Abstract—Cognitive Radio Networks (CRNs) provide a solution for the spectrum scarcity problem facing the wireless communications community. To be able to utilize CRNs in practical applications, a certain level of quality-of-service (QoS) should be guaranteed to the secondary users (SUs) in such networks. In this paper, we propose a packet scheduling scheme that orders the SUs’ transmissions according to the packet dropping rates and the number of packets queued waiting for transmission. A medium access control (MAC) protocol, based on the mentioned scheduling scheme, is proposed for a centralized CRN. In addition, the scheduling scheme is adapted for a distributed CRN, by introducing a feature that allows SUs to organize access to the available spectrum without the need for a central unit. Extensive simulation results are presented to evaluate the proposed protocols, in comparison with other MAC protocols designed for CRNs. The results demonstrate the effectiveness of our proposed protocols to guarantee the required QoS for voice packet transmission, while maintaining fairness among SUs.

I. INTRODUCTION

Wireless communication has become the desired means of communications for many applications, mainly because of the freedom associated with the mobility available for a wireless terminal. In each country, there is an organization responsible for allocating various bands of the radio spectrum to different users, and currently most useful bands of the spectrum have been allocated. However, it has been found out that the allocated spectrum bands are heavily underutilized. According to [1], the assigned spectrum in some cases is only being accessed 15% of the time. As a result of the scarcity and underutilization of the available spectrum, the concept of cognitive radios (CR) has emerged.

The CR concept was first introduced in 1999 [2], based on the architecture of software radios [3], and has been the focus of many research activities since then. Cognitive radio networks (CRNs) are networks where unlicensed secondary users (SUs) scan the available spectrum in order to transmit their information packets in spectrum holes unused by licensed primary users (PUs). In a CRN, the medium access control (MAC) layer has three main functions [1], [4]:

- Locating the unused spectrum, referred to as spectrum sensing;
- Coordinating the transmission of different users, referred to as spectrum sharing;
- Determining the optimal times of spectrum sensing and data transmission.

One way to classify MAC protocols for CRNs is according to the employed access mechanism; for instance, whether the protocol is a random access, a time slotted, or a hybrid protocol [4]. In order to guarantee a certain level of quality-of-service (QoS) for delay sensitive applications such as voice service, as any other telecommunication system, a CRN has to be able to avoid saturating the available channels and to allocate resources for each user according to its service requirements [5]. The saturation of a channel is avoided by employing what is called connection admission control [6], while appropriate resource allocation is known as service differentiation. Furthermore, QoS has different parameters depending on the application, such as delay, jitter, packet loss, signal to interference plus noise ratio (SINR), and bit error rate (BER) [7].

In the literature, there exist MAC protocols designed for CRNs. A carrier sensing multiple access (CSMA) based MAC protocol for centralized CRNs is proposed in [8]. This protocol allows simultaneous transmissions in the primary and the secondary networks by adjusting the transmission rate and power in both networks. However, QoS support for SUs is not addressed. On the other hand, a cognitive MAC protocol (C-MAC) [9], a synchronized MAC protocol (SYN-MAC) [10], a cognitive MAC protocol (CR MAC) with QoS provisioning [7], and QoS for voice CRNs [11] are four MAC protocols that are designed for distributed CRNs. The C-MAC and SYN-MAC do not support QoS in any form, while the CR MAC with QoS provisioning is designed to provide a level of QoS for delay sensitive applications with known message length. Finally, QoS for voice CRN aims to provide SUs with a level of QoS suitable for voice communications. However, fairness among SUs is not maintained.

In this paper, we propose two MAC protocols that schedule SUs’ packet transmissions in a CRN, while achieving the following goals:

- Guaranteeing the packet dropping rate below a certain bound for all SUs,
- Maintaining fairness among all SUs of the same service class, and
- Avoiding interference with activities of users in the primary network.

To overcome the shortcomings of the existing MAC protocols for CRNs, we propose Centralized MAC and Distributed MAC protocols for centralized and distributed CRNs respectively. The protocols are based on scheduling packet transmissions by ordering SUs, first according to their current packet dropping rate and then according to their numbers.
of packets queued waiting for transmission. Both protocols successfully provide QoS suitable for voice communications and maintain better long-term and short-term fairness among SUs than existing schemes.

The remainder of this paper is organized as follows. In Section II, we describe our system model, while two MAC protocols that guarantee QoS support for SUs and maintain fairness among SUs in the CRN are proposed in Section III. Performance evaluation of the proposed MAC protocols is presented in Section IV. Finally, in Section V, we conclude this research.

II. SYSTEM MODEL

In this section, we present our system model. The network structure of the primary and secondary users is described in Section II-A. The voice traffic model is introduced in Section II-B, and the QoS requirements are given in Section II-C.

A. Network Structure

Consider a single-cell primary system that utilizes time division multiple access (TDMA) in its operation. Time is divided into frames of constant duration, and each frame is uniformly divided into timeslots. The number of slots in one frame is equal to the number of primary users, referred to as the capacity of the primary system and denoted by $C$. Each timeslot in a frame is assigned to a primary user. Figure 1 shows a time frame for the primary system with capacity of 3 PUs. The secondary system utilizes the leftover spectrum of the primary system. During a certain frame, if one of the PUs does not have a packet to transmit, its idle timeslot is used by one of the SUs to transmit a packet. For simplicity, we assume that, if a PU is idle for a given frame, its assigned timeslots will be free for the duration of the said frame.

![Frame Diagram](image)

Fig. 1. One frame duration for the primary system with $C = 3$ PUs.

A user, either primary or secondary, occupies only one timeslot per frame, and each user has its own queue with a buffer size equal to the packet delay bound. If a PU has a packet to transmit, the PU waits for its assigned timeslot and transmits its packet. However, if an SU has a packet to transmit, the packet is queued until the SU is granted access to an idle timeslot. Finally, we assume that the functionality of the physical layer is ideal, and we are not concerned with establishing a link between the source and the destination before transmission.

B. Voice Traffic Model

A voice source typically alternates between active and silent periods. During the active period (ON state) the source generates packets at a constant rate, denoted by $V$ packets/frame, while in the silent period (OFF state) the source is idle. It has been found out that the time spent by a voice source in one of the periods can be represented by an exponential distribution [12]. In our model, both primary users and secondary users are voice users. Hence, the time spent by a user in an ON state, $t_{ON}$, is exponentially distributed with parameter $\lambda$, while the time spent by the user in an OFF state, $t_{OFF}$, is exponentially distributed with parameter $\mu$. As a result, $T_{ON} = E(t_{ON}) = \frac{1}{\lambda}$ frames, and $T_{OFF} = E(t_{OFF}) = \frac{1}{\mu}$ frames. The active probability, $P_{ON}$, is defined as the ratio of the time spent in the ON state to the total time of the conversation. Hence, $P_{ON} = \frac{t_{ON}}{t_{ON} + t_{OFF}}$. A typical value for $P_{ON}$ is 0.4.

C. QoS Requirements for Voice Communications

In voice communications, a voice packet has to be transmitted from the source to the destination within a certain time limit, otherwise the packet is considered useless and dropped. We denote this delay bound by $m$ (in frames). The delay bound can be set in various manners, such as a bound on the end to end transmission delay. In our model, with single-hop transmissions under consideration, the delay bound is defined as the maximum duration that a packet can be queuing for, from its generation up to its transmission.

Another requirement for voice packet transmission is the acceptable packet dropping rate, $P_{D}$. This bound indicates the percentage of packets that a voice source can drop out of the total number of packets generated without affecting the quality of the call. Normally, $P_{D}$ for voice communications is set to one percent.

III. PROPOSED MAC PROTOCOLS

A. MAC Protocol for Centralized CRNs

For a centralized CRN, the secondary system has a central unit (CU) that manages the activities of all SUs in the system. In addition, it is assumed that the central unit is aware of the activities of the PUs via spectrum sensing at the beginning of each timeslot. Thus, at the beginning of each frame, the CU is aware of whether or not a certain PU will use its allocated timeslot in the frame.

A MAC protocol, referred to as Centralized MAC I, is designed for a centralized CRN that utilizes the empty spectrum of a primary system. Each SU has its own queue. At the beginning of each frame, SUs will be ordered based on their actual packet dropping rates and then based on the number of packets queued waiting for transmission. According to this order, SUs will have access to the idle timeslots in each frame.

The following is a high level description of Centralized MAC I, for the $n^{th}$ time frame $f_{n}$:

- **Beginning of frame $f_{n}$**
  1. Determine if a new packet will be generated for each SU;
  2. Order SUs according to their current packet dropping rates, in a descending order;
  3. Order SUs with same packet dropping rate according to the numbers of packets queued for transmission, in a descending order;
4) Identify idle timeslots by determining whether or not the respective PUs have a packet to transmit, via spectrum sensing;
5) Allocate idle timeslots to the top secondary users with packets to transmit;
   - Top SU transmits their packets successfully;
6) Determine if any SU has packets being queued beyond the delay bound;
   - If yes, drop packets;
7) Update SU status;

* End of frame $f_n$.

The MAC protocol operates under the assumption that the CU has accurate knowledge of channel (timeslot) availability in the frame by spectrum sensing. The ideal scenario provides a performance benchmark for MAC protocols in CRNs.

### B. MAC Protocol for Distributed CRNs

The major difference between a centralized CRN and a distributed CRN is whether or not there is a central unit in place. Without a CU, we need to introduce a property to the protocol proposed in Section III-A, that allows SUs to access idle timeslots without either colliding with each other and with the PUs. To organize channel access among SUs, we propose utilizing the concept of backoff durations. PUs have pre-assigned timeslots during which they transmit their packets. A PU with a packet to transmit is not required to sense the channel, i.e., all PUs’ backoff duration is zero for transmission in their assigned timeslots. On the other hand, SUs are only allowed to transmit in an idle timeslot. Thus, an SU that wants to transmit a packet has to listen to each timeslot for a certain duration; if the timeslot remains idle after the sensing and backoff duration, the SU transmits its packet.

Since SUs are ordered according to the packet dropping rate followed by the number of packets queued, we propose that each user calculates its backoff duration, taking into consideration the ordering criteria. Let $B$ denote the backoff duration, $P_d$ the actual packet dropping rate, and $Q$ the number of packets queued waiting for transmission. The two main criteria on which access to idle timeslots is granted to SUs are based on $P_d$ and $Q$. Thus, while calculating the backoff duration, each of the SUs has to take into consideration its values of $P_d$ and $Q$. An SU with a high $P_d$ should have access to an idle timeslot before an SU with a low $P_d$, even if the latter has a higher value of $Q$. Thus, the weight of $P_d$ should be higher than the weight of $Q$ in the calculation of the backoff duration. Following is an equation that can be used by each SU to determine its backoff duration:

$$
\frac{1}{B} = (10^{j+4} \times P_d) + (10^j \times Q) + (10^j \times I)
$$

where $I$ is a random variable uniformly distributed in $[0, 1]$, and $j$ is one plus the order of $m$.

In this manner, the SU with the smallest $B$ will have priority to transmit a packet in the next available timeslot. In addition, because of the random variable used, collisions among SUs, with equal values of $P_d$ and $Q$, are avoided. Table I shows examples of the $B$ value for secondary users with various $P_d$, $Q$, and $I$ values.

<table>
<thead>
<tr>
<th>SU</th>
<th>$P_d$</th>
<th>$Q$</th>
<th>$I$</th>
<th>$B$</th>
</tr>
</thead>
<tbody>
<tr>
<td>A</td>
<td>0</td>
<td>1</td>
<td>.542</td>
<td>0.006485</td>
</tr>
<tr>
<td>B</td>
<td>0</td>
<td>2</td>
<td>.127</td>
<td>0.004701</td>
</tr>
<tr>
<td>C</td>
<td>0.005</td>
<td>2</td>
<td>.632</td>
<td>0.00019</td>
</tr>
<tr>
<td>D</td>
<td>0.007</td>
<td>1</td>
<td>.097</td>
<td>0.000114</td>
</tr>
<tr>
<td>E</td>
<td>0.005</td>
<td>2</td>
<td>.278</td>
<td>0.00019</td>
</tr>
</tbody>
</table>

However, as shown in Table I, because of the small differences among the $B$ values, an implementation difficulty may arise. To overcome the problem of small time differences among the $B$ values, we propose to discretize the sensing duration into mini-slots as shown in Figure 2. Each SU maps its $B$ value to a mini-slot identification number, $ID$, to reduce collisions. As an example, the mapping can be done by using

$$
ID = \left\{ \begin{array}{ll}
\text{round}(B \times 10^j), & \text{if } P_d = 0 \\
\text{round}(B \times 10^5) + X, & \text{if } P_d > 0
\end{array} \right.
$$

where $X$ is an integer random variable taking on a value randomly over the interval $[0, 9]$. Note that, Eq. (2) distinguishes between SUs with $P_d = 0$ (i.e., no packet dropping) and SUs with $P_d > 0$ (i.e., with packet dropping). This improves the ordering mechanism and ensures that SUs with a larger $P_d$ transmit before SUs with a smaller $P_d$.

![Fig. 2. Sensing Duration mini-slots.](image)

**Table II**

<table>
<thead>
<tr>
<th>SU</th>
<th>$B$</th>
<th>$X$</th>
<th>$ID$</th>
</tr>
</thead>
<tbody>
<tr>
<td>A</td>
<td>$6.851 \times 10^{-4}$</td>
<td>–</td>
<td>65</td>
</tr>
<tr>
<td>B</td>
<td>$4.014 \times 10^{-4}$</td>
<td>–</td>
<td>47</td>
</tr>
<tr>
<td>C</td>
<td>$19.000 \times 10^{-5}$</td>
<td>2</td>
<td>21</td>
</tr>
<tr>
<td>D</td>
<td>$14.065 \times 10^{-5}$</td>
<td>5</td>
<td>19</td>
</tr>
<tr>
<td>E</td>
<td>$19.129 \times 10^{-5}$</td>
<td>7</td>
<td>26</td>
</tr>
</tbody>
</table>

Table II shows the resulting $ID$s, corresponding to the SUs given in Table I. A disadvantage of discretizing the sensing duration is the possibility of collisions. The value of $B$, calculated by Eq. (1), is in continuous time and thus unique. However, after mapping $B$ to $ID$, two or more SUs can end up using the same mini-slot. To deal with this issue, once a collision occurs, all SUs recalculate their $B$ and $ID$ values. After introducing the access mechanism to organize SUs in a
distributed CRN, next we introduce our proposed protocol for a distributed CRN, referred to as Distributed MAC I.

The following is a high level description of the Distributed MAC I protocol, for an SU active in frame \( f_n \):

- **Beginning of frame** \( f_n \):
  1) Determine if a new packet will be generated;
  2) Calculate the backoff duration \( B \) using Eq. (1);
  3) Map the backoff duration \( B \) to an ID using Eq. (2);
  4) Listen to the channel from the beginning of timeslot to the mini-slot corresponding to the ID;
     - If the channel is idle, transmits a packet;
     * If collision occurs, recalculates \( B \) and ID and waits for next timeslot;
     - If channel is busy, waits for next timeslot;
  5) Determine if any packets have been queued beyond the delay bound;
  6) Update status;
- **End of frame** \( f_n \).

IV. PERFORMANCE EVALUATION

A. Simulation System Setup

MATLAB is chosen as the simulation environment for our evaluation. We start by creating a set of users. For each user, we choose a value randomly from 0 to 1. If the value is less than or equal to \( P_{ON} \), the user starts in the ON state; otherwise, the user starts in the OFF state. Once the starting state of the user is determined, we generate an exponential random variable, \( \Gamma \), using the parameter corresponding to the state. If the user starts in the ON state, \( \Gamma = \lfloor t_{ON} \rfloor \) frames, while if the user starts in the OFF state, \( \Gamma = \lfloor t_{OFF} \rfloor \) frames. Finally, we alternate the parameter used to generate \( \Gamma \), between \( \lambda \) and \( \mu \), until the sum of the generated \( \Gamma \)'s in frames is at least equal to the voice call duration, \( L \).

B. Performance of MAC Protocols

In this subsection, performance of Distributed MAC I (presented in Subsection III-B) is compared with the performance of Distributed CRN II (proposed in [11]). Due to space limitation only distributed MAC protocols are evaluated here. The performance evaluation of Centralized MAC I can be found in [13].

A primary system with \( C \) equal to 30 PUs is considered. Each protocol is simulated for a hundred thousand frames (i.e., the voice call duration, \( L \), is equal to \( 10^5 \) frames).

We choose the following values for the simulation parameters: \( P_{ON} = 0.4, T_{ON}/T_{OFF} = 20 \) frames /30 frames, and \( m = 1 \) frame. Figure 3 shows the maximum number of SUs which can be admitted to the system by the two protocols under the QoS (in terms of delay and loss) constraints. Our protocol can admit slightly more SUs with guaranteed QoS. Note that the results for the ideal case is obtained with 100 % multiplexing, which corresponds to the case of no delay requirements in transmitting voice packets (i.e., the delay bound \( m \rightarrow \infty \)).

Distributed MAC II is one of the first protocols for successfully guaranteeing QoS for SUs in a distributed CRN.

However, because of the deterministic manner used to organize SUs access to idle slots, Distributed MAC II does not achieve fairness among SUs, specially when the voice call duration, \( L \), is short. Our protocol, on the other hand, grants SUs access to idle timeslots based on their current packet dropping rate and the number of packets queued for transmission, and thus the value of \( L \) is not a factor. Fairness among SUs can be classified into two types, namely short-term fairness and long-term fairness. Short-term fairness indicates the fairness among SUs over short time intervals, while long-term fairness reflects the fairness among SUs over the entire duration of the voice call. In the following, we discuss the short-term and long-term fairness achieved by both distributed MAC protocols.

Short-term fairness: We run the simulation for a duration of \( 10^4 \) frames and calculate the \( P_d \) values of the admitted SUs over disjoint intervals of 500 frames. Figures 4 and 5 show the short-term fairness for the protocols respectively, each for four randomly chosen SUs from the admitted SUs.

It is observed from Figure 4 that the variance of the packet dropping rate among the SUs is small when Distributed MAC I is used. In other words, over each interval (500 time frames), all SUs in the CRN endure similar values of \( P_d \). This reflects good fairness among SUs. However, when Distributed MAC II is used, the variance of the packet dropping rate among the SUs varies drastically, as shown in Figure 5.
indicates that Distributed MAC I provides much better short-term fairness among SUs than Distributed MAC II.

![Figure 5. Short term fairness of Distributed MAC II (T_{ON}/T_{OFF}=2/3, m = 1 frame, L = 10^4 frames, and P_{ON}=0.4).](image)

Long-term fairness: We run the simulation for the entire duration of the voice call and record the value of $P_d$ endured by each user. Figure 6(a) shows the variance of $P_d$ among SUs, when $L$ is equal to $10^5$ frames. It is clear that both MAC protocols achieve good long-term fairness. However, as shown in Figure 6(b), when $L$ is reduced to $10^4$ frames, the improvement in long-term fairness among SUs is obvious when Distributed MAC I is utilized.

![Figure 6. Long term fairness of distributed MAC (T_{ON}/T_{OFF}=20/30, m = 1 frame, and P_{ON}=0.4).](image)

Overall, the newly proposed packet scheduling strategy improves the fairness in channel access using Distributed MAC I, in comparison with the existing Distributed MAC II protocol. Finally, we study the effect of the system parameters, $P_{ON}$, $T_{ON}/T_{OFF}$ ratio, $P_D$, and $m$, on the value of $N$ (the number of SUs admitted with QoS support). It is observed that $N$ is inversely proportional to $P_{ON}$, and directly proportional to $m$ and $P_D$. In addition, as the average time spent by the users in a certain state is reduced (smaller $T_{ON}/T_{OFF}$ ratio), the value of $N$ is increased because users switch faster between the ON state and OFF state, which results in better multiplexing among the voice traffic flows. Detailed analysis of the effect of system parameters can be found in [13].

V. CONCLUSION

This paper investigates link-layer channel resource allocation for voice users in cognitive radio networks, under the assumption of accurate spectrum sensing. Our research objective is to design MAC protocols that maintain fairness among SUs in a CRN, while guaranteeing a certain level of QoS suitable for voice communications. Two MAC protocols for CRNs are proposed. With packet transmissions based on ordering SUs according to their endured packet dropping rates and the numbers of packets queued for transmission, the MAC protocols are to provide a measure of QoS suitable for voice communications and maintain fairness among all SUs in the CRN. Simulation results demonstrate that the proposed Distributed MAC I protocol can admit slightly more SUs and achieve better short-term and long-term fairness among SUs than the existing Distributed MAC II protocol.

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