \cdots, W_m^{(i)} - 1 = 2^m \cdot W_0^{(i)} - 1$. The following recursive relation holds in the steady state:

$$p_{m-1,0}^{(i)} p_f^{(i)} = p_{m,0}^{(i)},\ 0 < m \leq M^{(i)}. \quad (16)$$

By using (16), we can obtain

$$p_{m,0}^{(i)} = \left(p_f^{(i)}\right)^m p_{0,0}^{(i)},\ 0 < m \leq M^{(i)}. \quad (17)$$

From the structure of the Markov chain, the following relations can also be found. For $n \in \{1, \cdots, W_0^{(i)} - 1\}$,

$$p_{0,n}^{(i)} = \frac{W_0^{(i)} - n}{W_0^{(i)} \cdot p_0^{(i)}} \left[\left(1 - p_f^{(i)}\right) \cdot \sum_{k=0}^{M^{(i)}-1} p_{k,0}^{(i)} + p_{M^{(i)},0}^{(i)}\right] \quad (18)$$

while for $0 < m \le M^{(i)}$ and $n \in \{1, \cdots, W_m^{(i)} - 1\}$,

$$p_{m,n}^{(i)} = \frac{W_m^{(i)} - n}{W_m^{(i)} \cdot p_0^{(i)}} p_f^{(i)} p_{m-1,0}^{(i)}. \tag{19}$$

Substituting (17) into (18) and (19), we can obtain

$$p_{m,n}^{(i)} = \frac{W_m^{(i)} - n}{W_m^{(i)} \cdot p_0^{(i)}} p_{m,0}^{(i)} \forall m,\, n \in \left\{1, 2, \ldots, W_m^{(i)} - 1\right\}. \tag{20}$$

Finally, $p_{0,0}^{(i)}$ can be obtained by using $\sum_m \sum_n p_{m,n}^{(i)} = 1$, (17), and (20):

$$\begin{aligned} p_{0,0}^{(i)} = & \left[\left(1 - \frac{1}{2p_0^{(i)}}\right) \frac{1 - \left(p_f^{(i)}\right)^{n_1}}{1 - p_f^{(i)}} + \frac{W_0^{(i)}}{2p_0^{(i)}} \frac{1 - \left(2p_f^{(i)}\right)^{n_1}}{1 - 2p_f^{(i)}} \right. \\ & + \left(p_f^{(i)}\right)^{m^{(i)}+1} \left(1 + \frac{2^{m^{(i)}} W_0^{(i)} - 1}{2p_0^{(i)}}\right) \\ & \left. \times \frac{1 - \left(p_f^{(i)}\right)^{n_2}}{1 - p_f^{(i)}} \right]^{-1} \end{aligned} \tag{21}$$

where $n_1 = \min(n^{(i)}, m^{(i)}) + 1$ and $n_2 = \max(0, n^{(i)} - m^{(i)})$.

One should note that $p_0^{(i)}$ and $p_f^{(i)}$ themselves are functions of $p_{0,0}^{(i)}$. Let $p_t^{(i)} = \sum_{m=0}^{M^{(i)}} p_{m,0}^{(i)}$ represent the probability that station i transmits a frame. Then, we have

$$p_0^{(i)} = \prod_{\forall\, k \neq i} \left(1 - p_t^{(k)}\right) \tag{22}$$

and

$$p_c^{(i)} = 1 - p_0^{(i)}. \tag{23}$$

So, a system of nonlinear equations with N parameters has to be solved for a wireless LAN with N stations, if all stations have different backoff parameters.

With this Markovian model, the required $CW_{\min}[i] (= W_0^{(i)} - 1)$ for a given set of ϕ_i''s can be calculated as follows. First, note that $p_{0,0}^{(i)}$, $p_0^{(i)}$, and $p_f^{(i)}$ in (21) can all be represented by $p_t^{(i)}$ using (17), (22), and (23), respectively. Therefore, we can obtain a system of equations for $p_t^{(i)}$ and $W_0^{(i)}$ from (21). Since station i's airtime usage ratio ϕ_i' is proportional to the probability that station i can access the wireless channel, namely, $p_t^{(i)}$, we can then replace $p_t^{(i)}$ by

$$p_t^{(i)} = c \cdot \frac{\phi_i'}{\sum_j \phi_j'} \tag{24}$$

step 1 Let $W_0^{(i)} = (W_0^{(1)} - 1) \cdot \frac{\phi_1'}{\phi_i'} + 1$
step 2 Calculate $p_t^{(k)}$ by solving the system equations given by Eq. (21)
    if $\left|\frac{p_t^{(1)}}{p_t^{(i)}} - \frac{\phi_1'}{\phi_i'}\right| \le$ error
        go to step 3
    else
        if $\frac{p_t^{(1)}}{p_t^{(i)}} < \frac{\phi_1'}{\phi_i'}$
            $W_0^{(i)} = W_0^{(i)} + 1$;
        else
            $W_0^{(i)} = W_0^{(i)} - 1$;
        go to step 2
step 3 Set $CW_{min}[i] = W_0^{(i)} - 1$ and stop

Fig. 7. Pseudo-code for calculating $CW_{\min}[i]$ values.

TABLE II
PARAMETERS FOR SIMULATION

SIFS	DIFS	Slot Time	Preamble length	Packet length	ACK length	MAC heard + CRC
10 usecs	50 usecs	20 usecs	144 usec	1500 bytes	14 bytes	34 bytes

and solve the resulting system of equations to obtain the required $W_0^{(i)}$s. [7] However, as in the case of controlling AIFSs, it is not an easy task to solve a system of discrete nonlinear equations. Therefore, we again propose an iterative algorithm, shown in Fig. 7, to calculate $W_0^{(i)}$s.

In this algorithm, the initial solution for $W_0^{(i)}$'s, according to (14), is set as

$$\frac{CW_{\min}[i]}{CW_{\min}[1]} \approx \frac{\phi_1'}{\phi_i'} \tag{25}$$

where $W_0^{(i)} = CW_{\min}[i] + 1$. As we will show in the next section, (14) gives a very good solution for the above system of equations. Thanks to good selection of the initial solutions, we were able to obtain the required random backoff parameters in less than five iterations for most of the simulated cases.

IV. NUMERICAL AND SIMULATION RESULTS

We consider the EDCA mode of an IEEE 802.11e wireless LAN. It is assumed that each station can only transmit/receive frames to/from the AP (i.e., in the infrastructure mode) and may transmit at 11, 5.5, 2, and 1 Mb/s. Furthermore, we assume that all frames have the same length and $retry\ limit = 7$. All stations are assumed to be continuously backlogged and each station can only transmit one frame on each transmission opportunity. Throughout this section, we only simulate the CSMA/CA process and exponential random backoff with the properly-chosen EDCA parameters. We do not include the RTS/CTS and ACK frames, but the associated overheads are considered when evaluating the throughput. The simulation is conducted by using an event-driven scheduler written in Matlab code. The parameters used in the simulation are based on the IEEE 802.11b [2] standard and are summarized in Table II.

[7]Here, we have N variables, including c as a constant to be determined and $W_0^{(i)}$ for $i = 2$ to N with $W_0^{(1)} = 32$.

TABLE III
DECREMENTING LAG: $N = 4$ AND $AIFS[i] - AIFS[i-1] = 2$

(K_1, K_2, K_3, K_4)	$E[D^{(2)}]$	$E[D^{(3)}]$	$E[D^{(4)}]$
(1, 1, 1, 1)	1.96	3.80	5.42
	1.969	3.817	5.35
(2, 1, 1, 1)	1.93	3.65	5.08
	1.937	3.63	4.91
(3, 1, 1, 1)	1.90	3.50	4.77
	1.906	3.45	4.57
(4, 1, 1, 1)	1.88	3.37	4.51
	1.89	3.31	4.17
(4, 2, 1, 1)	1.88	3.38	4.47
	1.89	3.29	4.04

A. Airtime Usage Control via Controlling AIFS Values

Before presenting the results of stations' airtime usage, we first give an example to show the accuracy of (10). Table III shows the average decrementing lags for the case of four classes of stations, each with a different airtime usage ratio. Here, we assume that $AIFS[i] - AIFS[i-1] = 2$ for $i = 2$ to 4 and we change the number of type-i stations, K_i, to investigate their effects on the decrementing lag. Even though the derivation of (10) needs some approximations, the estimation error is shown to be small. The error mainly results from the use of $CW_{\min}/2$ to approximate the mean value of a station's backoff time. This may not be accurate enough due to the exponential increment and reset mechanism of contention window size in the 802.11 standard. This problem can be alleviated by using

$$E[BT] \approx \left(1 - \frac{\sum_i K_i}{CW_{\min}}\right)\frac{CW_{\min}}{2} + \frac{\sum_i K_i}{CW_{\min}} CW_{\min} \quad (26)$$

to include the effects of collisions and the subsequent exponential increase of stations' backoff times. Here, $\sum_i K_I/CW_{\min}$ accounts for the collision probability and $CW_{\min}$ represents the average backoff time a station may choose after the first collision. We do not consider the effect of exponential increase of CW resulting from more than two consecutive collisions because they rarely occur. In general, (10) gives better estimation of $E[D^{(i)}]$. In this example, the largest estimation error occurs when $K_1 = 4$, $K_2 = 2$, $K_3 = 1$ and $K_4 = 1$ but it is only about 10%.

Next, we show that small differences among stations' AIFS values suffice to provide differentiated airtime usage to different stations. We consider three different cases in which the wireless LAN provides two (Case I), three (Case II), and four (case III) classes, respectively. Stations in each class will be allocated the same amount of airtime. In Case I, we assume $K_1 = K_2 = 3$ and choose $AIFS[2] - AIFS[1] = 4$ according to Section III-C so that a station in the first class can have twice the airtime of another station in the second class. In Case II, we set $AIFS[2] - AIFS[1] = 3$ and $AIFS[3] - AIFS[2] = 4$ so that the ratio of stations' airtime usage in each class is close to 3:2:1, given that there are two stations in each class. Finally, we consider an airtime ratio as 4:3:2:1, given that there are two stations in each class in Case III. In this case, $AIFS[2] - AIFS[1] = 2$, $AIFS[3] - AIFS[2] = 2$, and $AIFS[4] - AIFS[3] = 3$. The achievable ratio (by the chosen AIFS values) and the simulation results are plotted in Fig. 8. The results match well with each other (with the largest error $\approx 6\%$) and show that a small difference among stations' AIFS values suffices to achieve the desired airtime allocation. One of the reasons why we cannot obtain the exact ratio in Case III (4.25 instead of 4.00 in Case III) is that we only use integer multiples of slot times when choosing the AIFS values. If we are allowed to use any value, the exact ratio can be achieved.

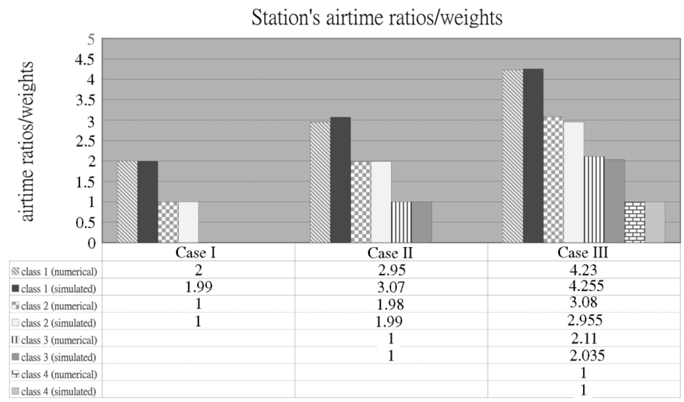

Fig. 8. The stations' airtime usage by controlling AIFS values.

TABLE IV
RANDOM BACKOFF PARAMETERS FOR THE AIRTIME FAIRNESS

Station no.		STAs 1-2	STAs 3-4	STAs 5-6	STAs 7-8
Assigned weight		8	4	2	1
I	$W_0(=CW_{\min}+1)$	32	64	128	256
II	$W_0(=CW_{\min}+1)$	35	66	128	254
Station no.		STAs 1-4	STAs 5-8	STAs 9-12	STAs 13-16
III	$W_0(=CW_{\min}+1)$	32	64	128	256
IV	$W_0(=CW_{\min}+1)$	64	128	256	512
V	$W_0(=CW_{\min}+1)$	34	64	128	258

B. Airtime Usage Control via Controlling $CW_{\min}$ Values

Two sets of analysis are conducted in this subsection—both have four different airtime usage ratios assigned to different stations. Consider the first set (Case I and II), in which there are eight stations with their assigned ratios/weights shown in Table IV. In Case I, we use (14) so that the value of $CW_{\min}$ is inversely proportional to a station's weight. The numerical result plotted in Fig. 9 shows that this simple control cannot achieve the desired airtime allocation. STA 1's or STA 2's share of airtime is $(8.94-8)/8 = 12\%$ more than the assigned ratio. Moreover, the largest overuse of airtime (by STA 1 or STA 2) is almost equal to the smallest-weight station's share of airtime. Again, the error of (14) results from using $CW_{\min}/2$ as the average value of random backoff time. The results can be substantially improved in Case II by using the algorithm in Section III. The resultant ratio is almost equal to the assigned value with an error less than 1%. The largest overuse of transmission time by any station is less than 3% of the share of the smallest-ratio station, as compared to 94% in Case I. In Cases III–V, we consider 16 stations. The random backoff parameters used are also shown in

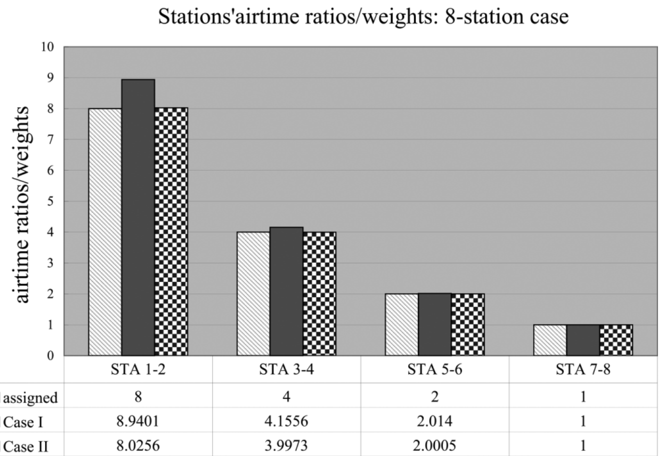

Fig. 9. Comparison between basic and optimal controls: 8 stations.

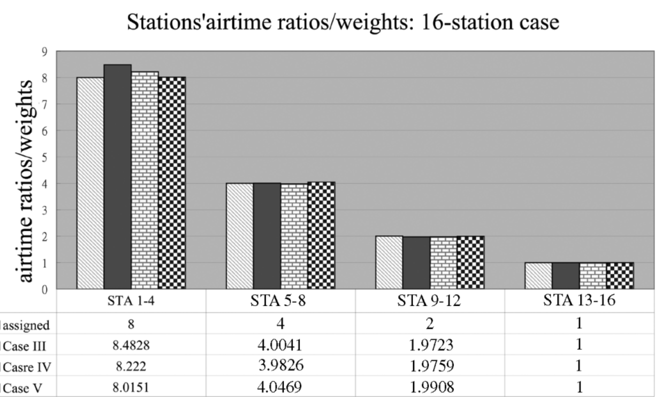

Fig. 10. Comparison between basic and optimal control: 16 stations.

Table IV. By using the parameters in Case IV, which double the $CW_{\min}$ values in Case III, the results can be improved because the number of collisions is reduced by using a larger contention window size. In this case, $CW_{\min}/2$ well represents a station's average backoff time. However, there are still some discrepancies between the assigned and the actual airtime usage ratios. In Case V, we use the parameters obtained from our Markovian model, and they achieve the best result under this scenario as shown in Fig. 10. The numerical and simulation results (using the parameters in Case II and V of Table IV) are compared in Table V. The largest error is less than 2%, showing that the parameters determined by our model can accurately provide stations the assigned share of airtime.

C. AIFS Versus $CW_{\min}$

Even though controlling AIFS and $CW_{\min}$ can both achieve the desired airtime allocation, each method has its own advantages and disadvantages. For airtime control using AIFS, we only need small differences between stations' AIFS values. Since stations do not rely on contention window sizes for airtime usage control, they can use the same and smaller $CW_{\min}$. As a result, the airtime wasted due to stations' backoff is reduced compared to the case of using $CW_{\min}$. Despite of its efficiency, such a control is sensitive to the number of stations in the wireless LAN. That is, if the number of stations changes, the required AIFS values of all stations may need to

TABLE V
COMPARISON BETWEEN ANALYTICAL AND SIMULATION RESULTS: 8 AND 16 STATIONS

Assigned weight		8	4	2	1
8 stations	Numerical results	8.0256	3.9973	2.0005	1.0000
	Simulation results	8.088	4.004	1.979	1.002
		8.081	3.998	1.996	0.9983
16 stations	Numerical results	8.0151	4.0469	1.9908	1.0000
	Simulation results	7.954	3.950	1.998	1.009
		7.894	4.011	1.947	0.9985
		7.942	4.021	1.957	0.9831
		8.002	3.969	1.962	1.0086

change according to Section III-C and the simulation results. On the other hand, controlling airtime usage using $CW_{\min}$ is affected less by the changes in the number of stations. If we double the number of stations, using the same set of $CW_{\min}$ values (e.g., 32, 64, 128, and 256 in Cases I and III of Table IV can still approximate the desired airtime ratio. In fact, if larger values are used (e.g., $CW_{\min}$ values in Case IV—64, 128, 256, and 512), the airtime ratio will be closer to the desired values irrespective of whether there are 8 or 16 stations in the system. That is, the larger the $CW_{\min}$ values, the more insensitive our control will be to the changes in the number of stations. The disadvantage of using larger $CW_{\min}$ values is the waste of more system airtime due to stations' longer backoff times. When the number of stations is small, this may lead to the reduction of overall system throughput.

In general, we may need to change AIFS or $CW_{\min}$ when a new station joins/leaves the wireless LAN, or when a station reduces its transmission rate as a result of link adaptation. In an infrastructure wireless LAN using EDCA, the AP should compute the parameters and broadcast them via beacon frames to the stations. Therefore, the aforementioned sensitivity of the AIFS control is not a big problem. Note, however, that such a requirement is significantly different from that in the centralized or polling-based access control. Here, transmission of individual frames relies on stations' properly-chosen AIFS or $CW_{\min}$. The AP need not schedule the transmission of individual frames to/from the stations, thus reducing the scheduling overhead as compared to centralized or polling-based algorithms. More importantly, our contention-based algorithm outperforms the centralized ones when the coverage of two wireless LANs overlaps with each other. In such a case, the schedulers (usually, the APs) need to resolve potential conflicts in their schedules via Inter Access Point Protocol (IAPP) [26]. However, such negotiation between APs is not required if the contention-based access control is used.

In an ad hoc 802.11 wireless LAN, computation of optimal parameters may be infeasible since stations may not have complete information of the entire wireless LAN. In such a case, we can rely on $CW_{\min}$ for coarser airtime usage control according to (14). The simulation results in the next section show that even with such a coarse control, the station's airtime usage is still well controlled. Additional signaling via the RTS/CTS frame exchange can also be used for better airtime usage control and even for QoS support as suggested in [27].

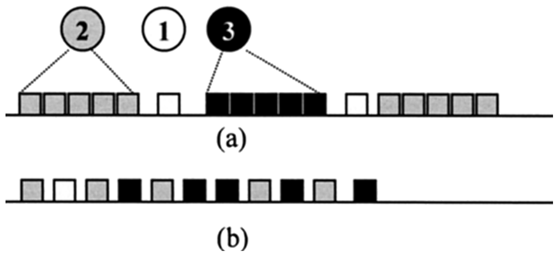

Fig. 11. Channel access via controlling TXOP versus controlling $CW_{\min}$.

D. Comparison With Other Airtime-Usage Controls

As mentioned in Section III, a station may control its airtime usage by adjusting the duration of its possession of the wireless channel (called *transmission opportunity* (TXOP) in the 802.11e standard). This can be achieved by controlling the frame size or the number of frames allowed to be transmitted during each possession of the wireless channel [22]. Controlling the frame size has its limitation since transmitting small frames reduces network efficiency due to the header overhead [28], while transmitting large frames is limited by the maximum frame size (usually 1500 bytes). Therefore, controlling the number of frames (of a fixed, reasonable size) transmitted during each possession of the channel is a logical solution.

The advantage of controlling the number of frames is that each station can use the same CSMA/CA and backoff parameters, thus reducing the computational complexity. The drawback is that by allowing back-to-back transmission from each station, the stations may suffer a longer average delay. This can be illustrated by a simple example—STAs 1, 2, and 3 require airtime usage with a ratio of $\phi_1 : \phi_2 : \phi_3 = 1 : 5 : 5$. To achieve the target ratio by controlling the number of frames, these stations transmit 1, 5, and 5 frames, respectively, during each of their possessions of the channel as shown in Fig. 11(a). As a result, STA 3 (or 2) cannot transmit until the completion of 5 frame transmissions from STA 2 (or 3). However, by controlling a station's chance of winning the contention (via controlling the AIFS or $CW_{\min}$ value), STAs 2 and 3 have an equal probability—which is 4 times larger than STA 1's—to access the channel. STA 2 or 3 only needs to transmit a frame each time as shown in Fig. 11(b), thus reducing the average frame delay.

Fig. 12 shows the average frame delay by controlling the TXOP and controlling $CW_{\min}$. We assume that there are 16 stations, each transmitting at 11 Mb/s. Furthermore, these stations are divided into four classes, each with four stations requiring the same data rate. Each station in the four classes requires 100, 200, 400, and 800 kb/s, respectively. Therefore, the total traffic load is $4*100+4*200+4*400+4*800 = 6000$ kb/s or 6 Mb/s. To meet this requirement (i.e., $\phi_1 : \phi_2 : \phi_3 : \phi_4 = 1 : 2 : 4 : 8$) by controlling TXOP, we set TXOP to 1.5, 3, 6, and 12 ms so that stations in these four classes can transmit up to 1, 2, 4, and 8 back-to-back frames, respectively. To meet the same requirement by controlling $CW_{\min}$, we set $CW_{\min} = 256$, 128, 64, and 32 for stations in these classes, respectively, and TXOP = 1.5 ms for all stations. The simulation result shows that the stations suffer a longer delay if their airtime usage is controlled by TXOP.

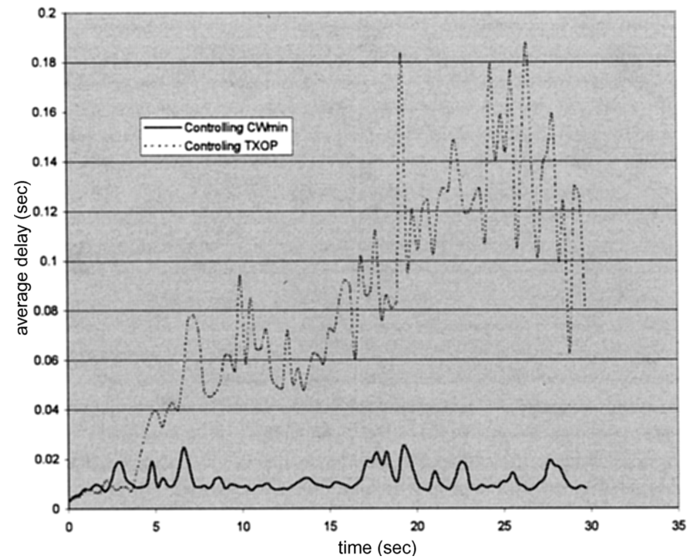

Fig. 12. Comparison of average frame delay using TXOP and $CW_{\min}$.

E. Airtime Usage Control With Location-Dependent Transmission Rate (LDR)

In the previous subsections, we assume that all stations use the same transmission rate and show how the stations' airtime usage can be controlled in a distributed manner. To investigate the impact of LDR on stations' airtime usage and show how the parameters should be adjusted, we consider eight stations using different transmission rates. We assume that stations 1 and 2 use 11 Mb/s, stations 3 to 5 use 5.5 Mb/s, and stations 6 to 8 use 2 Mb/s. For an illustrative purpose, we assume that they should use an equal amount of airtime. To achieve such airtime allocation, the number of times a station accesses the channel [i.e., n_i in (4)], should be inversely proportional to its transmission rate. For example, station 1 should access the channel twice as frequently as station 3 because the airtime needed by station 3 to transmit a frame is doubled. Based on the ratio of n_i, we choose W_0 to be 35, 66, and 176 for stations 1 and 2, stations 3 to 5 and stations 6 to 8, respectively, according to Section III. The values of $CW_{\max}$ are $2^5 \cdot W_0 - 1$, $2^4 \cdot W_0 - 1$, and $2^3 \cdot W_0 - 1$ for these three groups of stations. For a comparison purpose, we also let all stations use $W_0 = 32$ and $CW_{\max} = 1023$ as in a regular IEEE 802.11 wireless LAN without airtime control. The station's airtime $T_i(0, t)$ is plotted in Fig. 13 for both cases. Thanks to the properly-chosen $CW_{\min}$ values, all stations can have an equal share of airtime regardless of their underlying transmission rates. In contrast, as explained in Section II, stations receive the airtime inversely proportional to their transmission rates if there is no control over their airtime usage. The corresponding throughputs are listed in Table VI. If there is no control over stations' airtime usage, all stations will have an equal throughput but the system throughput is reduced. In this simulation, the system throughput with airtime control is 57% higher than that in a regular wireless LAN. Of course, the improvement

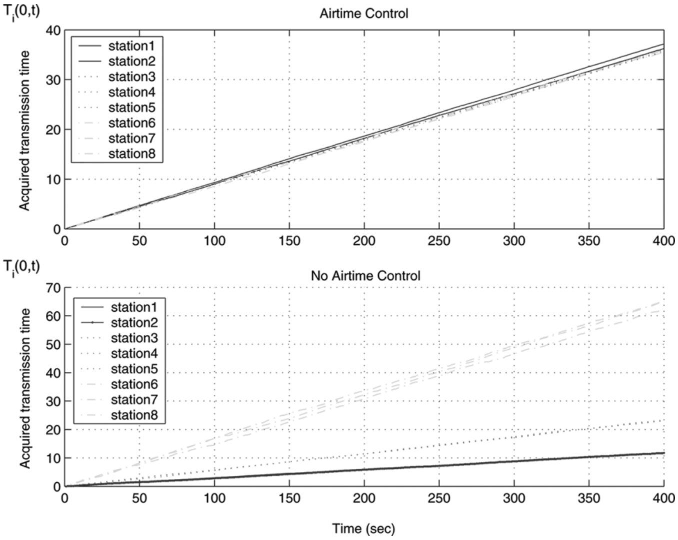

Fig. 13. Station-received airtime with and without airtime control.

TABLE VI
THROUGHPUT (Mb/s) PERFORMANCE WITH AND WITHOUT AIRTIME CONTROL IN CASE OF LDR

	STAs 1-2	STAs 3-5	STAs 6-8	Total
Airtime Control	0.988 1.013	0.494 0.490 0.497	0.177 0.179 0.181	4.021
No airtime Control	0.322 0.320	0.317 0.321 0.319	0.326 0.310 0.325	2.56

depends on the stations' transmission rates and the assigned ratios, and may vary case by case. However, our control can yield a higher system throughput since lower-transmission rate stations will not "use up" all network resources.

If stations change their transmission rates, either the AIFS or $CW_{\min}$ values have to be changed in order to maintain the negotiated airtime usage. If a station lowers its transmission rate, it should then avoid using too much airtime. That is, it should reduce the frequency of accessing the wireless channel [i.e., smaller n_i in (4)]. For example, if STA 1 lowers its transmission rate from 11 Mb/s to 5 Mb/s, it should access the channel 50% less frequently than before. As in the case of changes in the number of stations, the AP should re-compute the optimal AIFS or $CW_{\min}$ values for such adjustment in the contention mode of an infrastructure wireless LAN or the station adjusts its $CW_{\min}$ according to (14) in an ad hoc wireless LAN.

V. CONCLUSION

In this paper, we proposed the airtime usage control of stations in multirate wireless LANs. Two different controls, one using AIFS and the other using $CW_{\min}$ values, are developed in order to achieve the desired airtime allocation on a contention and distributed basis. Both the analysis and simulation results show that we can finely control the stations' share of airtime with the proposed control mechanisms. With properly-chosen parameters by the admission control, service differentiation or prioritized QoS support can be achieved, even in the presence of location-dependent transmission rates in a multirate wireless LAN.

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