Abstract—Two Admission Control schemes are investigated for wireless Local Area Networks (WLANs) in the context of the 802.11 standards. Every node handles the admission rules locally and the system can operate in ad hoc mode. Specifically, two possible admission mechanisms are compared. The terminals operate in admission state when accessing the channel for the first time and according to the number of collisions or the delay characteristics, block themselves in order to prevent the system from saturation. The second method is proved to provide a tighter admission performance. The analysis and simulation presented in this paper, show the improvement in quality when Voice over Internet Protocol (VoIP) applications contend for the common channel. The advantage of the proposed schemes is that they can be used without the need of an Access Point.

I. INTRODUCTION

Since 1999 when IEEE announced the 802.11 standards for WLANs [1], a lot of research activity has been focused on performing Quality of Service (QoS) [2] in such networks. The need, for supporting multimedia services, is driving a great effort in research community due to the fact that the wireless channel proves to be unreliable, as regards stringent delay demand services. To prevent the system from collision and overloading, both of which result in degradation of quality levels, the use of admission control is proposed in the literature by many, for example [3], [4] and [5].

A thorough overview on admission control in WLANs can be found in [6]. This overview classifies admission control into many categories and refers to a number of previous works on this matter. However, all these works assume the existence of an access point as a prerequisite for the admission control even when referring to distributed access. The access point is always responsible for making the measurements and processing the channel state in order to broadcast the rules of admission.

In the context of ad hoc mode admission control in the MAC layer (layer-2), Gu and Zhang [7] proposed an algorithm based on traffic load measurements using the Relative Occupied Bandwidth or the Average Collision Ratio as measures. By means of simulation they proved that the first performs better, but both of their proposed schemes showed extensive delay, although incorporating 802.11e. In [8] there is an example of ad hoc admission control performed in the layer-3. A lot of work can be found in this matter as well.

In this paper we propose a MAC layer admission algorithm to be used in ad hoc mode. According to [5], [6] and many other authors, these admission mechanisms must be simple and mostly protect the channel from overloading. Our models provide backward compatibility to already installed wireless networks by proposing minor alternations comparing to [1]. The performance is both analysed and simulated via Opnet Simulator.

In the rest of the paper basic knowledge of [1] and [2] is assumed. An extended overview of the standards can be found in [9]. In section II the proposed schemes are explained, in III analyzed and in IV simulated. In section V, the paper is concluded with some insights to future work.

II. PROPOSED SCHEMES

Two simple admission schemes are proposed. In the subsequent sections the two-state model and the delay-based model are explained.

A. Two-state Model

The first proposed model is based in the two-state concept, the admission state and the normal operation state, which are presented in the transition diagram of figure 1. When a new call is ready to send the first packet, it switches to the admission state by using the respective access parameters, minimum Contention Window ($CW_{min}$) equal to $CW_{ac}$ and
Arbitrary Interframe Spacing (AIFS) equal to $AIFS_{ac}$ (Note that the term AIFS is used here instead of DIFS since it refers to an IFS that is arbitrarily set for the needs of the proposed admission algorithm). The terminal attempts to access the channel with these settings. If it succeeds to send, the first packet is automatically accepted.

Therefore the admission control is based on the first packet to transmit for every application. The application is blocked after an idle (a busy) period and at least another terminal transmits and no terminal transmits of state $f$. According to the same paper, the admission algorithm does not affect the analysis of the backoff counter markov analysis initially proposed by Bianchi in [10] and the improvement proposed by Foh et al. in [11] and by Paschos et al in [12]. Figure 2 from [11] is assumed to describe the states of the admission state and $N_{ac}$, where $N$ is the number of terminals in normal state and $N_{ac}$ the number of terminals in admission state, which is the case for most of the time and especially for great values of $N$.

The analysis is based on the backoff counter markov analysis. The respective probabilities are, at least one terminal transmitting after an idle period $p_{ac}(0)$, no terminal transmitting $q_{ac}(0)$ and accessing the channel $\tau_{b,ac}$ ($\tau_{b,ac}$) Analysing the chain we get:

$$s_{0,0} = s_{1,0} (CW_{ac} - 1)$$
$$s_{0,j} = (CW_{ac} - 1 - j) s_{1,0}, j \in [1, CW_{ac} - 2]$$
$$s_{1,j} = \frac{1 + p_{ac}(CW_{ac} - 1 - j)}{1 - p_{ac}} s_{1,0}, j \in [1, CW_{ac} - 1]$$

Regarding the probabilities of transmission:

$$q_{0,ac} = q_0$$
$$q_{1,ac} = q_1$$
$$p_{0,ac} = 1 - q_0$$
$$p_{1,ac} = 1 - q_1$$

Equation 2 shows that the admission is based on the channel condition (described by $q_0, q_1$). The average number of backoff slots for every connection attempt will be:

$$BD_{ac} = \frac{CW_{ac} - 2}{\sum_{j=0}^{J} j \cdot s_{0,j}} = \frac{s_{1,0}}{6} CW_{ac} (CW_{ac} - 1) (CW_{ac} - 2)$$

where $s_{1,0}$ is found by the following equation,

$$s_{1,0} = \frac{CW_{ac} (CW_{ac} + 1)}{2} + \frac{CW_{ac} - 1 + p_{0,ac}}{1 - p_{1,ac}} \frac{(CW_{ac} - 1)(CW_{ac} - 2)}{2}$$

which is derived by the normalization condition in Figure 2.

The average delay for every attempt to connect will be:

$$E\{CD_1\} = BD_{ac} \cdot \sigma + N_{ac} \cdot P_{ac} \cdot (P_s T_s + P_c T_c)$$

where $\sigma$ is the slot duration, $P_s$ the probability that a transmission is successful, $P_c$ the probability that a transmission
is collided, $T_s$ and $T_c$ the respective durations and $N_F$ the average number of freezes of backoff counter given by:

$$N_F = \frac{BD_{ac}/P_i}{\max\{E[\Psi],1\}} - 1$$  \hspace{1cm} (6)

where $P_i$ is the probability that a slot is idle and $E[\Psi] = P_i/(1-P_i)$ is the average consecutive idle slots between two transmissions. The probabilities $P_a$, $P_c$ and $P_i$ refer to the already admitted terminals and the first two are calculated with respect to non idle slot (so as $P_a + P_c = 1$). The probability of accessing the channel in each attempt, when in the admission mode, is:

$$P_{ac} = P_i q_0 + (1 - P_i) q_1$$  \hspace{1cm} (7)

Finally the average connection delay can be calculated regarding the accepted calls only:

$$E\{CD\} = E\{CD_1\} \frac{\sum_{k=1}^{x_{ac}} k (1 - P_{ac})^k}{\sum_{k=1}^{x_{ac}} (1 - P_{ac})^k}$$  \hspace{1cm} (8)

The blocking probability will be:

$$P_B = (1 - P_{ac})^{x_{ac}}$$  \hspace{1cm} (9)

Figures 3 and 4 show the average connection delay and the blocking probability performance of the two-state scheme in case of channel saturation.

Figure 5 is the outcome of extensive simulation of the model #1. Blocking probability is a statistic difficult to measure in cases of small numbers. $x_{ac}$ is set to 1 and $CW_{ac}$ to 8 and the tightest admission strategy is obtained. Comparing figures 4 and 5, it can be seen that this admission strategy is very loose and cannot protect properly the system. If the admission settings are very tight, then calls can be rejected even when the system has available resources, which of course is very undesirable. For this reason, a delay-based model is proposed.

### B. Delay-based Model

This model is based on a delay threshold $T_{thr}$. While the wireless station remains in admission state, the access delay is monitored and the access is blocked if found greater than the threshold. After the admission period $T_{ac}$ has passed, the station turns to normal mode and continues transmitting uninterrupted. This concept is based on the fact that average MAC delay is proportional to the network traffic. This means, that imposing a specific threshold, the blocking probability will get higher when the traffic is greater. In order to choose effectively this threshold, the delay information must be known for several traffic conditions and number of transmitting terminals. For
this reason we define the traffic intensity normalized with channel rate as $\rho = \frac{\sum \rho_t}{N R}$, where $\rho_t$ is the traffic of each terminal. From opnet simulation, figure 6 is obtained.

An analytical approach of the delay model is beyond the scope of this paper. Using figure 6, an optimal delay threshold is chosen. The desirable effect of admission algorithm is to provide access when the traffic is low and deny it when the channel is overloaded. We set the threshold to the value $T_{thr} = 0.03$ sec, and we simulate the blocking probability relative to the normalized traffic intensity and number of terminals shown in figure 7.

Note that in figure 7, the small values are calculated separately using a much greater number of simulation samples for better precision of blocking probability. From the same figure, the desired phenomenon can be observed. In cases of small traffic load, all attempts are accepted whereas in cases of high traffic, all calls are blocked.

III. PERFORMANCE ANALYSIS

In order to compare the functionality of the two proposed admission schemes with the legacy 802.11 protocol, a specific environment is assumed. A series of 20 VoIP calls is initiated in a wireless channel of 1Mbps data rate and the resulting throughput and delay are measured to provide a Quality measure for each scheme. The VoIP traffic is considered a constant bitrate (CBR) application sending 160B each 20msec resulting in 64kbps G.711 codecs. The results of figures 8-9 are found using the OPNET simulator. As regards the admission settings, model #1 has $CW_{ac} = 32$, $x_{ac} = 2$ and model #2 has $T_{thr} = 0.03$ sec and $T_{ac} = 1$ sec. Such a value of $T_{thr}$ is set so as the proposed admission control to be able to handle effectively the VoIP traffic. In such a case, and since 160B packets generate each 20msec in a G.711 codec [13], VoIP will be transmitted without significant degradation of the quality of speech [14].

In figure 8, the aggregating throughput is showcased. Throughput for the model 2 is always kept below saturation which results in better offered quality. More specifically, when the saturation conditions have been reached (i.e. 8 calls have been accepted) new calls are being dropped. In this random scenario, the stochastic nature of model 1 is showcased as well. The 8th call gets blocked although the saturation condition are not yet reached. The tighter the admission algorithm, the more throughput rate is achieved at high load since we have less amount of collisions. However, the difference is small.

The real gain of admission control is shown in figure 9, where the total MAC delay is monitored in a real-time manner. The proposed models use the admission control to keep the delay lower than in the legacy 802.11 protocol. This is achieved by blocking new calls when the traffic is high. Model 2 provides better quality than model 1 since it keeps
the MAC delay below the given threshold of 30\(\text{m sec}\). When no admission control is used, the system quickly becomes overloaded and the quality of all ongoing calls is severely deteriorated.

IV. FUTURE WORK AND CONCLUSION

Two admission control models are proposed both of which are simple to implement and backward compatible to 802.11 protocol. The advantage of the proposed models is that they can be used in a distributed manner without the need of an Access Point. The second model is found to perform more efficiently providing better Quality of Service for VoIP calls in high traffic conditions. The proposed models can be applied to 802.11e protocol as well. Simulation and analysis for this case are left as future work. Another interesting issue is the analytical derivation of per packet MAC delay. The full model of delay can be used to calibrate the \(T_{\text{thr}}\) more efficiently and to optimize the values.

REFERENCES