Design and Evaluation of a Contention-Based High Throughput MAC with Delay Guarantee for Infrastructured IEEE 802.11 WLANs

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Abstract: **This paper proposes a complete solution of a contention-based medium access control in wireless local networks to provide station level quality of service guarantees in both downstream and upstream directions. The solution, based on the mature distributed coordination function protocol, includes a new fixed contention window backoff scheme, a tuning procedure to derive the optimal parameters, a super mode to mitigate the downstream bottleneck at the access point, and a simple admission control algorithm. The proposed system guarantees that the probability of the delay bound violation is below a predefined threshold. In addition, high channel utilization can be achieved at the same time. The numerical results show that the system has advantages over the traditional binary exponential backoff scheme, including efficiency and easy configuration.**

Index Terms: **Wireless local area network (WLAN), medium access control (MAC), quality of service (QoS).**

I. INTRODUCTION

In recent years, wireless local area networks (WLAN) have been widely used because of their easy deployment and low cost. The primary medium access control (MAC) protocol [1] of IEEE 802.11 is called the distributed coordination function (DCF) by which stations contend for the wireless channel using the carrier sense medium access with collision avoidance (CSMA/CA) protocol. With the increasing demand for real-time applications, traffic differentiation has become insufficient and quality of service (QoS) guarantees are desired. This paper focuses on the MAC based on the mature DCF protocol with the burst transmission feature by which a station can continuously send traffic for a period called the transmission opportunity ($TXOP$) after it contends the medium successfully. Although this feature is introduced in the new IEEE 802.11 standard [1], many commercial DCF-based products already have this packet bursting feature prior to the publication of the standard. The basic idea of this paper is to add the capability of QoS guarantee for the DCF MAC with the burst transmission feature. The major contribution of this paper is to present a complete solution including the parameter tuning procedure to achieve the best performance and the super mode to mitigate the traffic asymmetry issue between upstream and downstream. In addition, an admission control algorithm is proposed to control the admission of new stations and corresponding parameters.

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The performance of the DCF has been extensively analyzed and evaluated, but tuning the parameters for a good operating point has seldom been addressed. Bianchi [2] developed a two dimensional Markov chain to compute the saturation throughput of the IEEE 802.11 DCF protocol. Chatzimisios *et al.* [3], [4] extended Bianchi's work for the case with a finite retry limit and computed the average packet delay in addition to the throughput. Afterwards, a new approach for deriving the service time delay generating function is presented by [5]. However, those papers [3]–[6] are limited to the average delay. More recently, the authors in [7], [8] proposed analytic models to estimate the delay distribution with given system parameters such as the number of stations, the minimum contention window, and the retry limit. Although previous research indicated that the performance heavily depends on the parameters, none provided a method to adjust the parameters for the best performance. Zhai *et al.* [9] proposed a method to achieve high throughput by controlling the aggregated traffic loading in the network. However, the implementation cost is high because each station requires a traffic regulator. This paper proposes a different approach to control the operating point by adjusting the relative parameters such that the high channel utilization and QoS guarantee can be achieved at the same time.

In addition to QoS, this paper also deals with the traffic asymmetry problem between upstream and downstream in the infrastructure mode where users access the Internet through the access point (AP). Because of the contention-based MAC, the AP needs to contend with numerous stations and becomes the bottleneck for the downstream traffic. There are papers [10]–[13] about the asymmetric problem in the IEEE 802.11 WLANs. The authors in [10], [11] modified the IEEE 802.11e EDCA parameters and use a burst transmission mechanism to mitigate the asymmetric problem. The AP also needs to distinguish between transmission control protocol (TCP) data frames and TCP acknowledgement frames, but in fact the AP cannot identify this transport layer information because it is typically a layer 2 or layer 3 equipment. Therefore, this approach is not realistic. Wu *et al.* [12] proposed a DCF plus scheme to solve the asymmetric problem. After the destination station transmitted an acknowledgement frame, it could transmit a data frame back to the same source station. Although the overall traffic is symmetric, it treats users unfairly because users with large upstream traffic can gain more downstream bandwidth than others. Blefari-Melazzi *et al.* [13] used the token bucket scheme above the MAC layer to control the traffic rate and decrease the effect of TCP congestion

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control. Although it solves the asymmetric problem, it also decreases the total throughput because of shaping. To solve the asymmetric problem, this paper proposes a bidirectional transmission scheme allowing the AP to control the downstream traffic.

This paper is organized as follows. Section II presents a brief overview of the IEEE 802.11 DCF protocol and focuses on the backoff procedure. The design objective is formulated as an optimization problem in Section III. Based on the optimization result, this paper proposes the fixed contention window backoff (FCWB) scheme. Section IV presents the solution to the asymmetric problem and a simple admission control algorithm. Section V outlines and discusses the simulation environments and results. Finally, Section VI presents conclusions.

II. PRELIMINARIES

A. DCF Protocol

The DCF protocol is based on a standard ethernet-like contention-based service and adopts a slotted binary exponential backoff (BEB) scheme to avoid collisions. When a station has a frame to transmit, it needs to sense the wireless medium. If the medium is busy, it defers the transmission until the medium is idle. If the medium is detected to be idle for a time interval, which is a DCF inter-frame space (DIFS), the source station starts a backoff operation with a randomly-selected backoff count value. The backoff counter is decreased by one after an idle slot time and is frozen when the source station detects that the medium is busy. When the backoff counter reaches zero, the source station starts transmitting the frame. If multiple stations count down to zero at the same time, they transmit simultaneously and a collision occurs. When the destined station receives this frame successfully, it transmits an immediate positive acknowledgment (ACK) frame back after a time interval which is a Short Inter-Frame Space (SIFS). After the source station receives the ACK frame, the transmission is successfully completed. If the source station does not receive the ACK frame, it schedules a retransmission and the backoff operation restarts.

The randomly-selected backoff count value is chosen uniformly from [0, W_i-1] where W_i is the current contention window size and i denotes the backoff stage or the number of failed transmissions of a frame. The initial value of i is zero for each frame and it is increased by one after a failed transmission. The contention window size is controlled according to the BEB. At the first transmission attempt, W_0 is equal to the minimum contention window size denoted by W. When a station detects a failed transmission, it doubles W_i until $W_{\max}$ is reached as shown in (1). After that, W_i remains the same until the frame is transmitted or dropped.

$$W_i = \begin{cases} 2^i * W, & 0 \le i \le m, \\ 2^m * W, & m < i \le R \end{cases} \quad (1)$$

where R is the retry limit and m is the maximum number of times that W_i can be doubled. When a station fails to transmit the frame at stage R, it drops the frame and starts a new transmission for the head-of-queue packet with $W_0 = W$.

B. Problem Formulation

Unlike QoS-capable networks such as asynchronous transfer mode (ATM) where the user traffic can be controlled by policing or shaping, a WLAN network is unable to control the traffic entering the network such that end-users can arbitrarily send their packets. To guarantee the delay QoS at any time, it is necessary to consider the worst case that every station is always backlogged.

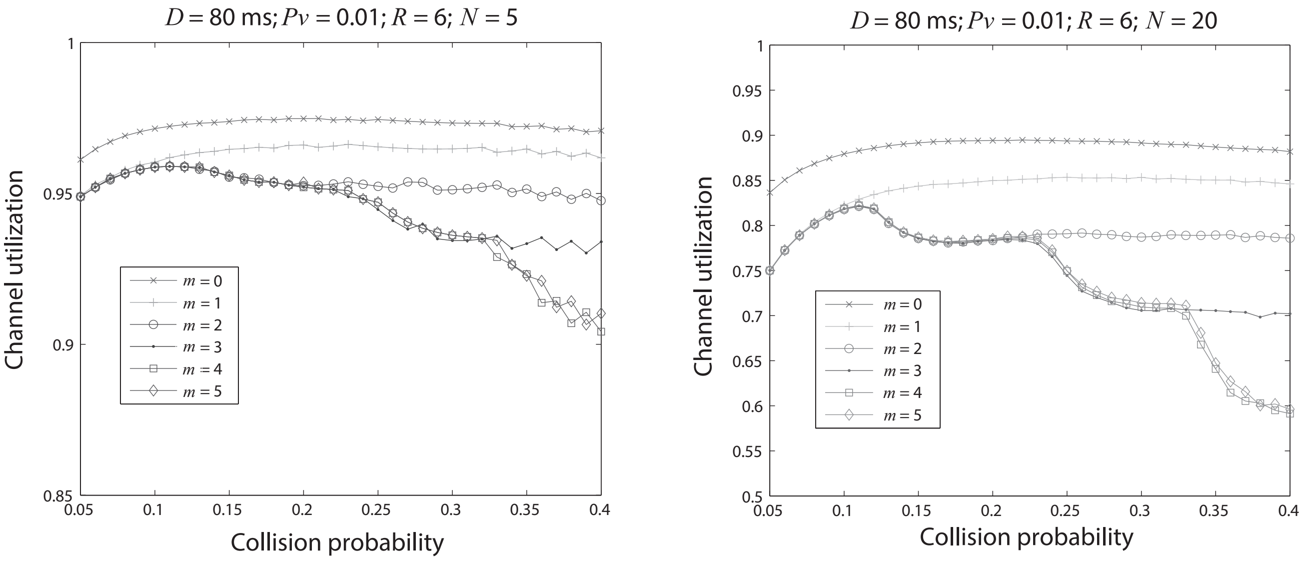
The considered wireless network consists of N stations including one AP and $(N-1)$ user stations (STAs). To increase the channel utilization, this paper assumes that every station supports the feature of burst transmission by which a station can continuously send traffic for a time period ($TXOP$) after it contends the medium successfully. Because the $TXOP$ duration is typically long, the request-to-send/clear-to-send (RTS/CTS) mechanism is adopted to protect the transmission burst from the other stations. The AP assigns equal TXOP for each station according to the delay requirement and the number of stations. Let d be a random variable representing the time period between the ends of two successive $TXOP$s for a station. A transmission burst is considered a violation if d is larger than a predefined delay bound, denoted by D. Therefore, the QoS objective is to keep the violation probability below a threshold, denoted by P_v, given by the network administrator. That is, $1 - P(d < D) \le P_v$. It should be pointed out that the violation probability is not equal to the loss probability. In most cases, packets can be sent to their destination successfully, but some of them suffer larger delays. Only frames with $(R+1)$ collisions are dropped by the AP or STAs.

Let τ be the probability that a station transmits in a randomly chosen slot time. For a transmission, the collision probability, denoted by p, is $1-(1-\tau)^{N-1}$. We can model the system by a two dimensional Markov chain [2], [3] where each state represents the backoff count value at the ith retry. After deriving the state probability of the Markov chain, one can get τ by adding the probabilities of the states with a zero backoff counter value. Although there are no closed form solutions for τ and p, they can be uniquely solved by numerical techniques for a given W [3].

Following the analysis in [3], we can extend and derive the channel utilization for the scenario with burst transmission. There are three cases in a randomly chosen slot time: (1) All stations are idle, (2) a station transmits a burst successfully, and (3) a collision occurs. Let P_{tr} be the probability that there is at least one transmission in the considered slot time and $P_{tr} = 1-(1-\tau)^N$. Let P_s be the probability of a successful transmission conditioned on the fact that at least one station transmits and $P_s = N\tau(1-\tau)^{N-1}/(1-(1-\tau)^N)$. It is obviously that the actual channel utilization depends on the packet size in a transmission burst, but it is too complicated if the traffic model is involved. For computational simplicity, the channel utilization, denoted by μ, is approximated as the average time for data in a cycle divided by the average time of a cycle and can be expressed by

$$\mu \approx \frac{P_{tr}P_s TXOP}{(1-P_{tr})T_e + P_{tr}P_sT_s + P_{tr}(1-P_s)T_c} \quad (2)$$

where T_e, T_s, and T_c are the slot time, the time to successfully transmit a burst, and the time wasted in a collision, respectively.



(a) (b)

Fig. 1. Channel utilization vs. collision probability: (a) $N = 5$ and (b) $N = 20$.

When the packet size is large or the frame aggregation feature in IEEE 802.11n [14] is enabled, the approximation error becomes small.

According to the QoS objective, the system needs to compute the probability $P(d < D)$. Since modern WLAN APs are typically an embedded system, the CPU, with limited computing power, needs to simultaneously handle many tasks such as network protocols, network management, and the Web server. The computation of delay distribution would be a burden for the CPU. We have developed a framework [15] to approximate the delay distribution where we have addressed the tradeoff between accuracy and complexity.

This paper considers both the channel utilization and the delay requirement, but unfortunately the two metrics conflict with each other. Equation (2) shows that the channel utilization is proportional to $TXOP$, but a large $TXOP$ also implies long frame delays because a station needs to wait longer while deferring. To maximize the channel utilization with the delay constraint, we can formulate the optimization problem as follows.

$$\text{Find } (W^*, m^*, TXOP^*) = \arg \max_{W,m,TXOP} \mu, \qquad (3)$$
$$\text{subject to } 1 - P(d < D) \leq P_v$$

where the parameters with superscript star represent their optimal values.

III. TUNING PROCEDURE AND THE FCWB SCHEME

The goal of the optimization problem is to find the optimal values for system parameters. Obviously, it is a complex task because there are many dependent parameters involved. This paper proposes a heuristic method that breaks the optimization process into two phases: Find m^* first and then find the operating point manually. Before going through the procedure, let us choose a proper control variable. Intuitively, a system with a large p suffers collisions, resulting in low channel utilization and large frame delays. Therefore, we take p as the control variable and calculate the corresponding values for τ and W. Because $p = 1 - (1-\tau)^{N-1}$, we have $\tau = 1 - \sqrt[N-1]{1-p}$. Based on the results in [3], we derive the corresponding minimum contention window as shown in (4).

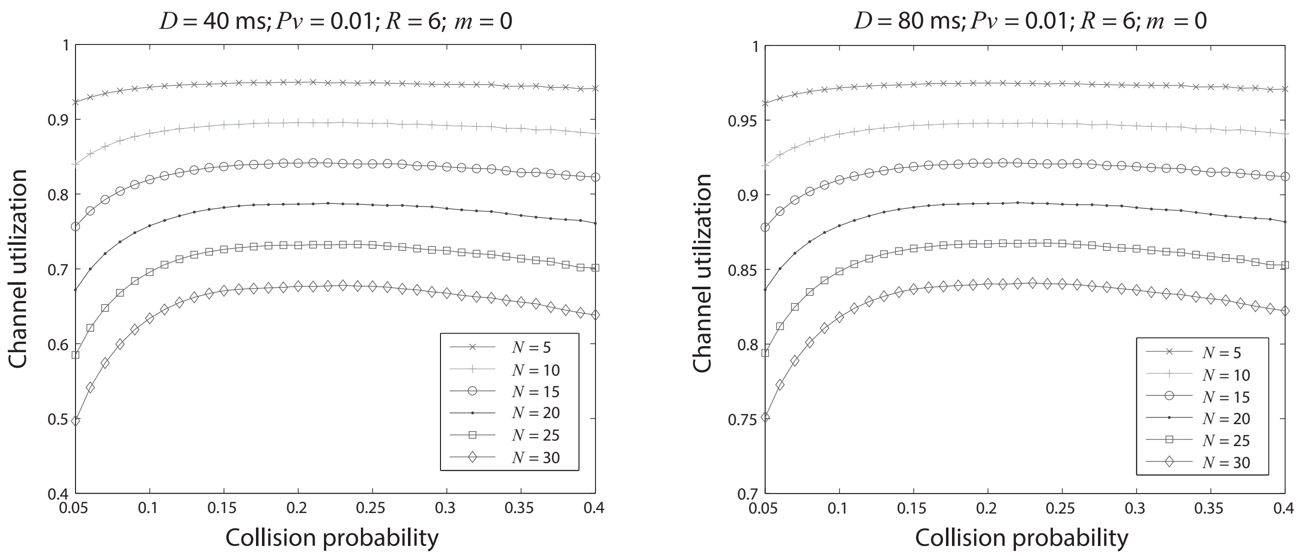
$$W = \begin{cases} \frac{(2-\tau)(1-2p)(1-p)^{m+1}}{\tau(1-(2p)^{m+1}(1-p))}, m \geq R, \\ \frac{(2-tau)(1-2p)(1-p^{R+1})}{\tau[(1-(2p)^{m+1}(1-p)+2^m p^{m+1}(1-2p)(1-p^{R-m})}, m < R. \end{cases} \qquad (4)$$

Phase 1: Find m^* according to the data generated by the following pseudo code.

1. Input R, N, m, D, and P_v.
2. Set $p = 0.05$.
3. Calculate τ and W.
4. Find the optimal $TXOP^*$ such that $(1 - P(d < D)) \leq P_v$ by the bisection method.
5. Calculate the channel utilization by (2).
6. If $p < 0.4$, then $p = p + 0.01$ and go to Step 3.

Fig. 1(a) shows the channel utilization for $D = 80$ ms, $P_v = 0.01$, $R = 6$, and $N = 5$. One can see that the channel utilization varies with different m, and the best performance is with $m = 0$. That is, it is unnecessary to double the contention window after a collision if the collision probability is well controlled. The same phenomenon exists for $N = 20$ as shown in Fig. 1(b). If the contention window of a tagged station is doubled, the probability that the other stations interrupt during its backoff process increases. The delay increases dramatically after several retries due to large contention windows, and the optimal $TXOP^*$ found in Step 4 is small, resulting in low channel utilization. Based on the above argument, we conclude that $m^* = 0$. As a result, this paper proposes the FCWB scheme that the contention windows for all backoff stages are equal to W.

Phase 2: Select p manually for $m = 0$. Following the same steps in Phase 1 for different N, the curves of channel utilization



(a) (b)

Fig. 2. Channel utilization vs. collision probability for different N: (a) $D = 40$ ms and (b) $D = 80$ ms.

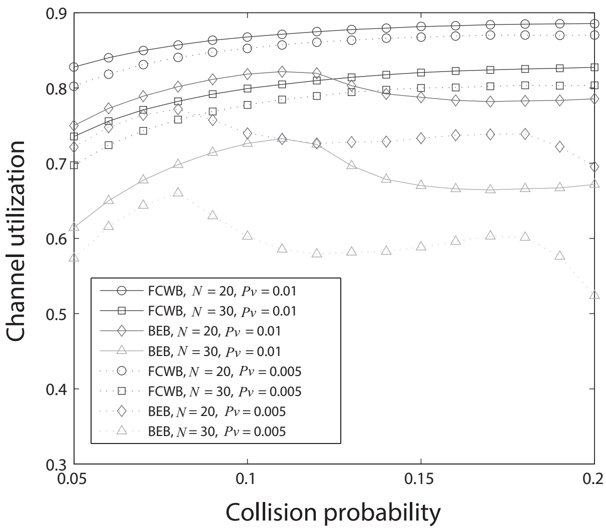


Fig. 3. Channel utilization for FCWB and BEB.

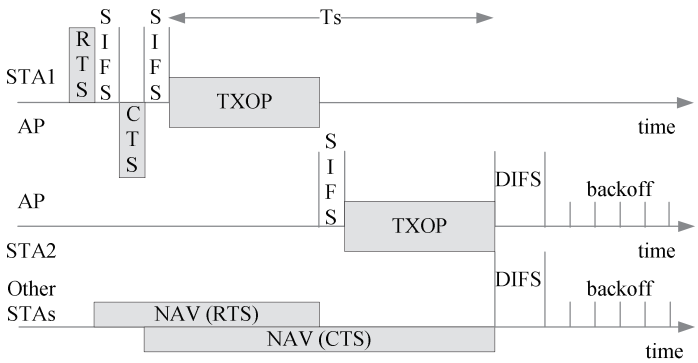


Fig. 4. Channel utilization for FCWB and BEB.

for $D = 40$ ms and $D = 80$ ms are plotted in Fig. 2. It is interesting that the peaks of all curves are situated around $p = 0.17$, which is the best operating point. Substituting $p = 0.17$ into (4), we can calculate the corresponding minimum contention window, which is W^*. Finally, $TXOP^*$ can be determined by Step 4 by the bisection search method. This completes the tuning procedure.

In summary, using the FCWB scheme, the AP only needs to determine W^* and $TXOP^*$ by Phase 2 when a STA joins or leaves. The optimal values are announced by the AP, and all STAs just follow the FCWB scheme and adjust their parameters accordingly. Phase 2 can also be applied to the BEB scheme. After some calculations, the channel utilizations for different combinations are plotted in Fig. 3. We can see that the FCWB scheme is more efficient because of the higher channel utilization. The flat curves for the FCWB scheme indicate that the proper operational range is large. For example, $p = 0.17$ is also a good choice for $P_v = 0.005$. As a result, one can substitute $p = 0.17$ into (4) for different N to get W^* for this case. On the other hand, the proper operational range for the BEB scheme is narrow, and a good choice of the operation point for the case of $P_v = 0.01$ is $p = 0.11$. However, it moves to $p = 0.08$ for the case of $P_v = 0.005$.

IV. SUPER MODE AND ADMISSION CONTROL

Traditionally the AP and STAs use the same parameters. If the number of STAs is large, the probability that the AP contends the media successfully is low, and downstream is the bottleneck for real-time applications. Since this paper aims at delay guarantees for both downstream and upstream, a bidirectional transmission method is proposed to solve this problem. In our approach, only the AP is modified and all STAs remain the same for ease of implementation.

There are two transmission modes in the AP. Upon receiving a RTS frame, the AP checks its queue to determine the operation mode. If backlog exists, it switches to the **super** mode as shown in Fig. 4. The major improvement in the **super** mode is that the AP can transmit a downstream burst after receiving an upstream burst. To notify other STAs of the duration being used, the AP calculates the corresponding network allocation vector (NAV) and put it in the CTS frame. The design differs from the bidirectional transmission method in [12], [16] where the receiving station (could be the AP or a STA) can piggyback a burst back to the source station. In this paper, the bidirectional transmission is only performed at the AP for two reasons. First, the overall system implementation complexity is low because only the AP is changed. Second, the traffic loading of downstream and upstream can be kept symmetric. In addition, our design allows the AP to select an active queue for the downstream transmission according to the round robin discipline. Only one packet is dequeued each time when a queue is visited. Since the downstream and upstream traffics in a STA may be unbalanced, this design can guarantee the downstream fairness between STAs. With the **super** mode and the round-robin scheduler, there is no random access and no collision for the downstream traffic. The simulation result, presented in the next section, shows that the violation probability at downstream is smaller than that at upstream. As a result, with the **super** mode, considering upstream is sufficient for the admission control.

After the AP finishes the transmission in the **super** mode, it switches back to the normal operation for the case of very light upstream traffic. To make the upstream and downstream traffic equal, the AP is configured with a lower priority than the STAs by setting the backoff counter to a fixed value k. A reasonable choice of k is W^*. After the medium is free, the AP needs to wait for k slots before transmission. In this manner, the AP is only active when there is no upstream traffic. In the case that the AP collides with the STAs, it needs to wait for another W^* slots. Because the backoff counter value of the STAs is smaller than W^*, the AP does not collide with the STAs again.

The admission control is performed when a new STA joins the network. The only two parameters configured by the administrator are the violation probability P_v and the delay bound D. When an STA leaves or joins the networks, the AP uses the tuning procedure proposed in Section III to calculate the optimal values for W^* and $TXOP^*$ and notifies all connected STAs to update. In addition, a slight modification is required in the calculation of channel utilization for the **super** mode because T_s is equal to (2*$TXOP + T_{\rm SIFS}$) where $T_{\rm SIFS}$ is the duration of SIFS.

Obviously $TXOP^*$ decreases as the network size increases. If the $TXOP$ per direction is too small to transmit the possible largest packet, the new STA is supposed to be rejected. Let $MSDU$, the abbreviation of MAC service data unit, be the size of payload. The minimum $TXOP$ value, denoted by $TXOP_{\rm min}$, can be calculated by

$$TXOP_{\rm min} = T_{PLCP} + \frac{\text{Max } MSDU}{\text{Min PHY rate}} + T_{\rm ACK} + T_{\rm SIFS} \quad (5)$$

where $T_{\rm PLCP}$ and $T_{\rm ACK}$ are the time periods used to transmit the PLCP header and the ACK packet, respectively.

Table 1. System parameters

Parameter	Value
Min PHY rate	24 Mbps
Max MSDU	2304 bytes
SIFS	16 μs
DIFS	34 μs
$T_{\rm RTS}$	14.7 μs
$T_{\rm CTS}$	12.7 μs
$T_{\rm ACK}$	12.7 μs
$T_{\rm PLCP}$	8 μs
Slot time	9 μs
W for BEB	32
m for BEB	5
R	6
D	80 ms
P_v	0.01
Simulation time	200 s

V. PERFORMANCE EVALUATION

This section shows the performance of the proposed system with the typical system parameters listed in Table 1. All simulations were conducted by the NS2 simulator [17]. We have built a new MAC module with the burst transmission and super mode functionalities. In addition, a new queue module with the round robin scheduler was also developed for the AP to complete the super mode.

A. FCWB Scheme

The first part focuses on the backoff scheme when the super mode is not involved. The test scenario consists of 20 stations including one AP and 19 STAs. Each STA has one UDP flow to the AP. There is no downstream traffic such that the performance depends only on the backoff scheme. The first STA is the tagged one with one 400 kbps CBR flow. We vary the data rate of non-tagged flows to emulate different network loadings. According the tuning procedure, the best operation point is around $p = 0.17$. W^* and $TXOP^*$ are 204 and 1098 μs, respectively, for the FCWB scheme. We also conducted simulations of the BEB scheme with and without the burst transmission feature for comparison. The $TXOP$ value for the BEB scheme is the same as that used for the FCWB scheme.

Fig. 5 contains 4 subplots each one showing the aggregated throughput, the average delay of the tagged flow, the delay standard variation (STD) of the tagged flow, and the violation probability, respectively. We can see that the throughput is significantly improved when the burst transmission feature is enabled. Because the collision is well controlled by the FCWB scheme, it outperforms the BEB scheme in throughput. In general, delay related metrics increase with the volume of total input traffic. Although the violation probability of the BEB without $TXOP$ scheme is small, the aggregated throughput be-

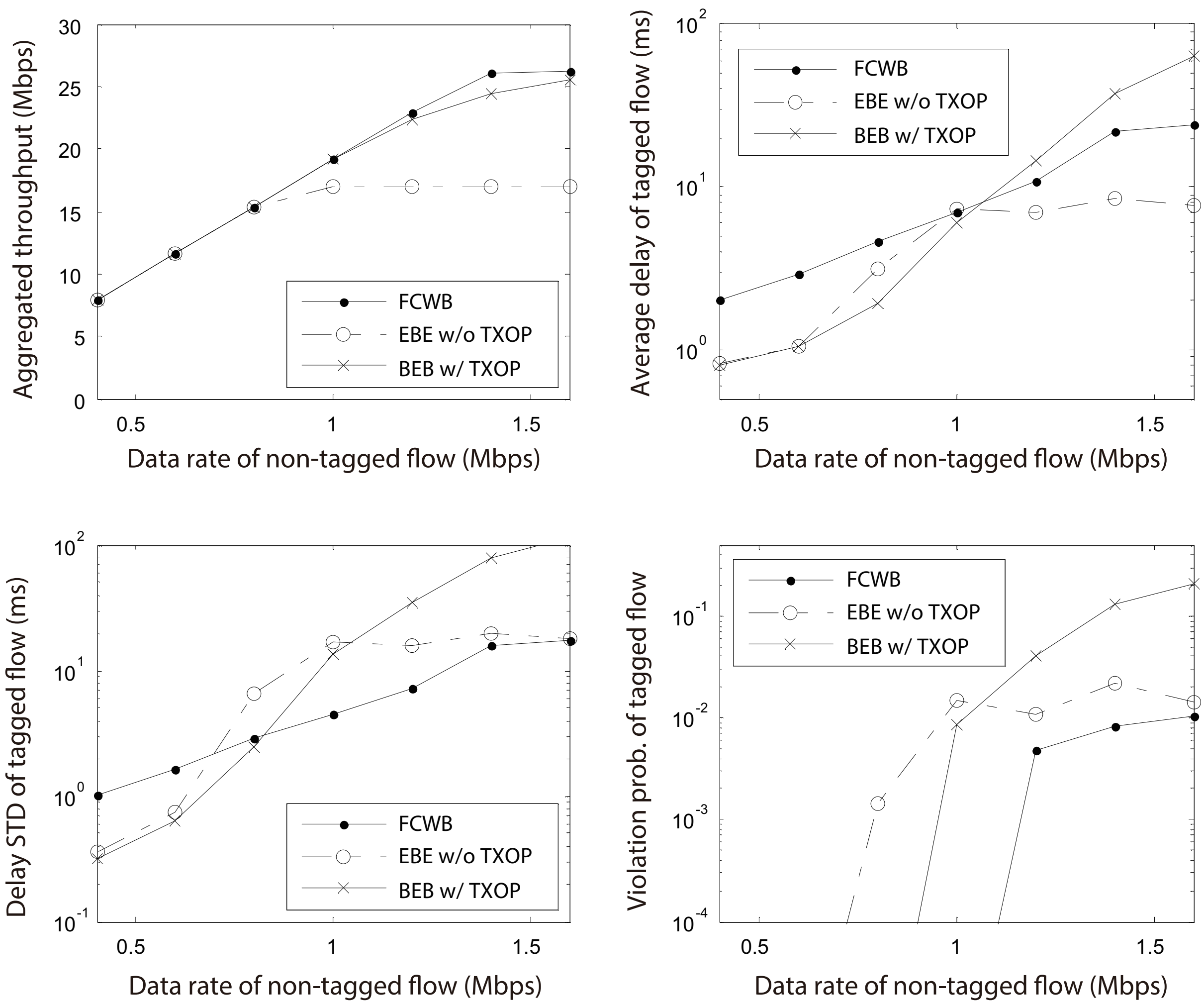


Fig. 5. Performance comparison.

comes saturated around 17 Mbps. On the other hand, the FCWB scheme can guarantee the violation probability while providing high throughput. If the burst transmission feature is applied to the BEB scheme, the throughput increases significantly with the cost of large violation probability. The plot of delay STD gives another aspect of behaviors between different schemes. The average delay of the BEB without $TXOP$ scheme is very small because of its small initial contention window. However, once a STA collides with others, its contention window increases dramatically. In a word, by the BEB scheme, the packets delivered without collision are transmitted to the receiver quickly, but collided packets may suffer long delay. As a result, the delay STD is larger than that of the FCWB scheme at high loads. In summary, the FCWB scheme trades the performance at light loads for the low delay variation at high loads by utilizing a fixed and large contention window.

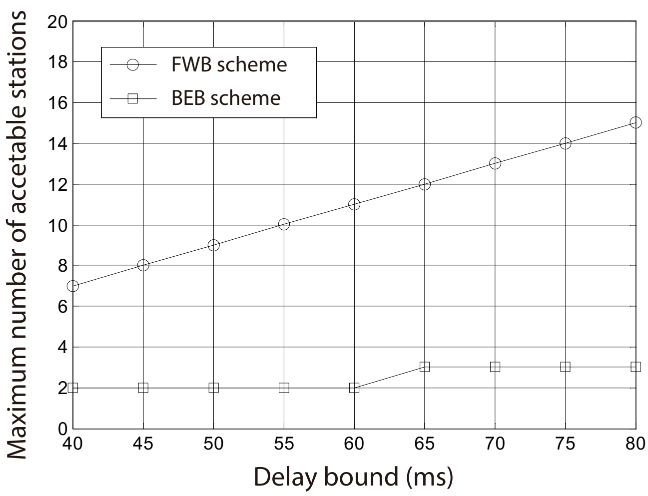


Fig. 6. The admission region.

B. Admission Control and **Super** *Mode*

This paper assumes that the minimum PHY rate and the maximum MSDU are 24 Mbps and 2304 bytes, respectively. Substituting the two parameters into (5), we obtain the minimum $TXOP$ of 805 μs. Because $TXOP^*$ decreases as the number of stations increases, this subsection presents the system capacity in terms of the number of acceptable stations with the super

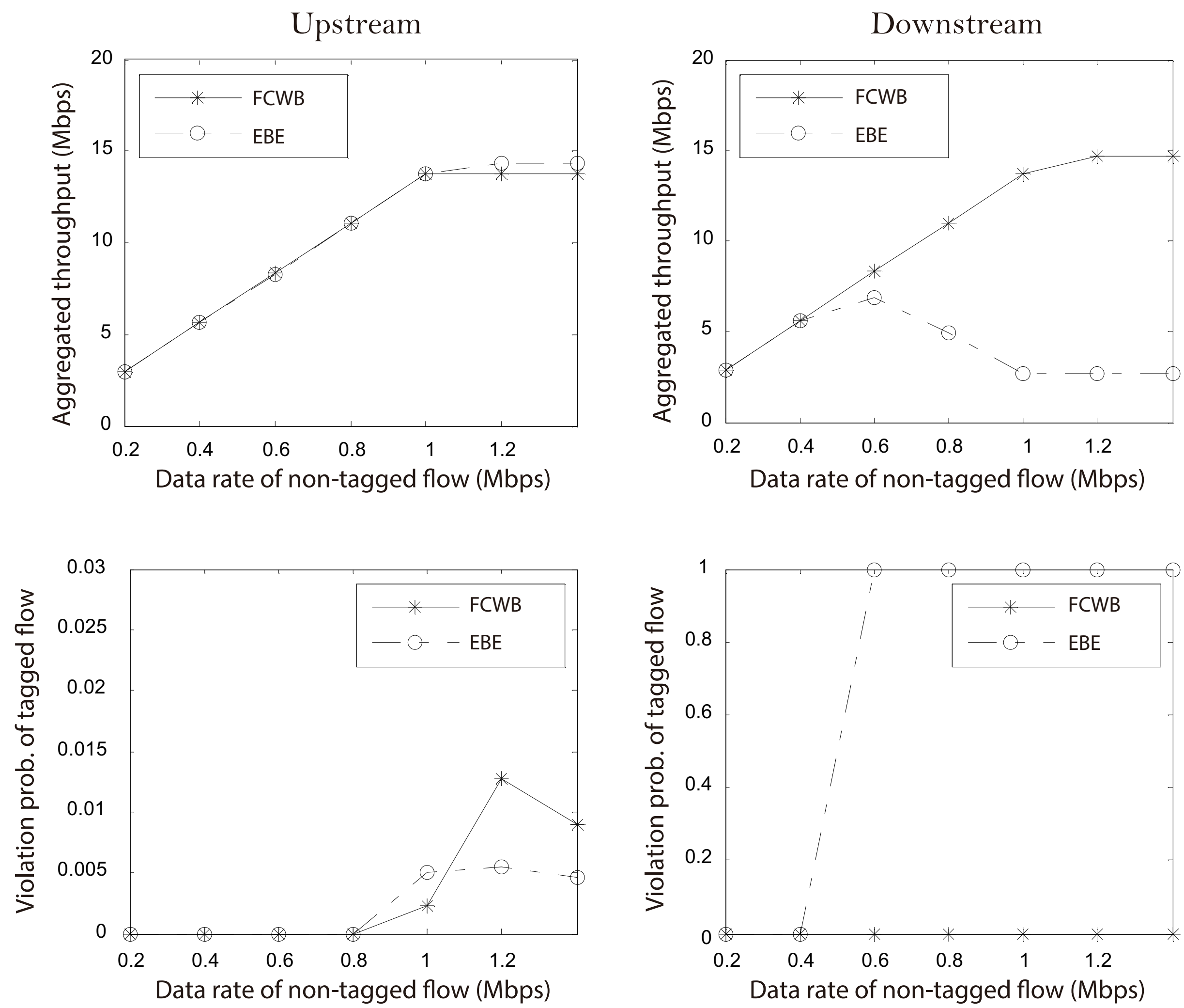
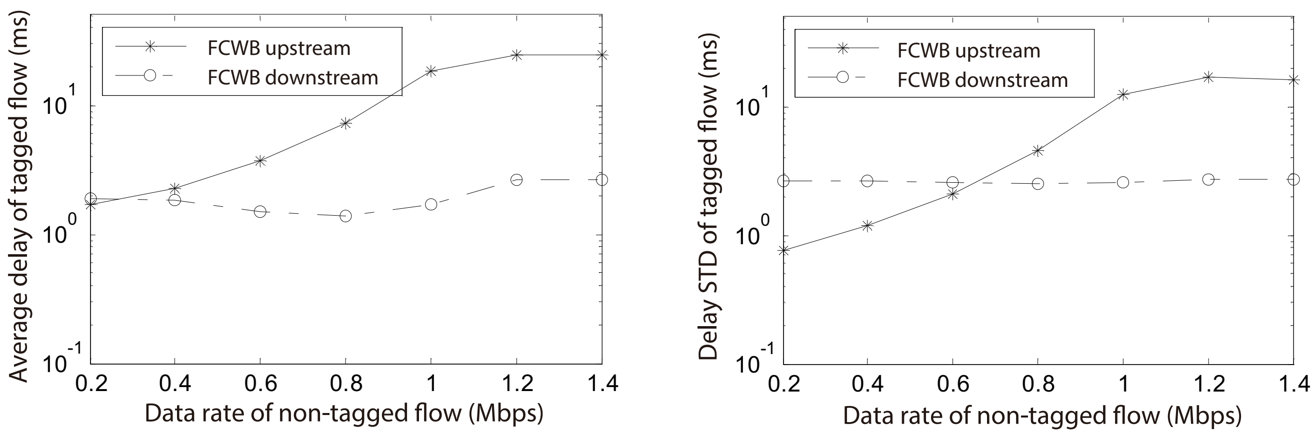


Fig. 7. Performance comparison for bidirectional traffic.



(a) (b)

Fig. 8. Comparison between downstream and upstream: (a) Average delay and (b) delay STD.

mode activated. In addition to the proposed system, the BEB scheme ($W = 32$, $m = 5$, $R = 6$) was also included in the numerical comparisons.

Fig.6 shows the maximum number of acceptable stations for the two schemes with different delay bounds. The delay of a tagged station is caused by the transmission of other stations during its backoff process. The maximum duration used by a station each time is $TXOP$. As a result, when the delay bound is relaxed, the network can use a larger $TXOP$ such that the number of acceptable stations increases. However, for the BEB scheme, this effect is not evident because the collision probability also increases with the network size. When the collision probability becomes large, a station needs to retry many times and the frame delay increases. In contrast, because the proposed tuning procedure controls the collision probability near the proper operation point, the maximum number of acceptable stations increases with the required delay bound for both the FCWB scheme. The advantage of the FCWB scheme becomes significant for large delay bounds.

We conducted a simulation for $N = 15$ to demonstrate the benefit of the proposed super mode. Because $N = 15$ is only valid for the FCWB scheme according to Fig. 6, the BEB without $TXOP$ scheme is selected for performance comparison. In this experiment, each STA has one upstream flow and one downstream flow. Again, the flows of STA 1 are the tagged flows with the data rate fixed at 200 kbps. Fig. 7 shows the plots for the throughput and the violation probability for both downstream and upstream. The traffic asymmetric problem for the BEB scheme is very serious because the upstream traffic consumes dominate capacity. Although the violation probability of the BEB scheme at upstream is very small, many downstream packets are dropped by the BEB scheme. Because the proposed super mode guarantees that the downstream traffic gains one transmission opportunity after the AP receives an upstream burst, the traffic asymmetric problem can be totally solved. In addition, the round robin scheduler in the AP equally allocates the downstream capacity among downstream flows. As shown in Fig. 8, the average delay and the delay STD of the downstream tagged flow are insensitive to the volume of input traffic. At high loads, the delay STD at upstream is about 6 times larger than that at downstream. As a result, the violation probability at downstream is 0 for all cases. That is why we only check if the upstream performance is met in the admission control.

VI. CONCLUSION

This paper has proposed a complete design for the contention-based WLAN MAC to support the station-level QoS guarantee. The design includes the FCWB scheme, a parameter tuning procedure, a super mode for the AP, and an admission control algorithm. Simulation results show that the proposed system has a larger capacity than the traditional BEB scheme when the delay bound requirement is considered. The major advantage of the FCWB scheme is that it reduces the delay variation. Because the contention window is not doubled, the collided frames are not punished and have an equal chance of being transmitted. Moreover, the wider operational range of FCWB allows us to use the same operating point no matter how the network is configured.

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