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A New MAC Protocol with Pseudo-TDMA Behavior for Supporting Quality of Service in 802.11 Wireless LANs

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A new medium access control (MAC) protocol is proposed for quality-of-service (QoS) support in wireless local area networks (WLAN). The protocol is an alternative to the recent enhancement 802.11e. A new priority policy provides the system with better performance by simulating time division multiple access (TDMA) functionality. Collisions are reduced and starvation of low-priority classes is prevented by a distributed admission control algorithm. The model performance is found analytically extending previous work on this matter. The results show that a better organization of resources is achieved through this scheme. Throughput analysis is verified with OPNET simulations.

1. INTRODUCTION

As wireless connectivity rapidly becomes a necessity, new protocols arise in order to cover certain flows of the old ones. In 1997, the first IEEE protocol, 802.11, proposed among others the distributed coordination function (DCF), a means of organizing access in a common medium in a distributed manner. Eight years later, the need for quality-of-service support is guiding the creation of an improved version called enhanced distributed coordination function (EDCF) under the 802.11e protocol, which was later called EDCA (enhanced distributed channel access). In this protocol, high-priority applications access the channel with greater probability. A thorough description of how this is achieved is found in [1, 2]. However, [3, 4] showed that in heavy-load cases, mobile stations have extremely low-probability of transmiting low-priority traffic when using EDCA, an effect called starvation of low-priority applications. The quality of high-priority classes is guaranteed in exchange of total surrender of low class quality.

Knowledge of 802.11 and 802.11e is assumed in this paper [5, 6]. Detailed overviews of this matter are found in [1, 7]. The 802.11 protocol utilizes a carrier sense multiple access with collision avoidance (CSMA/CA) technique. In the standards, two schemes are defined: point coordination function (PCF) is controlled by a central point called access point, whereas in distributed coordination function (DCF) the management is distributed in every node of the network. PCF, despite providing better quality, can only be used in infrastructure-based networks and due to the need for synchronization, has proved to be unreliable in some cases [8]. On the other hand, DCF has become the preferred MAC function due to versatility. Great effort has been put into improving the performance of DCF as regards throughput [9, 10], delay [11] and quality of service [12, 13]. Most of the proposed schemes and algorithms are focused on two settings, namely, the arbitrary interframe spacing (AIFS) and the contention window (CW). These settings are used by the mobile station in order to differentiate from the rest of the contenders and access the common channel.

Multimedia applications have been proved prone to end-to-end and jitter delay, a usual deficiency of packet-switched networks. On the other hand, circuit-switched networks offer great-quality support for multimedia services but they are abandoned due to their inferiority to packet-switched networks in providing data applications. Research in wireless ATM networks [14, 15] has shown that multimedia applications quality can be well-supported in packet-switched networks by means of TDMA schemes where an occasional access is guaranteed in one slot of every frame. The proposed model is designed to create a virtual TDMA environment in a system with variable packet length. The virtual timeslots are used in order to offer guaranteed service for high-quality classes and a reserved bandwidth for low-priority classes by means of an admission algorithm. The model is designed to be backward compatible with the other 802.11 protocols.

The rest of the paper is organized as follows. In Sections 2 and 3 the proposed model is described and evaluated, respectively. In Section 4, results are presented, and in the final section, the conclusion is discussed.

2. SYSTEM MODEL

The proposed model is a compatible enhancement to 802.11e protocol for quality of service using three already-defined access classes (AC0, AC1, and AC2) and alternating-priority access class (AC3). The design goal is to offer priority access to AC3, to prevent unfairness problem from occurring, and to guarantee low delay. The model is based on the functionality of a local timer. A virtual frame duration *Fd* is decided before the operation of the network. *Fd* can be decided separately for each physical (PHY) layer protocol, and it statistically defines the duration of a virtual frame that contains virtual timeslots. *Fd* should match the application requirements for the delay between successive packets. Since the duration of a virtual timeslot is bound by the transmit opportunity (TxOP) property of 802.11e, and the bandwidth and delay requirements of multimedia applications are known by the RTP protocol, the calculation of *Fd* is relatively easy. For a VoIP example with a codec 20 ms, *Fd* should be 20 ms as well.

Using this local timer, every mobile station that transmits priority class information can self-organize the manner in which it transmits. Specifically, for priority classes only, Figure 1 depicts a state diagram of the MAC functionality. Each state contains different values of AIFS and CW that the station uses to access the channel for this application. It is evident that during state 1 the application request is in admission condition where it contends with all other requests. After the admission, the station occupies two basic states. In state 2, it refrains from transmission for as long as an *Fd* counter runs. An interrupt from the timer leads to state 3 where low AIFS and CW guarantee channel access. Certain issues remain to be discussed: the possible collisions of priority class, and the admission and blocking issue.

Collision between low-priority calls is normally dealt with as in DCF. Thus the point of interest is a possible collision of a high-priority request. Specifically, this is divided into two cases: a collision between two ongoing high-priority calls and a collision between high-and-low-priority calls.

2.1. Collision of two high-priority requests

Assume *Application 1* successfully transmits at zero time and *Application 2* immediately follows after packet duration *pd*. When the timer of *Application 1* expires at *Fd*, *Application 1* will try to transmit the next frame. As shown in Figure 2, there is a chance that a low-priority application may have just started to transmit a full TxOP long packet transmission.

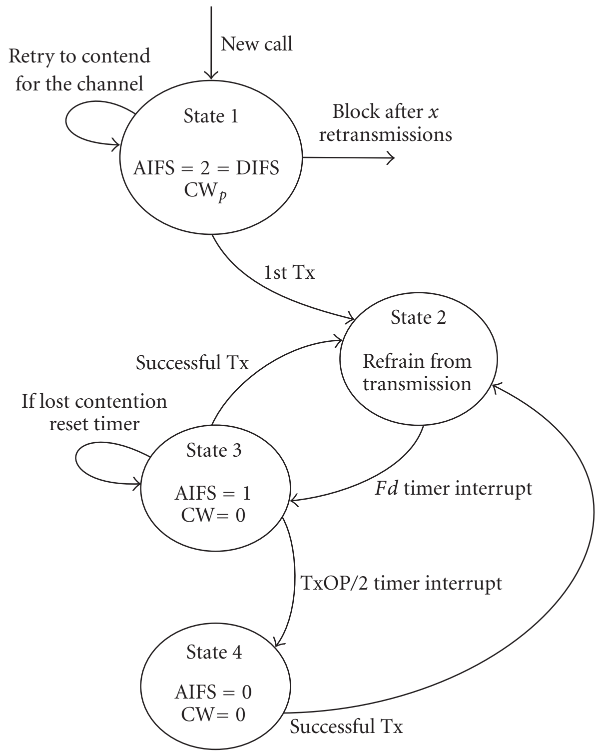

Figure 1: State diagram for priority class AC3.

Collision between high-priority applications can only occur in the case of accumulation of expired timers as in this case. This happens because *pd* will probably be smaller than TxOP and both *Application 1* and *Application 2* will be ready for transmission at *Fd* + TxOP (seconds). To avoid such a misfortunate occasion, we define a fourth state (state 4) in which the priority application hops to when it has already waited in state 3 for TxOP/2 (seconds). This extra state ensures an order between high-priority applications and enables a first-in-first-out (FIFO) functionality of the high-priority contention queue. This ensures collision-free behavior assuming that all transmissions of high-priority class are longer than TxOP/2 (seconds) and no transmission is greater than TxOP (seconds).

Although collisions due to MAC protocol are avoided, there is always a chance that the packet is not accepted correctly due to unpredictable behavior of wireless environment. In case a packet is lost, it is not retransmitted. This UDP-like behavior is in accordance with the TDMA-like nature of the proposed protocol.

2.2. Collision between a high-priority class and a low-priority class

Low AIFS and CW values ensure that no collision can occur between a high-priority class and a low-priority class when they contend simultaneously for the channel after a busy period. However, many consecutive idle slots can allow a low-priority call to collide with a high-priority one. For this to happen, one low-priority terminal should transmit

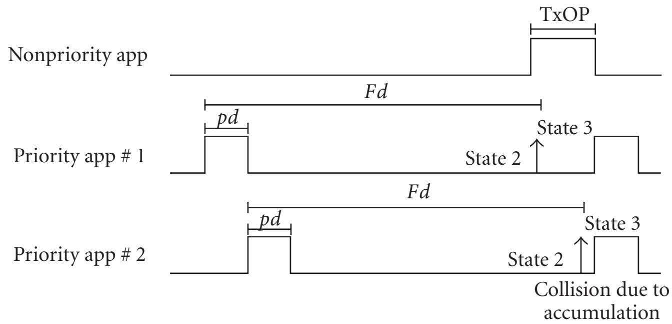

FIGURE 2: Collision between two high-priority applications when state 4 is not used.

in a slot which follows AIFS $[w]$ idle slots together with a high-terminal. The high-terminal transmission slots are very few in an Fd frame, and in case of high-priority traffic conditions, there is a tendency for high-priority transmissions to appear in groups (thus there is no room for consecutive idle slots). The high bound of the probability of collision between high-priority and low-priority classes is then calculated by assuming three independent events: the probability of high-priority transmission in a slot, the probability of AIFS $[w]$ consecutive idle slots, and the probability of low-priority transmission in the same slot:

$$p_c^{3,w} = \sum_{w=0}^{2} \frac{N_3 \cdot \text{Slot\_Time}}{Fd} \cdot (P_i)^{\text{AIFS}\,[w]} \cdot N_w \cdot p_{T_w}, \qquad (1)$$

where $p_c^{3,w}$ is the probability of interclass collision, N_3 the number of terminals demanding high-priority applications, Slot_Time the duration of a single slot, P_i the probability of the channel to be idle in a single slot, $w \in \{0,1,2\}$ the number of the low-priority access class (AC), AIFS $[w]$ the respective arbitrary IFS, N_w the number of terminals demanding such traffic, and p_{T_w} the probability that a terminal is accessing the channel in a specific slot for a packet of the specific AC. All these should be more clear after the following section.

The probability $p_c^{3,w}$ is shown in Figure 3 for the case of $N \in [1,10]$ simultaneously transmitting terminals with all four possible applications, AC0, AC1, and AC2 from 802.11e, and the AC3 modified according to the proposed scheme. For the case of 10 simultaneously transmitting terminals, a ratio of 1 collision per 160 seconds and 1 lost packet per 40 seconds can be estimated. These values show that the collision between low-priority classes and high-priority classes is kept very small and cannot deteriorate the functionality of the proposed model.

2.3. Admission and blocking

The Fd counter implies a fixed virtual frame length. This fixed length is very important for the quality of ongoing transmissions since an increase in the frame would cause a deterministic amount of delay in the system. This also implies that an admission strategy is necessary for preventing the system from overloading. Low-priority applications are somewhat led through an admission procedure when

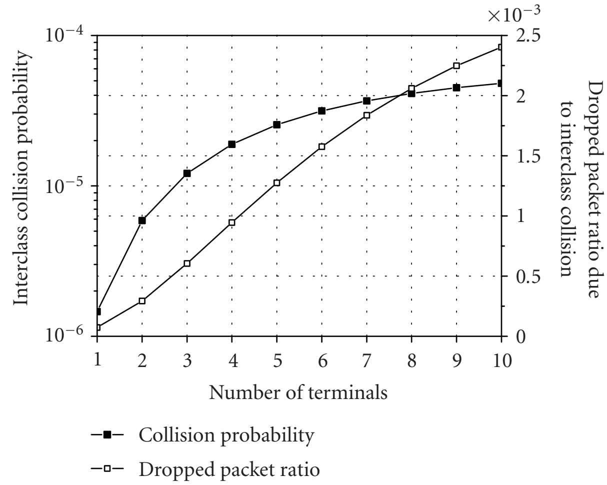

FIGURE 3: Probability of collision between high-priority and low-priority applications.

contending for access with backoff counters. For high-priority applications, a statistical admission is used. The call to be admitted senses the channel and makes x attempts to transmit. After x collisions the application becomes blocked. Blocking may also occur from the connection delay before the x retransmissions take place. AIFS setting is set equal to nonpriority case and CWp is set smaller. The result is that high-priority classes are easily admitted when the load is small. As the load increases, the blocking probability increases as well. If the remaining virtual slots are few, the connection probability will be very small due to high connection delay. Blocking probability and fairness for high-priority class are governed by x, CWp of state 1, and the number of nonpriority and priority contending mobiles. Figures 4 and 5 show some results on this matter.

As the number of mobile terminals increases, connection delay and blocking probability both increase. This increase depends on two separate effects: the available bandwidth and the number of contending mobiles. This is clearly shown in the case of connection delay. Since 14 is the maximum number of possible high-priority applications for 1 Mbps selected transmission rate [16], from that point and after, connection delay depends only on backoff contention. The analytic

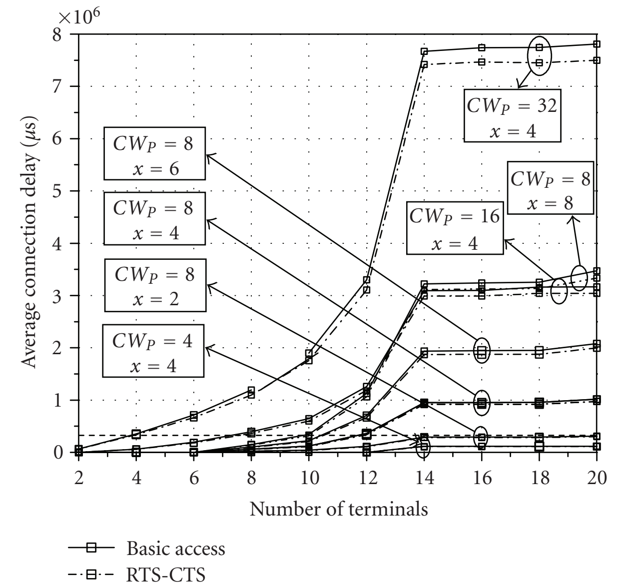

Figure 4: Connection delay for high-priority class, $CWp = [4, 8, 16, 32]$ and $x = [2, 4, 6, 8]$. Transmission rate is 1 Mbps.

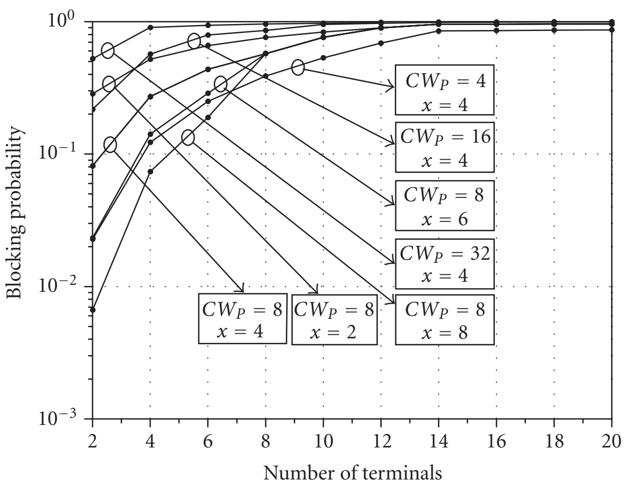

Figure 5: Blocking probability for high-priority class, $CWp = [4, 8, 16, 32]$ and $x = [2, 4, 6, 8]$. Transmission rate is 1 Mbps.

approach for blocking probability and connection delay is found in the next section. The values $CWp = 8$ and $x = 4$ are chosen in the rest of the analysis. Small CWp gives priority and smaller connection delay, but results in channel monopoly by the priority class. A large x reduces blocking probability, but it also causes greater connection delay and monopolization of channel. Smaller connection delay can be achieved by using smaller AIFS values as well.

3. MATHEMATICAL ANALYSIS

3.1. Analysis for nonpriority classes

Low-priority classes (AC0, AC1, and AC2) are treated separately from high-priority ones since they follow different state transitions. The analysis found in [17] is the basis of the one to be used here. In [18], Ziouva proposed a modified analysis considering freezing backoff counters, while in [19] the previous is applied on 802.11e. In our analysis we incorporate the recent findings of [20].

The Markov chain to be used can be found in [20, Figure 2]. For the solution of the Markov chain we assume that priority application admission procedure has a negligible influence on nonpriority access. The backoff procedure is normally analyzed and the bandwidth reduction due to priority transmissions is only taken into account in throughput and delay analysis. If $b_{i,j,k}$ is the stationary probability of backoff state i, j, k, we can solve the system of equations for $b_{1,0,0}$. The system of equations can be found in [20], as well.

We define the probabilities of accessing the channel after a busy period $\tau_{b,w}$ and after an idle period $\tau_{i,w}$ and the probabilities of an idle (busy) slot after a busy period q_0 ($p_{0,w}$) and after an idle period q_1 ($p_{1,w}$). Probabilities $\tau_{b,w}$, $\tau_{i,w}$, $p_{0,w}$, and $p_{1,w}$ are dependent on each specific class while q_0 and q_1 are the same for all low-priority classes, as in [19].

The probabilities of idle channel P_i, of each class successful transmission $P_{s,w}$, and of collision P_c are all defined in a free-from-priority contention slot and found in [20] as well. These probabilities are valid only for the proportion of bandwidth left free from high-priority access called $1-BWp$. BWp is the percentage of resources occupied by priority applications. A simple approach to BWp is

$$BWp = N_a \frac{pd}{Fd}. \tag{2}$$

N_α is the number of successfully accepted calls to the system and pd is the total duration of a high-priority application transmitted packet. Normalized throughput for nonpriority application w will be

$$S_w = P_{s,w}E\{P\}(1-BWp)/\left\{\left(P_i \cdot \text{Slot\_Time} + \sum_{w=0}^{2} P_{s,w}T_s + P_cT_c\right) \times (1-BWp) + BWp \cdot pd\right\}, \tag{3}$$

where $E\{P\}$ is the expected length of a nonpriority class packet, T_s is the average time that a successful transmission of a packet takes, and T_c is the average time that the channel is captured due to a collision, all found in [17].

The average delay for the class w will be

$$E\{D\}_w = E\{N\}_w(E\{B\}_w + T_c + T_T) + E\{B\}_w + T_s, \tag{4}$$

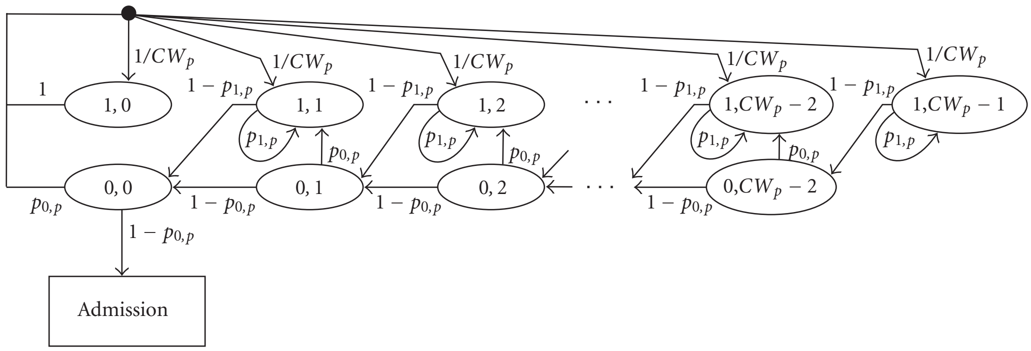

Figure 6: Backoff state diagram for high-priority traffic at admission time.

where $E\{N\}_w$ is the average number of retransmissions, $E\{B\}_w$ is the average delay between the transmissions due to backoff and freezing, T_T is the timeout duration after a collision, $E\{X\}_w$ is the delay of backoff slots, $E\{N_F\}_w$ is the average number of backoff freezing occurrences for each transmission, and BD_w is the average number of backoff counters to be reduced until the transmission. Equation (4) can be solved using (5)-(6):

$$
\begin{aligned}
E\{N\}_w &= \frac{1}{P_{s,w}} - 1,\\
E\{B\}_w &= E\{X\}_w + E\{N\}_w\\
&\quad\times\left((1-BWp)\left(\sum_{w=0}^{2} P_{s,w}T_s + P_cT_c\right) + BWp\cdot pd\right),\\
E\{X\}_w &= BD_w \times \text{Slot\_Time},\\
E\{N_F\}_w &= \frac{BD_w}{\max(\text{ConIdleSlots}, 1)},
\end{aligned}
\tag{5}
$$

where ConIdleSlots is the number of consecutive idle slots between each two backoff freezing occurrences, defined as

$$
\begin{aligned}
\text{ConIdleSlots} &= \frac{P_i(1-BWp)}{1-P_i(1-BWp)},\\
BD_w &= \sum_{j=0}^{m}\sum_{k=0}^{W_j-1} k\cdot b_{0,j,k}\\
&= b_{1,0,0}\sum_{j=0}^{m}\psi_j\frac{W_j(W_j-1)(W_j-2)}{3},
\end{aligned}
\tag{6}
$$

where $W_j = 2^j W_0$, ψ_j is multivalue function defined in [20], m is equal to $\log_2(CW_{\max}/CW_{\min}) - 1$. m, and W_0 depend on the class specifications. The average durations of several cases frames T_s and T_c for basic and RTS-CTS access are found in [16, 17].

3.2. Analysis for priority class

The throughput and delay analysis for priority class is much simpler than for nonpriority class as long as it is assumed that no hidden terminal effect exists. Throughput is given by

$$S_3 = N_a\frac{E\{P\}_p}{Fd + \text{TxOP/R}}, \tag{7}$$

where $E\{P\}_p$ is the expected packet length in bits per frame and R is the channel rate. The average delay will be

$$E\{D\}_3 = (F_{PA} - 1)Fd + \frac{\text{TxOP}}{2}, \tag{8}$$

where F_{PA} is the packet accumulation factor indicating how many high-priority packets are needed to be accumulated in a large packet that is longer than TxOP/2. The first part of the expected delay is a deterministic delay imposed by the packet accumulation. The second part is the expected value of a uniform random variable of how long a priority call may wait in states 3 and 4. Expected delay is independent of the load of the system.

Connection delay can be found with an analysis similar to EDCA as in [19]. The Markov chain for high-priority admission will be a simple chain with CWp, backoff stages with equal probability of selection and stages for freezing of backoff counter (Figure 6). The stationary probabilities are

$$
\begin{aligned}
b_{0,j} &= (CWp - 1 - j)b_{1,0}, \quad \text{for } j \in [1, CWp-2],\\
b_{1,j} &= \frac{1 + p_{0,p}(CWp - 1 - j)}{1 - p_{1,p}}b_{1,0}, \quad \text{for } j \in [1, CWp-1],\\
b_{0,0} &= b_{1,0}\frac{CWp - 1}{p_{0,p}}.
\end{aligned}
\tag{9}
$$

Using $b_{0,0} + \sum_{j=1}^{CWp-2} b_{0,j} + \sum_{j=1}^{CWp-1} b_{1,j} + b_{1,0} = 1$, the stationary probabilities can be calculated:

$$
\begin{aligned}
b_{1,0} = \Bigg[1 &+ \frac{(1-p_{0,p})(CWp-1)}{p_{0,p}} + \frac{CWp(CWp-1)}{2}\\
&+ \frac{CWp - 1 + p_{0,p}\frac{(CWp-1)(CWp-2)}{2}}{1-p_{1,p}}\Bigg]^{-1},
\end{aligned}
\tag{10}
$$

where p indicates that AC3 class is in admission state. The probabilities of channel access for priority admission are

$$\tau_{i,p} = \frac{b_{0,0}}{(q_{1,p}/(1 - q_{0,p} + q_{1,p}))},$$
$$\tau_{b,p} = \frac{b_{1,0}}{(1 - q_{1,p}/(1 - q_{0,p} + q_{1,p}))}, \tag{11}$$

where

$$q_{0,p} = \prod_{w=0}^{2} (1 - \tau_{i,w})^{N_w} (1 - \tau_{i,p}) \simeq q_{0,w},$$
$$q_{1,p} = \prod_{w=0}^{2} (1 - \tau_{b,w})^{N_w} (1 - \tau_{b,p}) \simeq q_{1,w}, \tag{12}$$
$$p_{0,p} = 1 - q_{0,w},$$
$$p_{1,p} = 1 - q_{1,w}.$$

Equations (12) show that the behavior of admission is very much depended upon low-priority access conditions, which in cases of heavy loaded channels prevents the phenomenon of resource starvation of low-priority class.

Further, the average number of backoff slots for every connection attempt can be found:

$$BD_p = \sum_{j=0}^{CWp-2} j \cdot b_{0,j} = \frac{b_{1,0}}{6}(CWp - 1)(CWp - 2)(2CWp - 3). \tag{13}$$

The average delay for every attempt to connect will be

$$E\{CD\}_1 = BD_p \cdot \text{Slot\_Time} + \left(\frac{BD_p}{P_i} - 1\right) \times \left[(1 - BWp) \times \left(\sum_{w=0}^{2} P_{s,w} T_s + P_c T_c\right) + BWp \cdot pd\right]. \tag{14}$$

The probability of successfully accessing the channel is

$$P_{a,p} = \left(P_i q_{0,w} \tau_{i,p} + (1 - P_i) q_{1,w} \tau_{b,p}\right)(1 - BWp). \tag{15}$$

The average connection delay, disregarding the calls that will drop due to extensive delay, is

$$E\{CD\} = P_{a,p} E\{CD\}_1 \sum_{l=1}^{x} l (1 - P_{a,p})^{l-1}. \tag{16}$$

Defining a threshold of acceptable connection delay Thr_{CD}, the fact that a priority demand will be blocked due to unacceptable delay will cause less number of retrials ($\tilde{x}$) and greater blocking probability. We calculate blocking probability as

$$P_B = (1 - P_{a,p})^{\min\{x, [\max(\tilde{x}) | CD \le \text{Thr}_{CD}]\}}. \tag{17}$$

Table 1: Simulation values.

802.11e values			
	$CW_{\min}$	$CW_{\max}$	AIFS
AC3	3	7	2
AC2	7	15	2
AC1	15	1023	3
AC0	15	1023	7
Saturation traffic			
Packet length	1024 B		
Interarrival time	0.01 s		
VoIP traffic			
Packet length	4 × 160 B		
Interarrival time	0.08 s		

4. RESULTS

In this section we calculate the behavior of the proposed model in comparison with EDCA. The performance of the MAC protocols is tested for variable number of contending stations, basic and handshaking access, and the several access classes. Every terminal is assumed to demand all four classes of access. In case of saturation analysis, it is assumed that a packet is always available for transmission. For AC3 of the proposed protocol, admission control is utilized to prevent the system from overloading. Since the demand is great, the throughput is found to be saturated. On the other hand, we present results where AC3 demand is not saturated. Every terminal initiates a VoIP call and saturated traffic for the rest of the classes.

Voice-over-IP (VoIP) applications use the G.711 codec [21]. Every 20 ms, 160 B of payload are transmitted . The Fd timer could be set to 20 ms for this case. However, the packet length (in bytes) needs to be greater than TxOP/2. Thus, a 4-packet accumulation is proposed before transmission, which yields a maximum of 60 ms buffer delay. This deficiency is necessitated by the priority class collision avoidance mechanism proposed in Section 2.1. Fd is then chosen to be 80 ms and the packet payload would be 640 B.

The analytical approach is compared with simulations with the OPNET simulator. The simulation values are described in Table 1.

Figures 7 and 8 show the results for saturation throughput in case of basic and RTS-CTS access, respectively. A clear advantage of the proposed model is obvious in case of basic access. However, this gain is compromised by the use of RTS-CTS, which indicates that EDCA results in more collisions. A small gain in throughput remains, which is expected from the fact that the proposed model uses the TDMA scheme.

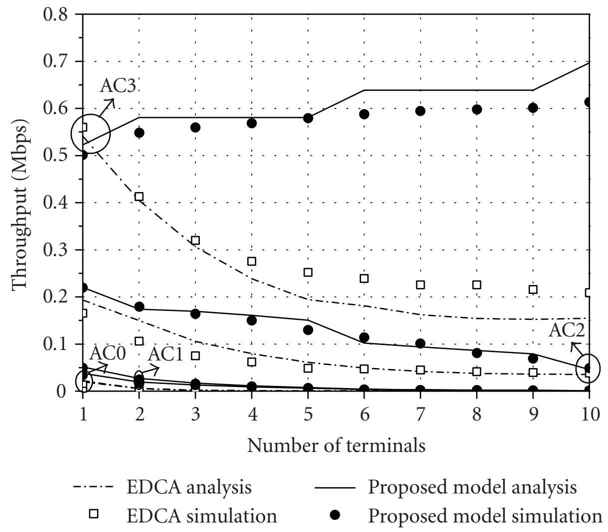

Figure 7: Saturation throughput for 1 Mbps channel rate and basic access.

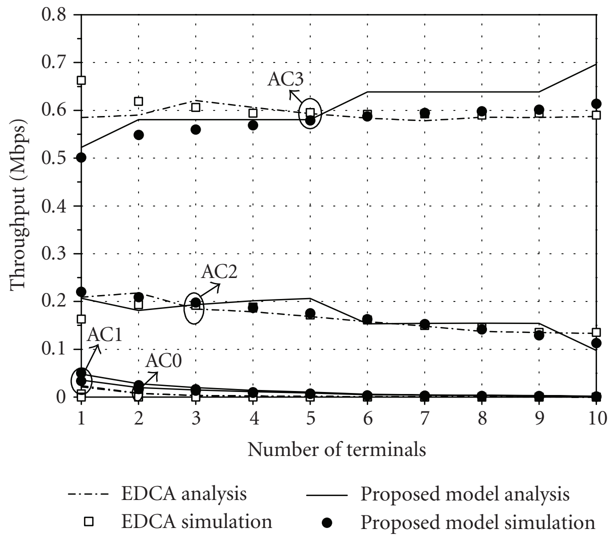

Figure 8: Saturation throughput for 1 Mbps channel rate and RTS-CTS access.

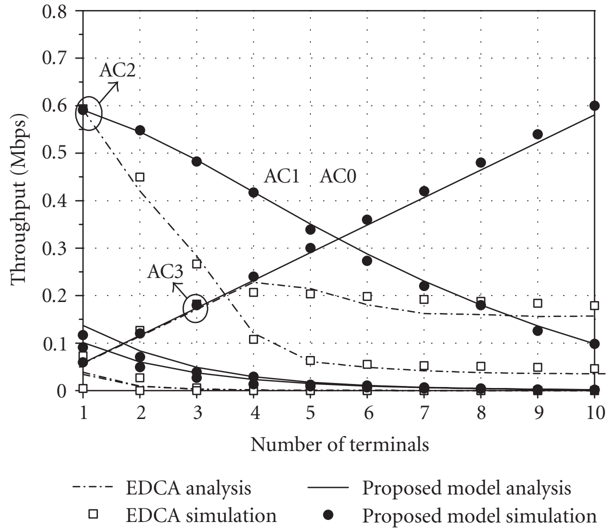

Figure 9: Throughput for nonsaturated AC3 traffic, 1 Mbps channel rate, and basic access.

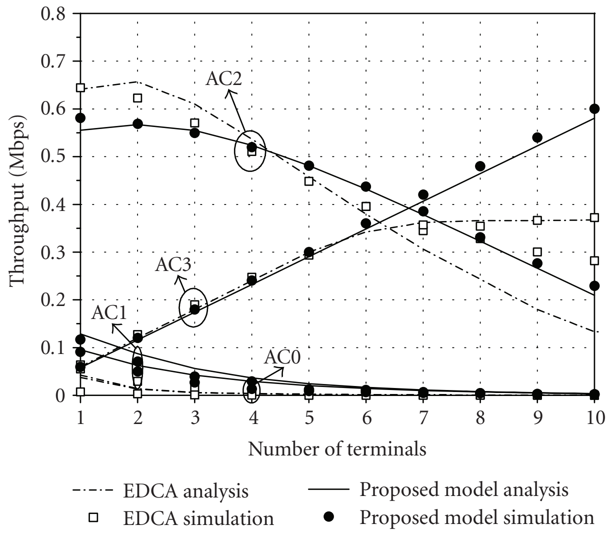

Figure 10: Throughput for nonsaturated AC3 traffic, 1 Mbps channel rate, and RTS-CTS access.

In terms of simulation and analysis comparison, the proposed model throughput is found to be a little worse in simulations, which is partially explained by the interclass collisions and the way admission control is used in analysis. In Figure 7, an unexpected difference between analysis and simulation for AC3 is shown.

Figures 9 and 10 show throughput for the case where AC3 traffic is not saturated. Specifically, one one-way VoIP application is considered to be generated by each of the terminals. The rest of the traffic sources are considered saturated. These conditions showcase the performance of the two protocols in realistic conditions. The proposed model is found to be superior in terms of throughput at most of the times. AC3 for EDCA is saturated earlier than expected due to the smaller packet length used. It can be seen that the proposed protocol performance for high-priority class is unaffected by the packet length as long as it remains larger than TxOP/2.

A deviation between AC2 analysis and simulation is found in this case. For better comparison, packet accumulation is used for EDCA and the throughput is kept high having a negative effect on delay (Figure 12). Admission control is not activated in this case.

Average medium delay is shown in Figures 11 and 12 for the case of saturated and nonsaturated AC3 traffic, respectively. A small gain is found in terms of delay for the proposed protocol high-priority traffic. However, an important

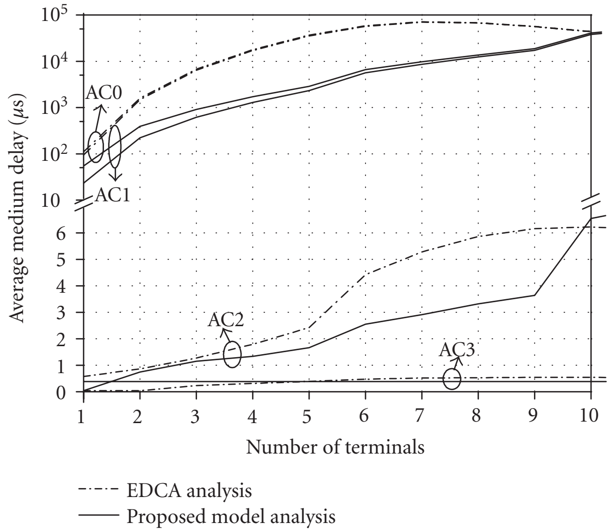

Figure 11: Average medium delay for 1 Mbps channel rate.

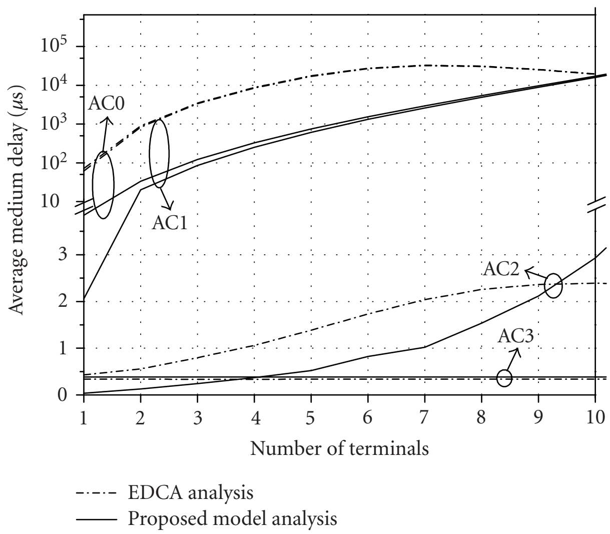

Figure 12: Average medium delay for 1 Mbps channel rate and non-saturated AC3 traffic.

characteristic is that the proposed protocol yields very small jitter delay. This is analogous to the TDMA performance.

Low-priority delay, on the other hand, can be very high when the high-priority applications occupy the greater portion of the available bandwidth. If an improvement is required on this matter, CWp can be modified to perform a tighter admission control for high-priority calls, nevertheless leading to higher blocking probability.

5. CONCLUSION

A new MAC protocol is proposed to be a backward compatible advancement to the wide-known 802.11e protocol. A timer called Fd timer is used in a distributed manner from each wireless terminal to create a virtual TDMA-like frame. Each terminal uses another timer to prevent collisions with other high-priority applications. A tradeoff between high-priority admission characteristics (connection delay and blocking probability) and low-priority performance can be used in quality-of-service optimization procedure. The results show a small improvement in throughput due to the decrease in the backoff delay. The average delay for priority class is independent of load conditions, as expected by the TDMA nature of the proposed protocol, thus making the proposed protocol ideal for VoIP communications. Other advantages of the proposed protocol are the small jitter delay and the independence of throughput from the packet length.

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