Voice-Service Capacity Analysis for Cognitive Radio Networks

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***Abstract*—In this paper, quality-of-service (QoS) provisioning for voice service in cognitive radio networks is considered. As voice traffic is sensitive to delay, the presence of primary users and the requirement that secondary users should not interfere with them pose many challenges for QoS support in cognitive radio networks. Two cognitive medium-access control (MAC) schemes are proposed in this paper for secondary voice users to access the available channel. One is the contention-based scheme, and the other is the contention-free scheme. An analytical model is developed to obtain the voice-service capacity (i.e., the maximum number of secondary voice users that can be supported with QoS guarantee) of the two proposed schemes, taking into account the impact of the primary users' activity. Both independent and correlated channel busy/idle state models for primary activity are considered. The analytical model is validated by simulations. The analytical results will be useful to support voice service in cognitive radio networks.**

***Index Terms*—Cognitive radio, medium-access control (MAC), quality of service (QoS), voice-service capacity.**

I. Introduction

COGNITIVE radio, which was the idea first introduced by Mitola [2], [3] and recently promoted by the U.S. Federal Communications Commission [4], provides an effective and efficient solution for the paradox between the shortage of the wireless spectrum resources and the underutilization of the licensed spectrum. An opportunistic (or cognitive) spectrum access approach has been proposed to allow the unlicensed users (which are also called *secondary users*) to exploit the spectrum that is not being used by the licensed users (which are also called *primary users*) [5]. In this manner, a highly economical and efficient usage of the wireless spectrum can be achieved while allowing the primary users to enjoy their licensed spectrum without facing any interference from the secondary users. Because of this property, cognitive radio has recently drawn much attention from researchers. Previous research efforts have focused on addressing the cooperative sensing of the primary users' activity at the physical layer. Recently, researchers have paid more attention to the issues of medium-access control (MAC) and quality-of-service (QoS) provisioning for the secondary users.

The goal of this work is to support the quality of voice service for secondary users at the MAC layer. As voice traffic is sensitive to delay, the presence of primary users and the requirement of secondary users not interfering with them pose many challenges for QoS provisioning in cognitive radio networks. First, an efficient and low-complexity cognitive MAC scheme is required for the secondary users to share the available spectrum not being utilized by the primary users. Second, to guarantee the QoS of voice service for secondary users, it is critical to obtain the voice-service capacity (i.e., the maximum number of secondary voice users that can be supported), taking the impact of the primary users' activity into consideration.

In this paper, we propose two cognitive medium-access schemes for the secondary voice users to access the wireless spectrum. We also develop an analytical model to obtain the voice service capacity for the two cognitive medium-access schemes. The analytical results are validated by simulations. The performance evaluation reveals how the activity of the primary users and the different cognitive medium-access schemes affect the cognitive voice-service capacity. The analytical model can be used for the radio resource management and call-admission control in cognitive radio networks.

The rest of this paper is organized as follows: In Section II, the related work is reviewed. The system model is described in Section III, and two cognitive medium-access schemes are presented in Section IV. Section V is devoted to the analytical model for cognitive voice service. Section VI presents the numerical results, followed by the concluding remarks in Section VII.

II. Related Work

Medium access in cognitive radio networks has recently been well investigated. In [6], an optimal channel-access strategy was obtained, based on a partially observable Markov decision process. In [7] and [8], the channel-sensing order problem was investigated for the single-user and two-user cases, respectively. In [9], a hardware-constrained (i.e., secondary users have limited spectrum sensing and transmission capability) multichannel cognitive MAC scheme was proposed to conduct efficient spectrum sensing and spectrum access decision in cognitive radio ad hoc networks. In [10], a time-division-multiple-access (TDMA)/frequency-division-multiple-access (FDMA) based Global System for Mobile communications (GSM)

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cellular network with or without the use of frequency hopping was considered to be the primary system, and a MAC scheme for opportunistic spectrum sharing was proposed for secondary users to utilize the resource which is unused by primary users in the GSM cellular system. In [11], a carrier-sense-multiple-access/collision-avoidance (CSMA/CA) based MAC protocol was proposed for cognitive wireless ad hoc networks, exploiting the statistics of spectrum usage for decision making on channel access. Before each transmission, a negotiation on transmission parameters between a sender and a receiver is performed through the control channel. In [12], a time-slotted single-channel system model was considered. Heuristic control policies were proposed with the objective to perform optimal sensing and to maximize the expected reward of secondary users with energy constraint. In [13] and [14], the optimal tradeoff in the sensing time setting was investigated in single-channel and multichannel cases, respectively.

In the literature, some research works have been done for the QoS provisioning in cognitive radio networks. In [15], an opportunistic scheduling policy was proposed to maximize the throughput utility, which reflects the QoS of the secondary users subject to maximum collision constraints with the primary users. In [16], a MAC framework based on game-theoretic dynamic spectrum allocation was proposed to achieve high spectrum utilization, collision-free channel access, and QoS support. In [17], a cross-layer-based opportunistic multichannel MAC protocol, which integrates the spectrum-sensing policy at the physical layer with packet scheduling at the MAC layer, was proposed for cognitive radio networks. The QoS performances in terms of throughput and delay were also analyzed. In [18], a cognitive MAC protocol was proposed, in which the secondary users are divided into several nonoverlapping groups. Each group uses a bonding/aggregation technique to guarantee the QoS requirements of the secondary users. In [19], a MAC protocol consisting of an admission-control module, rate-control module, and resource-allocation module was proposed for point-to-multipoint cognitive radio networks to guarantee the QoS requirements of different classes of admitted connections. In [20], a new concept of the delay bandwidth product was introduced for the variable bandwidth channels, and the secondary users select the optimal channels to achieve the highest throughput and to guarantee their QoS requirements. In [21], a distributed multichannel power-allocation problem was studied for cognitive radio networks. The problem was formulated as a noncooperative game in which each user aims at achieving its QoS using the least power consumption. Note that none of the aforementioned works focused on the voice-service support for secondary users, and the voice-capacity analysis has never been considered.

III. System Model

Cognitive communication technology has been studied for different wireless networks with large coverage [22] and small coverage [23]. In this paper, we consider a wireless network with small coverage, where the activity of all primary users can be sensed by each secondary user. The network has a single wireless channel, which is shared by all the primary and secondary users. TDMA and time-division multiplexing (TDM) are used in the primary network, in which the channel time is partitioned into slots, and each primary user is either active or idle in each time slot. For a secondary user, before sensing and accessing the channel, it first synchronizes with the slot structure of the primary users. In the secondary network, the multiplexing method is TDM, and the multiple-access method will be discussed in the next section. Generally, at the beginning of each time slot, the secondary users sense the activity of the primary users. If the channel is sensed idle, the secondary users can exploit the availability of the channel. A similar system model has been adopted in [6], [12], [24], and [25].

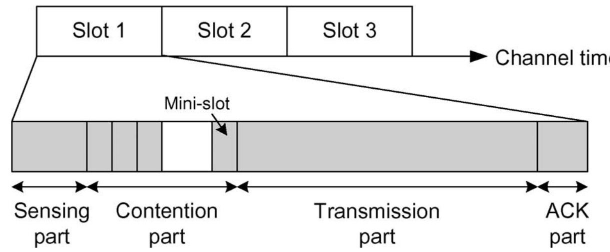

Fig. 1. Time slot structure.

Constant rate voice traffic is supported for the secondary users. Each secondary user, if active, has one voice session. As voice traffic is sensitive to delay, a voice packet with a large delay will be considered useless. Therefore, if a voice packet cannot successfully be delivered during a delay bound after its generation, this voice packet will be dropped by the voice sender. To maintain satisfactory voice quality, the voice-packet-dropping probability should not be higher than threshold P_l. Usually, the threshold is set to be 1%.

IV. Proposed Cognitive Medium-Access Schemes

In cognitive radio networks, a cognitive MAC scheme is required for the secondary users to efficiently share the available wireless channel when the primary users are not active. A cognitive MAC scheme has two basic functions. The first function is to ensure that the secondary users will not interfere with the primary users. The second function is to achieve low complexity, high efficiency, and fair medium access among the secondary users. Different cognitive medium access schemes will achieve different performances (e.g., in terms of resource utilization and efficiency), leading to different system capacities for secondary users. In this section, two cognitive MAC schemes are proposed. One is the contention-based scheme, and the other is the contention-free scheme.

A. *Contention-Based Medium Access*

The time structure is shown in Fig. 1. The channel time is partitioned into time slots, and a time slot is further divided into four parts. The first part, i.e., the sensing part, is used for all the secondary users to sense the activity of the primary users. If the channel is sensed busy, no secondary user should contend for that slot. The second part, i.e., the contention part, consists of a number of minislots. All the secondary users have the same contention window size. If the channel is sensed idle in the sensing part, a secondary user randomly chooses a backoff timer from the contention window and continues to sense the channel in the contention part. If the channel is sensed to be continuously idle for a duration of the backoff timer (in units of minislots) in the contention part, the secondary user will terminate the contention part and transmit its packet in the transmission part (the third part of the time slot); otherwise, it will quit the contention for the current slot. Therefore, for each contention, the secondary user with the smallest backoff timer will win and transmit its packet in the transmission part of the slot. Note that it is possible that more than one secondary user choose the same smallest backoff timer, resulting in a collision. To determine whether a packet has successfully been transmitted, the receiver sends an acknowledgment (ACK) at the ACK part (the last part) of each slot to the sender upon a successful packet reception.

B. Contention-Free Medium Access

Similar to the contention-based medium access, a time slot is also divided into four parts in the contention-free medium access. The difference is that, in the second part (which is also called the contention part for simplicity of presentation), the secondary users do not follow the backoff mechanism. Instead, each minislot in the contention part is assigned to a secondary user in a deterministic way. The minislot assignment procedure is to be discussed later. A secondary user (e.g., user A) with minislot index i first senses the channel from minislot 1 to minislot $i-1$ in the contention part. If the channel remains idle (i.e., no secondary user with minislot index smaller than i has packet to transmit), then user A can start transmission from minislot i of the contention part until the end of the transmission part. If the channel becomes busy from any minislot prior to minislot i, which indicates that another secondary user with a smaller minislot index has already started its transmission, user A should not transmit in the current slot. Since a secondary user is assigned a unique minislot, a contention-free medium access can be achieved. Note that the chance that a user transmits in a slot largely depends on its minislot index. The smaller the index, the larger the chance. To maintain fair medium access among all the secondary users, the minislot assignment will be rotated among all the secondary users after each slot. For example, the user assigned the first minislot in the current slot will be assigned the last minislot in the next slot, the user assigned the second minislot in the current slot will be assigned the first minislot in the next slot, and so on.

If the number of secondary users in the cognitive radio network is fixed, the minislot-assignment procedure can be done once at the initialization of the network. If the secondary users dynamically join or leave the cognitive radio network, a minislot-assignment procedure needs to be performed upon every user arrival or departure event. Any secondary user can be designated to perform the minislot-assignment procedure. We call the secondary user who performs the minislot assignment the *minislot assigner (MSA)*. When a new secondary user wants to join the cognitive radio network, it first broadcasts a JOIN message. Upon receiving the JOIN message, the MSA sends an ACK message, which includes the assigned minislot index, to the new user. Similarly, when a user leaves the cognitive radio network, it broadcasts a LEAVE message, and the MSA replies with an ACK message. The MSA then reassigns the minislots to all the remaining secondary users and broadcasts the new assignment result to them. When the MSA leaves the cognitive network, it designates another existing user to perform the minislot assignment before leaving and includes the new MSA ID and information of other existing users in the LEAVE message. Upon receiving the LEAVE message from the current MSA, the new MSA replies with an ACK and then reassigns the minislots and broadcasts the new result to all the existing secondary users. The JOIN/LEAVE/ACK messages are given high priority to be sent, compared with voice packets. To achieve this, the first minislot (which we call minislot 0) in the contention part of a slot is dedicated to the users with JOIN/LEAVE/ACK messages. Therefore, a user with a JOIN/LEAVE/ACK message can transmit, starting from the first minislot of the contention part, whereas other users having voice packets to send have to monitor the channel from the first minislot of the contention part and hence sense a busy channel and defer. As JOIN/LEAVE/ACK messages are infrequently sent, collisions caused by two or more simultaneous JOIN/LEAVE/ACK transmissions in one slot are negligible.

The robustness of the contention-free scheme is achieved by the ACK message. To further enhance the robustness, the following two actions can be taken. First, after a prespecified period of time, the MSA announces the latest status of the minislot-assignment rotation. Second, in the case where a secondary user detects consecutive collisions (which is very likely because two or more users, by some mistake, use the same minislot), a control message COLLISION-REPORT will be sent to the MSA, and the MSA will reassign the minislots and perform an announcement to all the users. The message COLLISION-REPORT is a high-priority packet and, thus, is also assigned minislot 0.

Regarding the overhead of the proposed contention-free scheme, the control message will be sent when the following conditions occur: 1) a secondary user joins or leaves the network; 2) the MSA announces the latest status of the minislot-assignment rotation after a prespecified period of time; and 3) collisions are detected. It can be seen that these events do not frequently happen. Therefore, the overhead is small.

As mentioned in Section III, we consider a wireless network with small coverage, where the activity of all primary users can be sensed by each secondary user. For a network with small area, the number of users is expected to be small; thus, scalability will not be a considerable concern for the proposed MAC schemes.

V. Analytical Model

To guarantee the QoS of voice traffic, it is critical to have appropriate call-admission control mechanism. Call-admission control is responsible for admitting or rejecting a new voice call based on the available resources to ensure that the QoS requirements (e.g., delay and packet loss rate) of all the admitted voice calls are satisfied. Therefore, it is essential to obtain the system capacity. In cognitive radio networks, the system capacity for the secondary users depends on the number

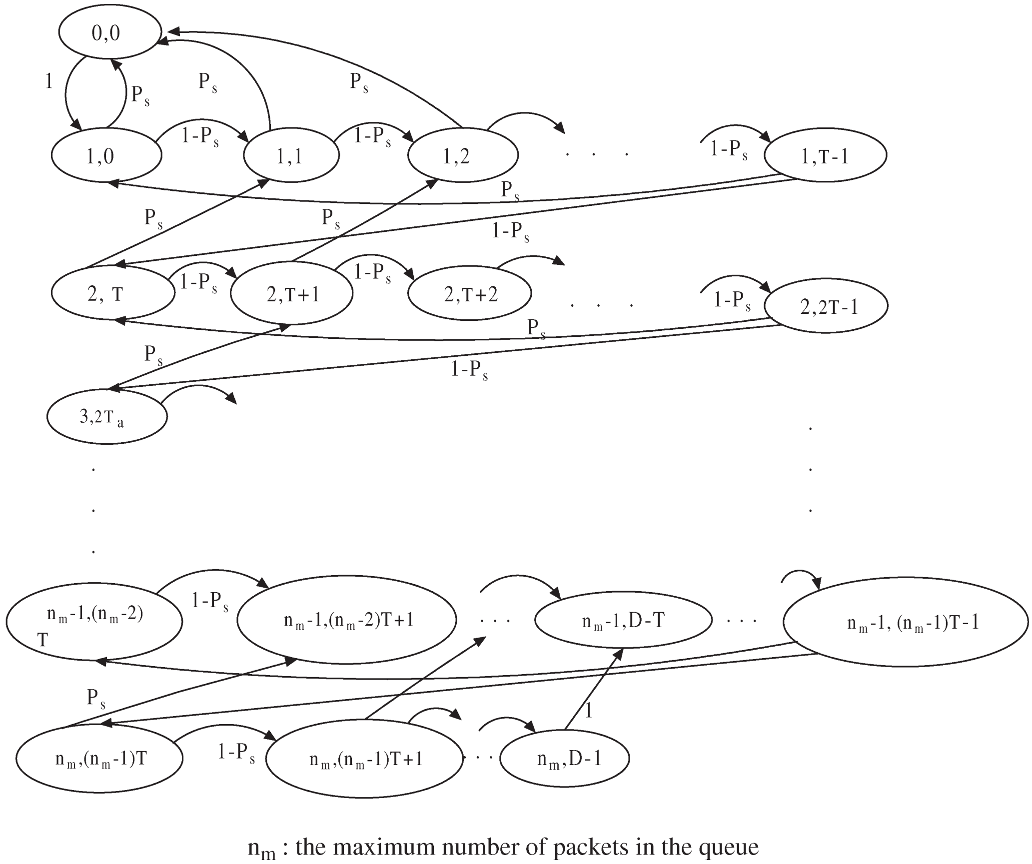

Fig. 2. State transition diagram for the independent channel-state model.

of primary users and their activities and also depends on the cognitive medium-access scheme. In this section, we present an analytical model to derive the voice-service capacity for the two proposed cognitive medium-access schemes. For the primary-user activity (busy or idle), we consider two different models, i.e., an *independent channel state model*, in which the channel busy/idle state (due to the primary user activity) is independent from slot to slot, and a *correlated channel state model*, in which the channel busy/idle state is correlated over neighboring slots. Without loss of generality, we assume that, if a user successfully transmits in a time slot, the number of transmitted packets is one. Denote the voice-packet interarrival time and the voice-packet delay bound as T and D (both in the unit of time slots), respectively.

A. Capacity Analysis of the Independent Channel-State Model

In the independent channel-state model, at each time slot, a primary user is idle with a probability θ. Let N_p denote the total number of primary users. Therefore, the probability that the channel is available for the secondary users to access is θ^{N_p}.

We arbitrarily choose a secondary user as the tagged user. At each time slot, we sample the state (n, t), where n is the number of voice packets in the queue of the tagged user, and t is the queueing delay (in the unit of time slots) experienced by the voice packet at the head of the queue of the tagged user. The initial state of (n, t) is (0, 0), indicating that there is no voice packet at the tagged user. As constant-rate voice traffic is considered, after no more than T time slots, a voice packet will arrive at the tagged user. Therefore, the state of (n, t) will move to (1, 0) with probability 1. Since then, after each time slot, the state of (n, t) will evolve, moving to another state. The state-transition process of (n, t) is modeled by a discrete-time Markov chain, as shown in Fig. 2. In Fig. 2, n_m indicates the maximum number of packets in the queue of the tagged user, which is given by $n_m = \lfloor (D-1)/T \rfloor + 1$, where $\lfloor \cdot \rfloor$ is the floor function. To describe this Markov chain, we use state (n, t) for $n > 1$ as an example. Note that when $n > 0$, we will have $n = \lfloor t/T \rfloor + 1$.

1) When $t < nT - 1$, the next state is $(n, t+1)$ if the tagged user cannot successfully transmit a voice packet in the current slot and $(n-1, t-T+1)$ if the tagged user successfully transmits a voice packet in the current slot. Note that since voice traffic has a constant rate, the voice-packet interarrival time T is a fixed value. For any two consecutive packets in the queue, the difference in their queuing delays (i.e., the waiting time in the queue, not including the time in packet transmission) is equal to T.
2) When $t = nT - 1$, the next state is $(n+1, t+1)$ if the tagged user cannot successfully transmit a voice packet in the current slot or $(n, t-T+1)$ if the tagged user successfully transmits a voice packet in the current slot. This is because when the delay of the packet at the head of queue is equal to $nT - 1$, a new voice packet will arrive in the next time slot.

Let P_s denote the probability that the tagged secondary user successfully transmits a voice packet in a randomly chosen time slot, given that the tagged user has packets to send. Let $P(n_1, t_1|n_0, t_0)$ denote the transition probability from state (n_0, t_0) to state (n_1, t_1) in the Markov chain, which is given by

$$\begin{cases} P(1,0|0,0)=1 & \\ P(1,t+1|1,t) = 1-P_s, & t<T-1 \\ P(0,0|1,t)=P_s, & t<T-1 \\ P(2,t+1|1,t)=1-P_s, & t=T-1 \\ P(1,0|1,t)=P_s, & t=T-1 \\ P(n,t+1|n,t)=1-P_s, & t<nT-1,\ n\geq 2 \\ P(n-1,t+1-T|n,t)=P_s, & t<nT-1,\ n\geq 2 \\ P(n+1,t+1|n,t)=1-P_s, & t=nT-1,\ n\geq 2 \\ P(n,t+1-T|n,t)=P_s, & t=nT-1,\ n\geq 2 \\ P(n_m-1,D-T|n_m,D-1)=1, & \text{if } D<n_mT-1 \\ P(n_m,D-T|n_m,D-1)=1, & \text{if } D=n_mT-1. \end{cases} \tag{1}$$

Given all the one-step transition probabilities of the Markov chain listed in (1), the steady-state probability vector of the Markov chain can be obtained. Let $\pi(n,t)$ denote the steady-state probability of state (n,t). Recall that $n = \lfloor t/T \rfloor + 1$ when $n > 0$.

Next, we derive P_s for the two proposed cognitive medium access schemes. The voice-packet-arrival rate of each secondary user is $1/T$ packet/slot. The average voice packet service rate of each secondary user is denoted as μ packet/slot. Therefore, the queue utilization of the secondary user is given by $\rho = 1/(T\cdot\mu)$.

For contention-based medium access, P_s is given by

$$P_s = \theta^{N_p}\cdot \sum_{i=0}^{N_s-1}\binom{N_s-1}{i}\cdot\rho^i\cdot(1-\rho)^{N_s-1-i} \cdot\left(\sum_{j=1}^{CW}\frac{1}{CW}\cdot\left(\frac{CW-j}{CW}\right)^i\right) \tag{2}$$

where θ^{N_p} is the probability that the channel is available to the secondary users, N_s is the total number of secondary users, and CW is the contention window size of the secondary users. The term in the second summation indicates the probability that the tagged user chooses a backoff timer value j, and other active secondary users[1] choose backoff timer values larger than j.

For the contention-free medium access, P_s is given by

$$P_s = \theta^{N_p}\cdot\sum_{i=1}^{N_s}\frac{1}{N_s}\cdot(1-\rho)^{i-1} \tag{3}$$

where the term in the summation indicates the probability that, for a randomly chosen slot, the tagged user has minislot index i, and all other secondary users with minislot index smaller than i have no packet to transmit.

To determine ρ, we need to characterize the average service time of a voice packet, i.e., $1/\mu$. Given that the voice packet at the head of the queue has already waited t slots, the average

[1]Active secondary users refer to the secondary users whose queues are not empty.

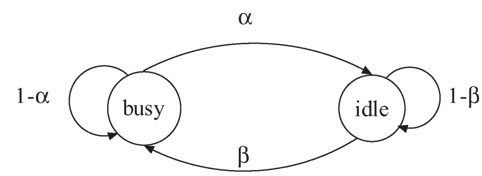

Fig. 3. Two-state Markov model of the channel busy/idle states.

time (in units of slots) required to serve this packet, i.e., the time to let the packet leave the queue (due to either successful transmission or packet dropping), is given by

$$T_s(t) = \left(\sum_{k=1}^{D-t-1}(1-P_s)^{k-1}\cdot P_s\cdot k\right) + (1-P_s)^{D-t-1}\cdot(D-t). \tag{4}$$

The average service time of a voice packet is derived as

$$\frac{1}{\mu} = \frac{\sum_{n>0}\pi(n,t)\cdot T_s(t)}{\sum_{n>0}\pi(n,t)}. \tag{5}$$

Given the equations in (1)–(5), P_s can be solved for the two proposed cognitive medium-access schemes, and the steady-state probability vector for the Markov chain can further be obtained, using a numerical method similar to that used in [26]–[28].

Let P_{drop} denote the voice-packet-dropping probability. P_{drop} can be expressed as

$$P_{\text{drop}} = \frac{\sum_{n>0}\pi(n,t)\cdot(1-P_s)^{D-t}}{\sum_{n>0}\pi(n,t)}. \tag{6}$$

Note that P_{drop} is a function of N_p, N_s, and θ. To guarantee the QoS of voice traffic, P_{drop} should not exceed the voice-packet-dropping rate bound P_l. Therefore, the capacity for voice service of secondary users is the maximum integer N_s (which is denoted by N_s^*) satisfying $P_{\text{drop}} \leq P_l$.

B. Capacity Analysis of the Correlated Channel-State Model

The preceding analysis is based on the assumption that the channel busy/idle state due to primary user activity is independent from slot to slot. However, in some scenarios (e.g., when the primary users' traffic is bursty), the channel busy/idle states over neighboring slots are correlated rather than independent. The two-state Markov model, as shown in Fig. 3, can be used to model the correlation of channel states [15], [17], where α and β are the probabilities that the channel state transits from busy (in the current slot) to idle (in the next slot) and from idle to busy, respectively. The larger the values of $|\alpha - 0.5|$ and $|\beta - 0.5|$, the more correlated the channel states.

To take the correlated channel into account, we extend the two-dimensional (2-D) state (n,t) (as defined in Section V-A) to the three-dimensional (3-D) state (n,t,s), where s represents the channel state ($s=0$ for the idle state and $s=1$ for the busy state). Another difference from the 2-D model is that t has a different meaning for the cases of $n=0$ and $n>0$. When $n>0$, t is the queueing delay experienced by the voice packet that is at the head of the queue of the tagged user (i.e., the same as defined for the 2-D model). When $n=0$, t is the time required (from the current slot) until the next voice packet arrives. For

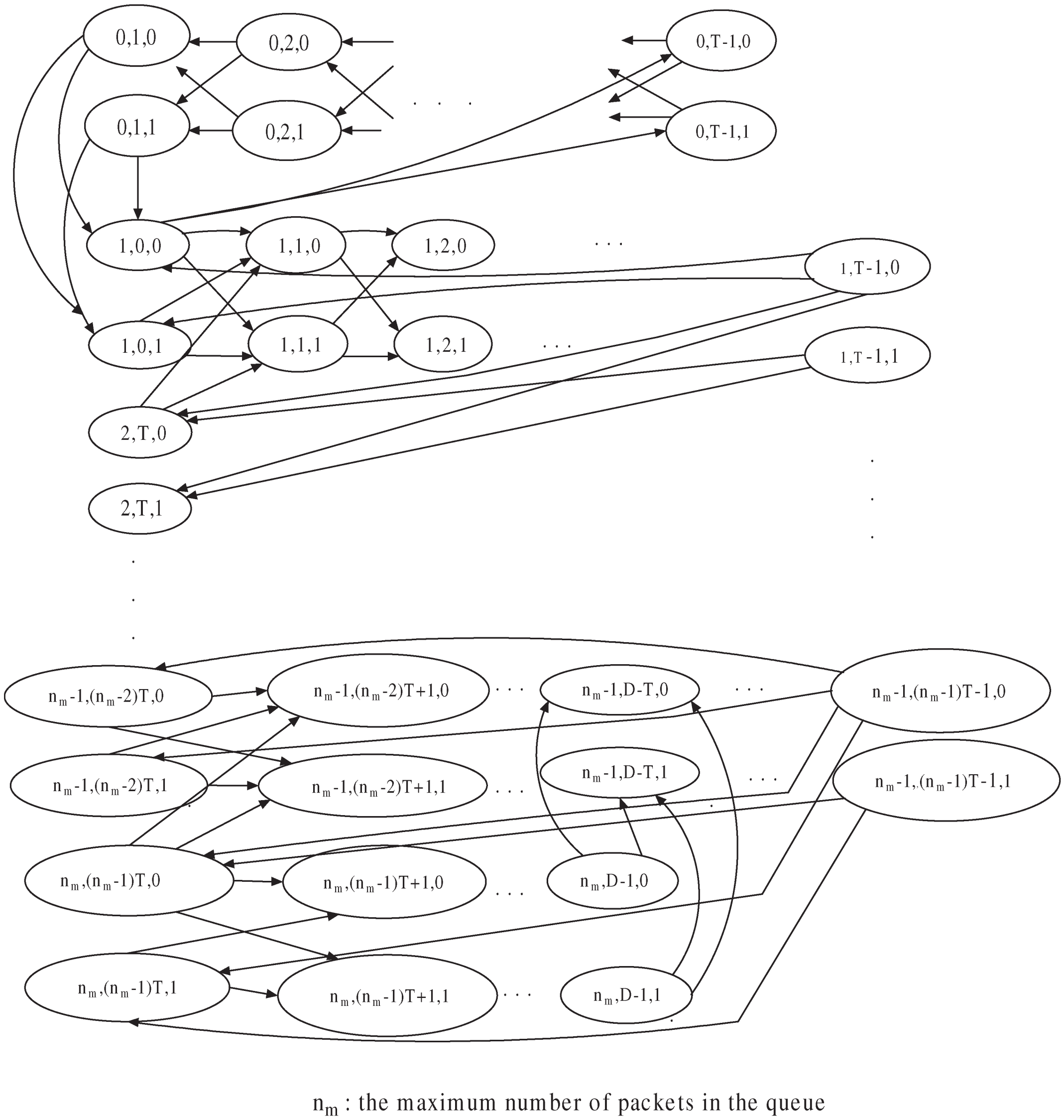

Fig. 4. State-transition diagram for the correlated channel.

example, the state $(0, t, 0)$ means that there is no voice packet at the tagged user, the user needs to wait t slots for the next voice packet arrival, and the current channel state is idle. The state $(2, t, 1)$ means that there are two packets at the tagged user, the first arrived packet has waited t slots in the queue, and the current channel state is busy. The state-transition process is modeled by a Markov chain shown in Fig. 4. State (n, t, s) can transit to a new state as given here.

1) When $n = 0$ and $T - 1 \geq t > 1$, no new voice packet will arrive in the next slot. Therefore, state $(0, t, s)$ will move to $(0, t - 1, 0)$ if the channel is idle in the next slot and to $(0, t - 1, 1)$ if the channel is busy in the next slot.
2) When $n = 0$ and $t = 1$, a voice packet will arrive in the next slot. Therefore, state $(0, 1, s)$ will move to state (1, 0, 0) if the channel is idle in the next slot and to (1, 0, 1) if the channel is busy in the next slot.
3) When $n = 1$, and $t < T - 1$, two cases are considered.
   a) For state $(1, t, 0)$ (i.e., the channel is idle in the current slot), the next state is $(0, T - 1 - t, s')$ if the tagged user successfully transmits a voice packet in the current slot or $(1, t + 1, s')$ if the tagged user cannot successfully transmit a voice packet in the current slot. If the channel is still idle in the next slot, $s' = 0$; otherwise, $s' = 1$. Note that we consider voice traffic with constant rate, and the voice-packet interarrival time is T. As a result, when the current packet has already waited t slots in the queue before being transmitted in the current slot, the next packet will arrive after $T - t - 1$ slots.
   b) For state $(1, t, 1)$ (i.e., the channel is busy in the current slot), the next state is $(1, t + 1, 0)$ if the channel becomes idle in the next slot or $(1, t + 1, 1)$ if the channel is still busy in the next slot.
4) When $n > 1$ and $t < nT - 1$, two cases are considered.
   a) For state $(n, t, 0)$, the next state is $(n - 1, t - T + 1, s')$ if the tagged user successfully transmits a voice packet in the current slot or $(n, t + 1, s')$ if the tagged user does not successfully transmit a voice packet in the current slot. The value of s' depends on the channel state in the next slot.

   b) For state $(n, t, 1)$, the next state is $(n, t+1, 0)$ if the channel is idle in the next slot or $(n, t+1, 1)$ if the channel is busy in the next slot.

5) When $n \geq 1$ and $t = nT - 1$, a new packet will arrive in the next slot.
   a) For state $(n, t, 0)$, the next state is $(n, t-T+1, s')$ if the tagged user successfully transmits a voice packet in the current slot or $(n+1, t+1, s')$ if the tagged user does not successfully transmit a voice packet in the current slot.
   b) For state $(n, t, 1)$, the next state is $(n+1, t+1, 0)$ if the channel becomes idle in the next slot or $(n+1, t+1, 1)$ if the channel is still busy in the next slot.

6) When $t = D - 1$, we have $n = n_m = \lfloor (D-1)/T \rfloor + 1$, which means that the packet at the head of the queue will violate the delay bound in the next slot. Therefore, if the packet can successfully be transmitted in the current slot, it will leave the queue (the packet will be dropped if transmission is not successful).
   a) If $D < nT - 1$, state $(n, D-1, s)$ will move to state $(n-1, D-T, 0)$ if the channel is idle in the next slot or to state $(n-1, D-T, 1)$ if the channel is busy in the next slot.
   b) If $D = nT - 1$, a new packet will arrive in the next slot. Therefore, state $(n, D-1, s)$ will move to state $(n, D-T, 0)$ if the channel is idle in the next slot or to state $(n, D-T, 1)$ if the channel is busy in the next slot.

Let P'_s denote the probability that the tagged user successfully transmits a voice packet in a randomly chosen time slot, given that the channel in that slot is idle and the tagged user has packets to transmit. The one-step transition probabilities of the Markov chain shown in Fig. 4 are listed in (7), shown at the bottom of the page.

For the contention-based medium access, P'_s is given by

$$P'_s = \sum_{i=0}^{N_s-1} \binom{N_s-1}{i} \cdot \rho^i \cdot (1-\rho)^{N_s-1-i} \cdot \left(\sum_{j=1}^{CW} \frac{1}{CW} \cdot \left(\frac{CW-j}{CW} \right)^i \right). \quad (8)$$

For the contention-free medium access, P_s is given by

$$P'_s = \sum_{i=1}^{N_s} \frac{1}{N_s} \cdot (1-\rho)^{i-1}. \quad (9)$$

Note that the expressions of P'_s are similar to those of P_s, as in (2) and (3). The difference is that P'_s is the conditional probability, given that the current channel is idle. Therefore, we do not have the term θ^{N_p}.

The derivation of the average service time of a voice packet $1/\mu$ is more complex for the 3-D model than that for the 2-D model. We denote any state after a successful transmission or packet dropping as state S. For a state $(n, t, 0)$ ($n > 0$ and $t < D - 1$), in the next slot, it will move to state S with probability P'_s, to state $(n', t+1, 0)$ with probability $(1 - P'_s)(1 - \beta)$, and to state $(n', t+1, 1)$ with probability $(1 - P'_s)\beta$. The value of n' can be the same as or different from n, depending on the value of t. For a state $(n, t, 1)$ ($n > 0$ and $t < D - 1$), in the next slot, it will move to state $(n', t+1, 0)$ with probability α and to state $(n', t+1, 1)$ with probability $1 - \alpha$, as shown in Fig. 5. For state $(n, D-1, s)$, the packet will be either successfully transmitted or dropped; hence, the next state will be S with probability 1. For the transition shown in Fig. 5, we use branch transmittance, which is denoted by $g_{a,b}$, to associate with the state transition from a to b. The branch transmittance is defined as $g_{a,b} = p_{a,b} z^{t_{a,b}}$, where $p_{a,b}$ is the

$$\begin{cases}
P(0,t-1,0|0,t,0) = 1-\beta; \quad P(0,t-1,1|0,t,0) = \beta, & 1 < t \leq T-1 \\
P(0,t-1,0|0,t,1) = \alpha; \quad P(0,t-1,1|0,t,1) = 1-\alpha, & 1 < t \leq T-1 \\
P(1,0,0|0,t,0) = 1-\beta; \quad P(1,0,1|0,t,0) = \beta, & t = 1 \\
P(1,0,0|0,t,1) = \alpha; \quad P(1,0,1|0,t,1) = 1-\alpha, & t = 1 \\
P(0,T-1-t,0|1,t,0) = P'_s(1-\beta); \quad P(0,T-1-t,1|1,t,0) = P'_s\beta, & t < T-1 \\
P(1,t+1,0|1,t,0) = (1-P'_s)(1-\beta); \quad P(1,t+1,1|1,t,0) = (1-P'_s)\beta, & t < T-1 \\
P(1,t+1,0|1,t,1) = \alpha; \quad P(1,t+1,1|1,t,1) = 1-\alpha, & t < T-1 \\
P(n-1,t-T+1,0|n,t,0) = P'_s(1-\beta); \quad P(n-1,t-T+1,1|n,t,0) = P'_s\beta, & t < nT-1,\ n > 1 \\
P(n,t+1,0|n,t,0) = (1-P'_s)(1-\beta); \quad P(n,t+1,1|n,t,0) = (1-P'_s)\beta, & t < nT-1,\ n > 1 \\
P(n,t+1,0|n,t,1) = \alpha; \quad P(n,t+1,1|n,t,1) = 1-\alpha, & t < nT-1,\ n > 1 \\
P(n,t-T+1,0|n,t,0) = P'_s(1-\beta); \quad P(n,t-T+1,1|n,t,0) = P'_s\beta, & t = nT-1,\ n \geq 1 \\
P(n+1,t+1,0|n,t,0) = (1-P'_s)(1-\beta); \quad P(n+1,t+1,1|n,t,0) = (1-P'_s)\beta, & t = nT-1,\ n \geq 1 \\
P(n+1,t+1,0|n,t,1) = \alpha; \quad P(n+1,t+1,1|n,t,1) = 1-\alpha, & t = nT-1,\ n \geq 1 \\
P(n-1,D-T,0|n,D-1,0) = (1-\beta); \quad P(n-1,D-T,1|n,D-1,0) = \beta, & \text{if } D < nT-1 \\
P(n-1,D-T,0|n,D-1,1) = \alpha; \quad P(n-1,D-T,1|n,D-1,1) = 1-\alpha, & \text{if } D < nT-1 \\
P(n,D-T,0|n,D-1,0) = (1-\beta); \quad P(n,D-T,1|n,D-1,0) = \beta, & \text{if } D = nT-1 \\
P(n,D-T,0|n,D-1,1) = \alpha; \quad P(n,D-T,1|n,D-1,1) = 1-\alpha, & \text{if } D = nT-1
\end{cases} \quad (7)$$

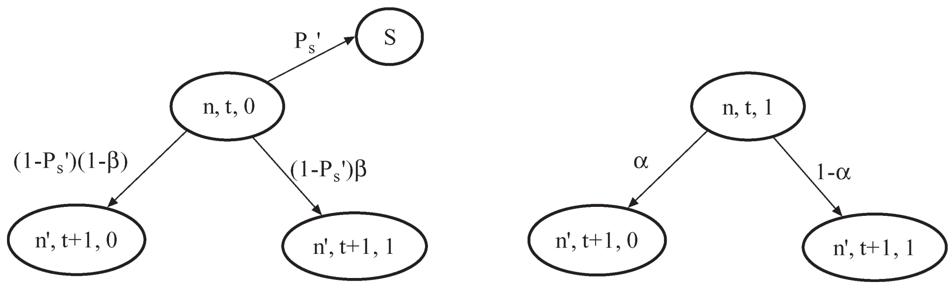

Fig. 5. State transition from states $(n, t, 0)$ and $(n, t, 1)$, with $n > 0$, and $t < D - 1$.

transition probability from state a to state b, $t_{a,b}$ is the time duration of the transition (i.e., one time slot in our model), and z is a dummy variable. Denote the probability-generating function of the time from state (n, t, s) to state S as $G_{n,t,s}(z)$. We have

$$\begin{aligned} G_{n,t,0}(z) &= g_{(n,t,0),S} + g_{(n,t,0),(n',t+1,0)} \cdot G_{n',t+1,0}(z) \\ &\quad + g_{(n,t,0),(n',t+1,1)} \cdot G_{n',t+1,1}(z) \\ &= P'_s z + (1 - P'_s)(1 - \beta) z \cdot G_{n',t+1,0}(z) \\ &\quad + (1 - P'_s)\beta z \cdot G_{n',t+1,1}(z) \\ G_{n,t,1}(z) &= g_{(n,t,1),(n',t+1,0)} \cdot G_{n',t+1,0}(z) \\ &\quad + g_{(n,t,1),(n',t+1,1)} \cdot G_{n',t+1,1}(z) \\ &= \alpha z \cdot G_{n',t+1,0}(z) + (1-\alpha) z \cdot G_{n',t+1,1}(z). \end{aligned} \tag{10}$$

It is straightforward to see that $G_{n,D-1,0}(z) = G_{n,D-1,1}(z) = z$. With these two values, we can calculate $G_{n,D-2,0}(z)$ and $G_{n,D-2,1}(z)$ based on (10). Similarly, with the values of $G_{n,D-2,0}(z)$ and $G_{n,D-2,1}(z)$, we can calculate $G_{n,D-3,0}(z)$ and $G_{n,D-3,1}(z)$. Eventually, we can obtain the value of $G_{n,t,s}(z)$ for any state (n, t, s) $(n > 0)$. Therefore, the expectation of the service time at state (n, t, s) can be obtained by $E_{n,t,s} = (d/dz) G_{n,t,s}(z)|z = 1$. The average service time in the system is then obtained from

$$\frac{1}{\mu} = \frac{\sum_{n>0} \pi(n, t, S) \cdot E_{n,t,S}}{\sum_{n>0} \pi(n, t, S)} \tag{11}$$

where $\pi(n, t, s)$ is the steady-state probability of state (n, t, s).

According to (7)–(9) and (11), P'_s can be solved based on which the steady-state probability vector for the 3-D Markov chain can further be obtained.

Next, we derive the voice packet-dropping probability P_{drop}. Let F denote the state after a packet dropping. For any state (n, t, s) $(n > 0)$, let $P^F_{n,t,s}$ denote the probability that, after one or multiple transitions, the state will evolve to state F. The derivation of $P^F_{n,t,s}$ is similar to that of $G_{n,t,s}(z)$. According to Fig. 5, we have

$$\begin{aligned} P^F_{n,t,0} &= (1 - P'_s)(1 - \beta) \cdot P^F_{n',t+1,0} + (1 - P'_s)\beta \cdot P^F_{n',t+1,1} \\ P^F_{n,t,1} &= \alpha \cdot P^F_{n',t+1,0} + (1 - \alpha) \cdot P^F_{n',t+1,1}. \end{aligned} \tag{12}$$

For states $(n, D - 1, 0)$ and $(n, D - 1, 1)$, the values of $P^F_{n,D-1,0}$ and $P^F_{n,D-1,1}$ can directly be obtained as follows:

$$\begin{aligned} P^F_{n,D-1,0} &= 1 - P'_s \\ P^F_{n,D-1,1} &= 1. \end{aligned} \tag{13}$$

Similarly, based on the values of $P^F_{n,D-1,0}$ and $P^F_{n,D-1,1}$, we can obtain $P^F_{n,D-2,0}$, $P^F_{n,D-2,1}$, and so on. After obtaining the value of $P^F_{n,t,s}$ for any state (n, t, s) $(n > 0)$, we have

$$P_{\text{drop}} = \frac{\sum_{n>0} \pi(n, t, S) \cdot P^F_{n,t,S}}{\sum_{n>0} \pi(n, t, S)}. \tag{14}$$

Then, the capacity for the voice secondary users N^*_s can be obtained by finding the maximum integer value of N_s that satisfies $P_{\text{drop}} \leq P_l$.

C. Nonperfect Channel Cases

Note that in the preceding analysis, an ideal channel (with no transmission error) is considered. However, the proposed analytical model can straightforwardly be extended to the nonperfect channel cases. For nonperfect channels, transmission errors may exist, due to fading and path loss. Accordingly, the state transition probabilities of the models shown in Figs. 2 and 4 will be modified by taking the packet transmission error into consideration. Specifically, in (2), (3), (8), and (9), P_s and P'_s will be adjusted by a factor $(1 - \eta)$, in which η is the probability that a packet transmission is not successful due to fading or path loss.

D. Optimal Contention Window Size

In our proposed scheme, the length of the time slot is fixed. For the proposed contention-based medium access, in each time slot, after the backoff period, either one secondary user successfully transmits a voice packet (i.e., with no collision) or a collision occurs. Therefore, if there is no collision in a time slot, the collision avoidance mechanism only affects the backoff period but does not change the fact that a packet is successfully transmitted from a secondary user in the time slot. In this case, the collision-avoidance mechanism does not affect the packet dropping at the secondary user due to the delay bound requirement. On the other hand, if collision occurs in a time slot, no packet is successfully sent in that time slot. Consequently, the queues at the secondary users are built up, giving rise to the packet dropping due to delay bound violation. Therefore, it can be concluded that the voice service capacity is degraded by collisions among secondary users but not contention avoidance.

In fact, in a cognitive radio network, other services (such as data) also exist. However, voice service should have higher priority than data service. Then, it can be seen that the collision avoidance affects the channel time which can be used by data service and, hence, affects the capacity of users with data service. This motivates us to obtain the optimal contention window size CW for voice users such that the most efficient utilization of the channel by the voice users can be achieved.

Generally, if a small CW is adopted for secondary voice users, collision is very likely to occur, leading to a poor performance of the proposed MAC scheme. On the other hand, a large CW may cause large transmission overhead since the voice user may experience an unnecessarily long backoff time before transmission. The optimal CW is the CW that leads to the highest efficiency of the proposed contention-based MAC scheme, whereas the efficiency is expressed as follows:

$$E(CW) = \frac{p^s \cdot T_v}{p^s \cdot \overline{T}_s + p^c \cdot \overline{T}_c} \tag{15}$$

where p^s is the probability that a successful voice transmission occurs in a slot, whereas p^c is the probability that a collision occurs in a slot. T_v is the voice packet transmission time, excluding the overhead (i.e., the backoff time), $\overline{T}_s$ is the average time of a successful packet transmission including the overhead, and $\overline{T}_c$ is the average time of a collision, including the overhead.

Let T_m denote the size of a minislot. $\overline{T}_s$ can be obtained by (16), shown at the bottom of the page, where the term $\binom{N_s}{i} \cdot \rho^i \cdot (1-\rho)^{N_s-i}$ indicates the probability that i out of N_s secondary users are active in a random slot, and the term in the second summation in the numerator indicates that j minislots are required before a successful transmission when an active secondary user chooses a backoff timer value j, and other active secondary users choose backoff timer values larger than j.

Similarly, $\overline{T}_c$ can be obtained by (17), shown at the bottom of the page, where the term in the second summation in the numerator indicates that $k(\geq 2)$ active secondary users choose a backoff timer value j, and other active secondary users choose backoff timer values larger than j.

It is straightforward to see that

$$\begin{aligned} p^s =& \sum_{i=1}^{N_s} \binom{N_s}{i} \cdot \rho^i \cdot (1-\rho)^{N_s-i} \\ &\cdot \left(\sum_{j=1}^{CW} i \cdot \frac{1}{CW} \cdot \left(\frac{CW-j}{CW} \right)^{i-1} \right) \\ p^c =& \sum_{i=1}^{N_s} \binom{N_s}{i} \cdot \rho^i \cdot (1-\rho)^{N_s-i} \\ &\cdot \left(\sum_{k=2}^{i} \sum_{j=1}^{CW} \binom{i}{k} \cdot \left(\frac{1}{CW} \right)^k \cdot \left(\frac{CW-j}{CW} \right)^{i-k} \right). \end{aligned}$$

From (15)–(17), $E(CW)$ can be obtained. The optimal contention window is $CW^* = \arg\max_{CW} E(CW)$.

VI. Performance Evaluation and Discussions

We validate our analytical results by simulation using MATLAB. The simulation for each run consists of 10 000 time slots. We choose $D = 450$ (in units of time slots). The voice packet dropping rate bound P_l is set as 1%. We vary the other parameters such as N_p, θ, α, β, and CW to investigate their impact on the voice service capacity. In the simulation, the voice call of each secondary user is generated at a random time at the beginning of the simulation and lasts for the whole simulation time. When active, the voice traffic has constant bit rate, and the voice packet interarrival time is $T = 40$ slots. In the simulation, the channel status of each slot is randomly generated according to the primary-user activity parameters (i.e., θ for the independent channel and α and β for the correlated channel). Each secondary user first checks the channel status. If the channel is idle, the proposed MAC scheme is performed accordingly. In the simulation, we keep track of the delay of each packet. A packet will be dropped if its delay exceeds the delay bound. The packet loss ratio is obtained as the ratio of the number of dropped packets to the total number of generated packets. We gradually increase the number of secondary users and repeat the simulation until the packet loss ratio is larger than the packet loss rate bound P_l. The maximum

$$\overline{T}_s = \frac{\sum_{i=1}^{N_s} \binom{N_s}{i} \cdot \rho^i \cdot (1-\rho)^{N_s-i} \cdot \left(\sum_{j=1}^{CW} i \cdot \frac{1}{CW} \cdot \left(\frac{CW-j}{CW} \right)^{i-1} \cdot (j \cdot T_m + T_v) \right)}{\sum_{i=1}^{N_s} \binom{N_s}{i} \cdot \rho^i \cdot (1-\rho)^{N_s-i} \cdot \left(\sum_{j=1}^{CW} i \cdot \frac{1}{CW} \cdot \left(\frac{CW-j}{CW} \right)^{i-1} \right)} \tag{16}$$

$$\overline{T}_c = \frac{\sum_{i=2}^{N_s} \binom{N_s}{i} \cdot \rho^i \cdot (1-\rho)^{N_s-i} \cdot \left(\sum_{k=2}^{i} \sum_{j=1}^{CW} \binom{i}{k} \cdot \left(\frac{1}{CW} \right)^k \cdot \left(\frac{CW-j}{CW} \right)^{i-k} \cdot (j \cdot T_m + T_v) \right)}{\sum_{i=2}^{N_s} \binom{N_s}{i} \cdot \rho^i \cdot (1-\rho)^{N_s-i} \cdot \left(\sum_{k=2}^{i} \sum_{j=1}^{CW} \binom{i}{k} \cdot \left(\frac{1}{CW} \right)^k \cdot \left(\frac{CW-j}{CW} \right)^{i-k} \right)} \tag{17}$$

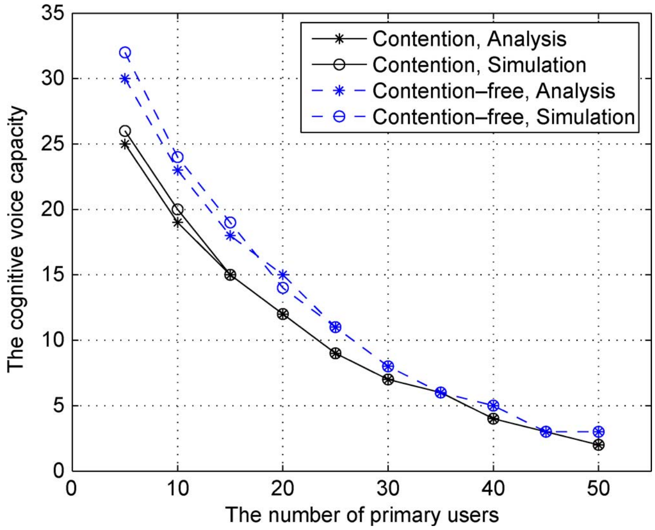

Fig. 6. Cognitive voice capacity with $\theta = 95\%$.

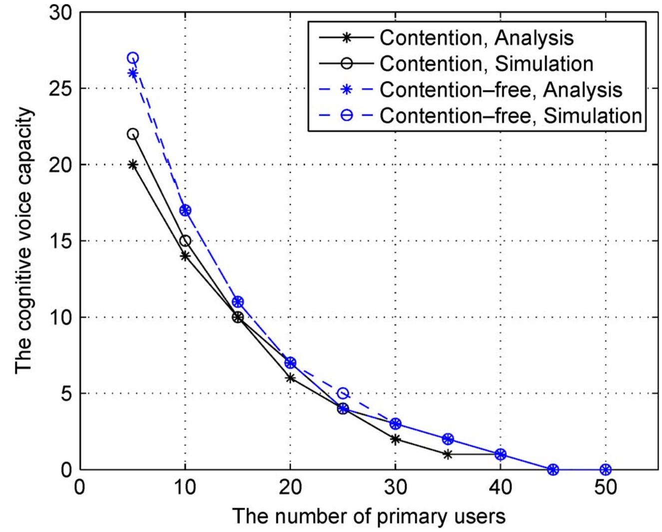

Fig. 7. Cognitive voice capacity with $\theta = 92\%$.

number of secondary users, which yields the packet-loss ratio less than P_l, is defined as the system capacity obtained from simulation.

A. Voice-Service Capacity for the Independent Channel State Model

First, we fix the value of θ as 95% and the contention window size CW as 40 minislots and vary the number of primary users N_p in the system. The cognitive voice-service capacity (i.e., the maximum number of secondary voice users N_s that can be supported with QoS guarantee) is calculated by using the proposed analytical model. The analytical results are shown in Fig. 6. It can be seen that, when the number of primary users increases, the capacity of secondary voice users decreases due to the reduced available channel resources. The capacity of the contention-free medium access is larger than that of the contention-based medium access. The reason is that the contention-free medium access can more efficiently utilize the channel than the contention-based medium access by eliminating collisions, resulting in a larger capacity. The simulation results are also shown in Fig. 6. It can be seen that the simulation results match well with the analytical results in all cases.

Next, we change θ from 95% to 92%, and the corresponding capacities of the two proposed medium-access schemes are shown in Fig. 7. It is clear that, with lower θ, fewer channel resources are left for the secondary users, leading to a smaller capacity. Again, the simulation results match with the analytical results.

For the contention-based medium access, the choice of CW also has an impact on the capacity. Fig. 8 shows the capacity of the contention-based medium access with two different CW sizes. When the number of primary users is small, a relatively large number of secondary users can be admitted. A larger CW can effectively reduce the collisions among them, resulting in a slightly larger capacity. However, with the increase in primary users, fewer secondary users can be admitted into the system (e.g., less than seven secondary users can be admitted when the number of primary users is larger than 30, as shown in Fig. 8). With a small number of contenders, collisions rarely occur. Therefore, a larger CW cannot help in increasing the capacity. Note that with a small size of CW, the collision probability may be high. A large size of CW can alleviate the collisions. However, the overhead also increases.

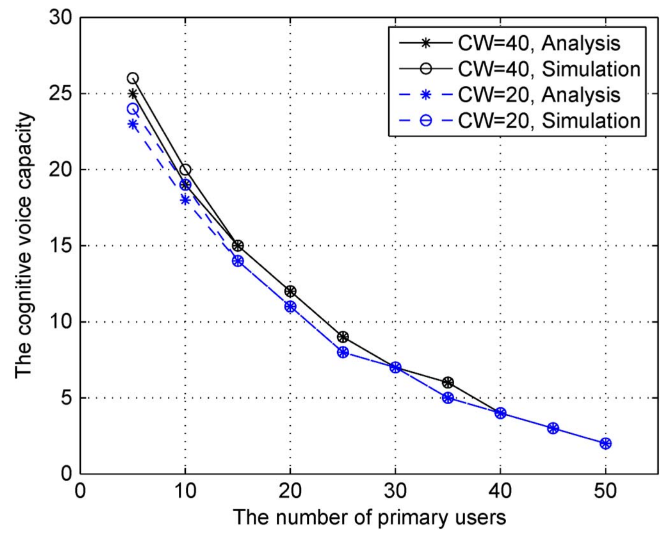

Fig. 8. Cognitive voice capacity of the contention-based medium access with $\theta = 95\%$ and different CW size.

B. Voice-Service Capacity for the Correlated Channel State Model

First, we fix the value of α (i.e., the probability that the channel moves from busy to idle) at 40% and contention window size CW at 40 minislots, and we vary the value of β (i.e., the probability that the channel moves from idle to busy). The analytical results and simulation results of the voice-service capacity for the two proposed MAC schemes are shown in Table I. It can be seen that the simulation results match well with the analytical results in most cases. When the value of β increases, the voice-service capacity decreases because the probability of the channel being busy increases

TABLE I
COGNITIVE VOICE CAPACITY N_s^* WITH $\alpha = 40\%$ AND DIFFERENT β'S

β		10%	20%	30%	40%	50%	80%
Contention based CW = 20	Analysis	21	17	15	14	13	10
	Simulation	21	17	15	14	13	10
Contention based CW = 40	Analysis	25	21	18	16	15	11
	Simulation	25	21	18	17	15	11
Contention-free	Analysis	32	26	22	20	17	12
	Simulation	33	27	23	20	18	13

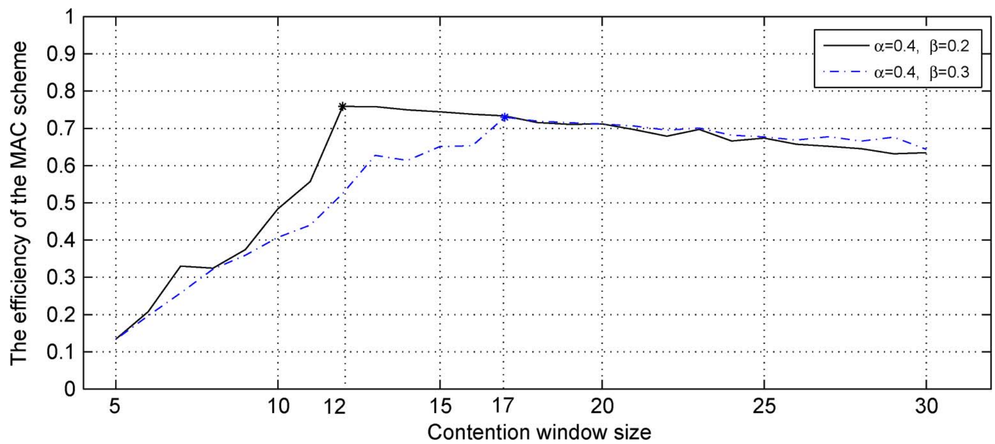

Fig. 9. Efficiency (simulation result) of the contention-based MAC scheme with different CW size.

with the increase in β, resulting in fewer available resources for secondary users. Again, the capacity of the contention-free medium access is larger than that of the contention-based medium access.

Table I also shows the capacity of the contention-based medium access with two different CW sizes. It can be seen that, with a smaller CW, the capacity decreases due to the increasing collisions among secondary users. Again, the simulation results match well with the analytical results.

Next, we investigate how the correlation of the channel affects the system capacity. We take contention-based medium access as an example and consider two cases. In the first case, both α and β are set as 50%, and in the second case, they both are set as 3%. It can be seen that, in both cases, the channel idle probability is $\alpha/(\alpha+\beta) = 50\%$. However, the channel in the first case is less correlated than that in the second case. We set CW as 20 minislots. It is observed that the voice-service capacity is 11 when $\alpha = \beta = 3\%$ and 14 when $\alpha = \beta = 50\%$. The reason can be explained as follows: When $\alpha = \beta = 3\%$, if the channel is busy at a time slot, it is very likely that the channel will remain busy for a long time, resulting in secondary users waiting for a long time to receive service. Consequently, more voice packets from the secondary users may violate the delay bound, leading to a higher packet-loss probability. To maintain the packet-loss probability below the threshold P_l, fewer secondary users can be supported.

C. Optimal Contention Window Size

As the analysis of the optimal contention window size is independent of the channel state model, here, we investigate the results in the correlated channel state model to validate the analysis. We set the length of a minislot $T_m = 5\ \mu$s, the voice-packet-transmission time $T_v = 112\ \mu$s, and the number of secondary users $N_s = 15$. Two different cases of primary-user activity are considered. When $\alpha = 0.4$ and $\beta = 0.2$, the optimal contention window CW^* obtained from the analytical model is 12. When $\alpha = 0.4$ and $\beta = 0.3$, the corresponding CW^* is 17. We also run simulations to obtain the efficiency of the proposed contention-based scheme with different contention window sizes, and the results are shown in Fig. 9. In the simulation, we record the time (e.g., t_1) used for successful voice-packet transmissions (excluding the backoff time) and the time (e.g., t_2) used for all successful and collided voice-packet transmissions (including the backoff time). The efficiency is the ratio of t_1 to t_2. The optimal CW values are marked by $(*)$ in Fig. 9, which also shows that the optimal CW is 12 when $\alpha = 0.4$ and $\beta = 0.2$ and 17 when $\alpha = 0.4$ and $\beta = 0.3$. The simulation results verify the accuracy of the analytical optimal values. When β increases from 0.2 to 0.3, the optimal contention window size increases because the channel idle time decreases when β increases. As a result, the number of active secondary users (i.e., the users with nonempty buffer) increases. Therefore, a larger contention window is required to alleviate the collisions.

VII. CONCLUSION

In this paper, we have proposed two cognitive MAC schemes to support the cognitive voice service in the presence of primary users. One is the contention-based scheme, and the other is the contention-free scheme. An analytical model has been developed to investigate the performance of the two cognitive medium-access schemes (i.e., the voice-service capacity). The optimal contention window size for the contention-based scheme has also been derived from the analytical model. In the analytical model, we have considered two different channel busy/idle state models. One is the independent channel-state model, and the other is the correlated channel-state model. Our analysis has been validated by simulations. Comparing the two proposed schemes, the contention-based MAC scheme is simpler to implement but yields a relatively smaller capacity due to the collisions among the secondary users. This work can provide useful insights to voice-service support over cognitive radio networks.

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