Improving Voice-Service Support in Cognitive Radio Networks

Alireza Homayounzadeh and Mehdi Mahdavi

Voice service is very demanding in cognitive radio networks (CRNs). The available spectrum in a CRN for CR users varies owing to the presence of licensed users. On the other hand, voice packets are delay sensitive and can tolerate a limited amount of delay. This makes the support of voice traffic in a CRN a complicated task that can be achieved by devising necessary considerations regarding the various network functionalities. In this paper, the support of secondary voice users in a CRN is investigated. First, a novel packet scheduling scheme that can provide the required quality of service (QoS) to voice users is proposed. The proposed scheme utilizes the maximum packet transmission rate for secondary voice users by assigning each secondary user the channel with the best level of quality. Furthermore, an analytical framework developed for a performance analysis of the system, is described in which the effect of erroneous spectrum sensing on the performance of secondary voice users is also taken into account. The QoS parameters of secondary voice users, which were obtained analytically, are also detailed. The analytical results were verified through the simulation, and will provide helpful insight in supporting voice services in a CRN.

Keywords: Cognitive Radio, Quality of Service, Spectrum Allocation, Voice Service, Opportunistic Communication, Discrete Markov Chain.

Manuscript received Oct. 15, 2015; revised Dec. 30, 2015; accepted Feb. 18, 2016.

Alireza Homayounzadeh (corresponding author, a_homayounzade@ec.iut.ac.ir) and Mehdi Mahdavi (m_mahdavi@cc.iut.ac.ir) are with the Department of Electrical and Computer Engineering, Isfahan University of Technology, Iran.

I. Introduction

Fixed spectrum allocation has been determined to be inefficient because a large portion of the spectrum is underutilized. Cognitive radio (CR) technology has been proposed for the opportunistic use of the spectrum, allowing the valuable frequency spectrum to be utilized more efficiently [1]. The idea of CR can overcome the paradox between the shortage of frequency resources and the inefficient use of the licensed spectrum. In a cognitive radio network (CRN), users are divided into two categories: licensed or primary users (PUs) with license to use the spectrum, and CR or secondary users (SUs) who can utilize the spectrum opportunistically. CRNs are based on the ingenious idea of determining the unoccupied portions of the frequency spectrum and adjusting the transmission frequencies for CR users according to the activity of the licensed users.

QoS provisioning for the SUs in a CRN is a complicated task with challenges to the various functionalities of the network [2]–[4]. Introducing various new secondary applications in a CRN that are in need of strict QoS parameters has signified the importance of this problem.

CRNs can be categorized into infrastructure-based (centralized) networks and decentralized (distributed) networks [5]. In the former, the network is dependent on the presence of an infrastructure that manages the spectrum allocation among the SUs and may also participate in the transmission of the SU packets. Conversely, decentralized cognitive radio networks (DCRNs) do not depend on such an infrastructure and are therefore more practical. Hence, DCRNs have gained more interest, although spectrum management in such networks is more complicated [6]–[10].

To provide the QoS requirements of SUs, special

http://dx.doi.org/10.4218/etrij.16.0115.0911

considerations should be devised for the various functionalities of the network, including spectrum sensing, spectrum decision and allocation, and admission control [11]–[14]. Voice service is one of the most demanding service types in current networks [15], [16]. Voice packets are delay sensitive. Hence, fluctuations of the available spectrum in a CRN, which is a result of the presence of the PUs, makes it complicated to provide the required QoS for such services. In addition, spectrum sensing has an important role in a CRN [17], [18]. Misdetection and false alarm in spectrum sensing may result in a collision of the transmitted packets and an underutilization of the available spectrum, respectively [19]. Hence, they have an important effect on the QoS of secondary voice users. Furthermore, the quality of the available frequency channels (that is, the SNR values) varies owing to noise and fading of wireless channel. This results in a variable data transmission rate on the frequency channels. Such an issue should also be considered to effectively satisfy the QoS requirements of voice users.

In recent years, the QoS provisioning for SUs in a CRN has been investigated. However, only a few studies have focused on voice service in a CRN. The authors in [14] investigated the performance of a CRN in which the arrival of the SUs is modeled by a Markovian arrival process, and the service time of the users is considered to have a phase-type distribution. Their analysis was also conducted at the call level. The authors in [20] proposed a queueing framework for the problem of opportunistic spectrum access of the SUs. The voice service capacity of a CRN was investigated in [21]. Two cognitive MAC schemes for the spectrum access of secondary voice users were proposed, and an analytical model was constructed to determine the voice service capacity of the two proposed schemes. In [22], the capacity of VoIP service in a CRN in the presence of spectrum sensing errors was analyzed.

The performance of VoIP traffic in a CRN was studied in [23]. The arrivals of both the PUs and SUs were modeled using Poisson processes. The authors in [24] studied the spectrum allocation and QoS provisioning for SUs. They considered the Poisson arrival and departure rates for the PU traffic. Two types of traffic, streaming and non-streaming, were considered for the SUs. The non-streaming type is simple traffic in which packet arrivals occur according to the Poisson process. However, in the streaming type, each SU generates traffic in a coinstantaneous manner. QoS provisioning for the real-time secondary traffic in a CRN was investigated in [25]. Here, the aggregate traffic of SUs was modeled using a discrete batch Markovian arrival process. In addition, the spectrum sensing is considered to be perfect.

In this paper, we try to overcome the shortcomings of these previous studies. The specific contributions of our work are as follows:

- We consider a model for voice traffic that captures the specific features of the voice packets. The on-off nature of the voice users is taken into account in the applied traffic model. Furthermore, the fixed delay bound of the voice packets is properly considered the packet transmission scheme.
- A packet scheduling scheme is proposed in which the delay-sensitive nature of voice packets is taken into account. Moreover, the quality of the frequency channels is considered in the proposed scheduling scheme. The proposed scheme tries to increase the throughput of secondary voice users in the network because it assigns the channels with the highest SNR values to the SUs in each time slot.
- The practical conditions of the system concerning the spectrum sensing are taken into account. In other words, the effects of misdetection and false alarm on the packet transmission rates of the SUs are investigated. Furthermore, the effects of these types of errors on the overall performance of secondary voice users in the network are studied.
- A novel analytical framework is presented through which the achievable QoS parameters of the secondary voice users are investigated.

The remainder of this paper is organized as follows: The system model is described in Section II. The problem formulation is then presented in Section III. The QoS parameters of secondary voice users are also derived analytically in this section. Simulation results are then provided in Section IV, and some concluding remarks are given in Section V.

II. System Model

1. Network Structure

We consider a homogenous DCRN consisting of secondary voice users who opportunistically use the spectrum originally dedicated to the PUs. In the considered network, each secondary transmitter can directly communicate with the corresponding secondary receiver. In other words, SUs in the network form transmitter-receiver pairs. It is assumed that the SUs in the network are clustered into several groups, each of which has a small coverage area. The structure of a typical group in the network is shown in Fig. 1. The group establishment procedure is as follows. Each SU sets itself as the Head of Group (HoG) with a specified probability p. It then broadcasts the result to the other SUs. Next, each non-HoG SU joins the group whose HoG can be reached using the least amount of communication energy. Therefore, when an SU enters the network, it joins one of the existing groups if it is in its transmission range, or forms a new group itself. We assume

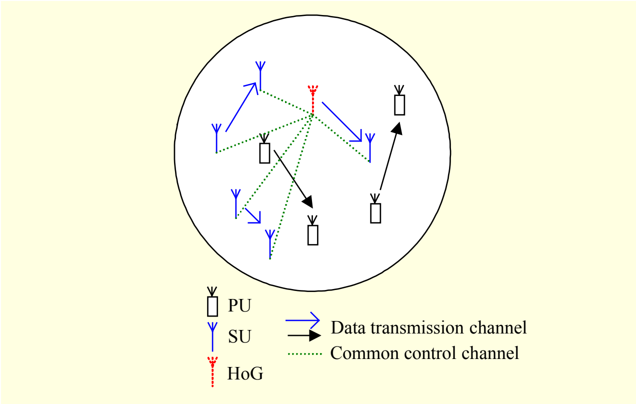

Fig. 1. Structure of a typical group in the network.

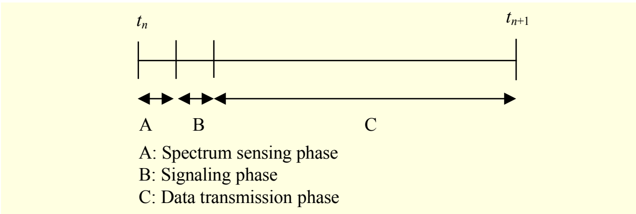

Fig. 2. Time slot structure.

that each SU in a group can directly communicate with other members of the group. Although any other group establishment procedure can be applied in the network, we use the one mentioned previously, which does not increase the complexity of the signaling in the network. If the HoG dies, one of the other SUs in the group is selected randomly as the new HoG. This can be achieved by initiating a random timer by each SU in the group and selecting the user whose timer first reaches zero. The new HoG then informs the other members of the group on a low-bandwidth common control channel (CCC), which is used for the signaling of the SUs in the group. Our goal is to investigate the performance of secondary voice communications within a typical group. The network is also considered to be homogenous. Therefore, it can be assumed that the number of SUs that leave a typical group equals the number of SUs that join the group from the neighboring groups.

It is also assumed that a low-bandwidth group control channel is used in the network for the coordination and signaling between the heads of the groups. The frequency spectrum used by the SUs of each group does not overlap with the neighboring groups. This can be achieved through the coordination of the heads of the groups. It is assumed that the frequency bandwidth of each group is divided into C equal bandwidth channels.

The system is considered to be time-slot based in which the time axis is divided into slots of equal length. The time slot duration is denoted by T. Each time slot has three phases, as shown in Fig. 2. At the beginning of each time slot, all SUs sense the spectrum and report their observations to the HoG. In the signaling phase, the HoG assigns the channels to the SUs of the group for a packet transmission and informs them on the CCC. This process will be described in more detail in section IV. Next, the SU packets are transmitted during the data transmission phase.

2. Voice Traffic Model

The packetized voice traffic of a single user can be generally described using an on-off model [22]. During 'on' or 'talk' periods, packets (which are considered to be of a fixed size) are generated at a constant rate. On the other hand, no packets are generated during 'off' or 'silence' periods. The 'on' and 'off' periods have exponential distributions with a means of $\overline{T}_{\text{on}} = 1/\alpha$ and $\overline{T}_{\text{off}} = 1/\beta$, respectively. A voice user is in an 'off' state with probability $P_{\text{off}} = \beta/(\alpha+\beta)$. This probability is $P_{\text{on}} = 1 - P_{\text{off}}$ for an 'on' state. It has been demonstrated that we can efficiently approximate the superposition of N on-off voice sources using a two-state Markov-modulated Poisson process (MMPP) [26], [27]. Consequently, herein we use a two-state MMPP to represent the aggregate traffic of voice sources in a typical group. The two-state MMPP is represented by the Poisson arrival rate matrix $\boldsymbol{\Lambda}_p$ and the transition rate matrix $\boldsymbol{R}_p$, which are defined as follows:

$$\boldsymbol{\Lambda}_p = \begin{bmatrix} \lambda_1 & 0 \\ 0 & \lambda_2 \end{bmatrix}, \quad \boldsymbol{R}_p = \begin{bmatrix} -s_1 & s_1 \\ s_2 & -s_2 \end{bmatrix}, \tag{1}$$

where s_1 denotes the transition rate from state 1 to state 2 in the two-state MMPP model. Similarly s_2 denotes the transition rate from state 2 to state 1. In addition, λ_1 and λ_2 are the arrival rates in states 1 and 2 of the MMPP model, respectively. The parameters of the MMPP ($s_1, s_2, \lambda_1, \lambda_2$) should be matched with the parameters of the individual on-off source (α, β). Various methods have been presented for this purpose. Here, we choose the index of dispersion for counts owing to its proper matching performance and low complexity compared to other methods [22]. Using this method, the four parameters are calculated as follows [26], [27]:

$$s_1 = \frac{2(\lambda_2 - \lambda_{\text{avg}})(\lambda_{\text{avg}} - \lambda_1)^2 (s_1 + s_2)^2 (\lambda_1 s_2 - \lambda_2 s_1)}{2(\lambda_2 - \lambda_1)(\lambda_1 - \lambda_2)^2 \lambda_{\text{avg}} s_1 s_2}, \tag{2}$$

$$s_2 = \frac{2(\lambda_2 - \lambda_{\text{avg}})^2 (\lambda_{\text{avg}} - \lambda_1)(s_1 + s_2)^2 (\lambda_1 s_2 - \lambda_2 s_1)}{2(\lambda_2 - \lambda_1)(\lambda_1 - \lambda_2)^2 \lambda_{\text{avg}} s_1 s_2}, \tag{3}$$

$$\lambda_1 = A . \frac{\sum_{i=0}^{M} i.\tau_i}{\sum_{i=0}^{M} \tau_i}, \tag{4}$$

http://dx.doi.org/10.4218/etrij.16.0115.0911

$$\lambda_2 = A.\frac{\sum_{i=M+1}^{N} i.\tau_i}{\sum_{i=M+1}^{N} \tau_i}. \tag{5}$$

Here, N is the total number of secondary voice users in the group, and A is the packet arrival rate of one voice source in an 'on' state. Furthermore, $\lambda_{\text{avg}} = N.A.P_{\text{on}}$, $M = \lfloor N.P_{\text{on}} \rfloor$ and $\tau_i = \binom{N}{i} P_{\text{on}}^i \left(1 - P_{\text{on}}\right)^{N-i}$.

As an efficient model for the packet arrival process, discrete time and continuous time MMPPs can be used interchangeably [28]. This is true when the time interval used in the discrete time version is small in comparison with the average sojourn times in each state of the underlying Markov chain. Here, we assume $T \ll T_{\text{on}}$ and $T \ll T_{\text{off}}$, where T_{on} and T_{off} denote the duration of the 'on' and 'off' periods, respectively. Therefore, we can use the discrete-time two-state MMPP as the aggregate traffic model for the SUs.

We define a set of probability matrices as

$$\boldsymbol{D}_k = \begin{bmatrix} (\lambda_1 T)^k e^{-\lambda_1 T} / k! & 0 \\ 0 & (\lambda_2 T)^k e^{-\lambda_2 T} / k! \end{bmatrix}. \tag{6}$$

Matrix $\boldsymbol{D}_k$ corresponds to k packet arrivals in the time slot duration. The diagonal elements of matrix $\boldsymbol{D}_k$ are the probabilities of k packets arriving in the time slot duration.

The matrix $\boldsymbol{\Phi} = (\boldsymbol{\Lambda}_p - \boldsymbol{R}_p)^{-1} \boldsymbol{\Lambda}_p$ keeps track of the phase of the packet arrival process model [26]. Here, $\boldsymbol{\Phi}$ is a 2 × 2 matrix because the underlying Markov chain of the MMPP model has two states. In addition, $\boldsymbol{\Phi}(i, j); i, j \in \{1, 2\}$, that is, the $(i, j)^{th}$ element of matrix $\boldsymbol{\Phi}$, is the probability of the underlying chain of the two-state MMPP model transiting from state i in the previous time slot to state j in the current time slot.

The maximum tolerable waiting time of each voice packet is denoted by $W_{\max}$. We consider $W_{\max} = q.T$, where T is the time-slot duration and q is an integer. We also assume that all of the packets scheduled for transmission during a time slot are transmitted before that particular time slot ends.

3. Channel Modeling

The occupancy of each of the frequency channels by the PUs can be modeled using a two-state discrete time Markov chain, as depicted in Fig. 3 [20], [22]. State '0' indicates a case in which the channel is occupied by a PU (busy), and state '1' indicates a case in which the channel is free (idle). In this model, P_{01} and P_{10} are the transition probabilities of the channel transiting from a busy state in the previous time slot to an idle state in the current time slot, and vice versa, respectively.

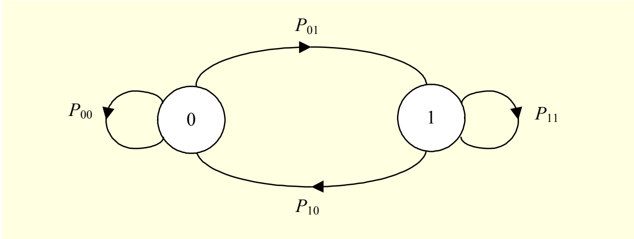

Fig. 3. Channel occupancy model in a CRN.

The transition probability matrix of this Markov chain is denoted by $\boldsymbol{P}_c$:

$$\boldsymbol{P}_c = \begin{bmatrix} P_{00} & P_{01} \\ P_{10} & P_{11} \end{bmatrix}. \tag{7}$$

Note that the correlation of the channel occupancy among subsequent time slots is considered in this model. The probability that a channel will be busy (P_{busy}) is as follows [22]:

$$P_{\text{busy}} = \frac{P_{10}}{P_{01} + P_{10}}. \tag{8}$$

This can also be interpreted as the PU activity factor for each frequency channel. The detection accuracy of the channel state through spectrum sensing is affected by certain aspects of the system, including the hidden terminal problem, limited sensitivity of the detectors, and channel noise and fading. Such factors can affect the reliability of the spectrum sensing. Hence, it can be readily seen that the channel state detection forms a binary hypothesis test that can be affected by two types of errors, that is, misdetection and false alarm, the probabilities of which are denoted by P_m and P_f, respectively. These can be defined as follows:

$$\begin{aligned} P_m &= \Pr[H_1 \mid H_0 \text{ is true}], \\ P_f &= \Pr[H_0 \mid H_1 \text{ is true}], \end{aligned} \tag{9}$$

where H_0 (H_1) denotes a busy (idle) channel. The effects of spectrum sensing errors on the packet transmission of the SUs will be taken into account in the analysis presented in Section III.

In wireless communication systems, noise and fading can affect the quality of the frequency channels. The packet transmission rate on each frequency channel depends on the quality of the channel, that is, its SNR value. This should be considered an important fact in the performance analysis of the system. In this paper, the frequency channels are thought to be slowly fading, and the channel fading amplitude distribution is considered to be of a Nakagami-m distribution. The fading power gain of each frequency channel is divided into k ranges $0 = \sigma_0 < \sigma_1 < \ldots < \sigma_i < \ldots. < \sigma_k = \infty$. Hence, each fading channel can be modeled as a Finite-State Markov Chain (FSMC) with k states [29]. State i in this model corresponds to the fading power gain (γ) within the range of $\sigma_i < \gamma < \sigma_{i+1}$. The transition probability matrix of this FSMC, which is denoted by $\boldsymbol{R}$ can be obtained as in [30].

$$\boldsymbol{R} = \begin{bmatrix} Pr(0\to 0) & Pr(0\to 1) & \dots & Pr(0\to k) \\ \vdots & & \ddots & \vdots \\ Pr(k\to 0) & Pr(k\to 1) & \dots & Pr(k\to k) \end{bmatrix}. \quad (10)$$

Here, $Pr(l \to m); 0 \le l,\ m \le k$ indicates the probability that the channel will transit from state 'l' in the previous time slot to state 'm' in the current time slot. The channel state is assumed to be fixed in the time-slot duration but can change from one time slot to another. Each state of the channel corresponds to the transmission rate during the time slot. We assume that the adaptive modulation and coding (AMC) is applied to adapt the transmission parameters of the SUs. Hence, we can consider that when channel i is in state $l; 0 \le l \le k$ in the n^{th} time slot, the packet transmission rate of this channel is $r_i^{(n)} = d.l$, where d is an integer that depends on the modulation type, the time slot duration, and the size of the voice packet. State '0' in this model corresponds to $r_i^{(n)} = 0$. As mentioned before, the total number of frequency channels of a group is denoted as C. Hence, the maximum number of packets that can be transmitted in a time slot is $V_{\max} = d.b$, where $b = k.C$.

4. Packet Transmission Scheduling

As mentioned previously, we consider a voice application as a secondary service. Voice packets are delay sensitive and have a limited waiting time. Hence, they can be buffered only for a certain amount of time. We assume that each packet has a time stamp indicating its time of arrival at the SU buffer. Those buffered packets whose waiting time exceeds the maximum tolerable value are dropped because they will be useless on the receiver side. Taking this issue into account, the HoG limits the number of voice packets admitted in each group. Because the maximum tolerable waiting time of each voice packet is $W_{\max} = q.T$ and the maximum packet transmission rate in each time slot is given by $V_{\max} = d.b$, the maximum number of waiting packets is $l_{\max} = q.d.b$. Therefore, the number of admitted packets in each time slot is limited to $L = l_{\max} + V_{\max} = (q+1).d.b$. As a result, some of the voice packets generated are lost because they do not obtain permission to enter the system. Those packets that are not admitted to the group are dropped from the buffers of the SUs.

Herein, we propose a packet-scheduling scheme in which the delay sensitivity of voice packets and the variations of the SNR of the frequency channels are taken into account. The proposed scheduling scheme for the secondary packet transmission is as follows:

- At the beginning of each time slot, that is, during the spectrum-sensing phase, all SUs present in the group sense the frequency channels and report their observations to the HoG on the CCC. The HoG determines the state of each frequency channel and finds the free channels. Each SU also reports the number of packets in its buffer, the time stamp of the packet at its head of queue, and the measured SNR values for each frequency channel.
- The HoG selects the secondary user that has reported the smallest time stamp. The HoG then assigns to this SU the channel with the highest SNR value from the set of free channels. This process is repeated until all of the free channels are assigned for each time slot.
- Next, the HoG determines the packet transmission rate in the time slot based on the channel assignment strategy explained in the previous step, the quality of the assigned channels, and the number of packets in the queue of each SU. Henceforth, the HoG schedules the packets that can be transmitted in the time slot. The SUs are informed of the result of the packet scheduling process during the signaling phase.
- During the transmission phase, the scheduled packets are transmitted on the free channels in their order of arrival. In other words, the scheduled packets of each SU are transmitted in the order of arrival on the free channel with the highest SNR value.

The scheduled packets are transmitted by the end of the time slot. This scheduling scheme is repeated in each time slot. The proposed scheduling algorithm results in an efficient way to utilize the available spectrum as the maximum transmission rates of the frequency channels are used, and at the same time, the delay sensitivity of the voice packets is taken into account.

III. Problem Formulation

In this section, we investigate the performance of secondary voice users in a typical group. To achieve this goal, a discrete-time Markov chain (DTMC) is proposed to model the presence of SUs in the group. The state transitions in the proposed DTMC occur at the beginning of each time slot, that is, at time instants $t_n = t_0 + nT$; n = 0, 1, … . The steps of the DTMC model formulation are described in the following sections.

1. Analytical Model

Let $N_P^{(n)}$ and $V_C^{(n)}$ represent the number of voice packets of all SUs of a typical group and the actual packet transmission rate for the SUs, respectively, at the beginning of the n^{th} time slot, that is, at time t_n. Note that $0 \le N_P^{(n)} \le L$ and $0 \le V_C^{(n)} \le V_{\max}$. In addition, let $\phi^{(n)} = 1, 2$ represent the

http://dx.doi.org/10.4218/etrij.16.0115.0911

phase of the arrival process for the secondary voice packets at time t_n, such that $\phi^{(n)}=1$ and $\phi^{(n)}=2$ correspond to the underlying chain of the packet arrival process in states 1 and 2, respectively. For the sake of simplicity, we remove the superscript '(n)' throughout the remainder of the paper.

Let $\{X(n);\ n = 0, 1, 2, \ldots\}$ be a discrete time stochastic process defined for the state space $\boldsymbol{S}=\{(N_P,\phi,V_C)\mid 0\le N_P\le L, 1\le\phi\le 2, 0\le V_C\le V_{\max}\}$. It can be seen that $\{X(n)\}$ is a DTMC that can keep track of the system states as time passes. To investigate the properties of $\{X(n)\}$, we have to know the probabilities of the packet arrivals and departures, as well as the probabilities of variations of the packet transmission rate for the SUs. These issues are described in the following sections.

2. Derivation of the Secondary Packet Transmission Rate

A. Error-Free Spectrum Sensing

In this section, the packet transmission rate for the SUs in a typical time slot is calculated. Herein, we assume that the spectrum sensing process is perfect. Let the random variable $C_i\in\{0,1\}$ represent the state of the i^{th} channel, where $C_i = 0$ and $C_i = 1$ represent the cases in which the i^{th} channel is occupied by the PUs and is free for the SUs, respectively. The joint state considering the primary user activity and the transmission rate for this channel can then be considered for the state space $\boldsymbol{\Gamma}=\{(C_i,r_i)\mid 0\le C_i\le 1, 0\le r_i\le kd\}$. Let $\boldsymbol{M}$ denote the transition probability matrix for the state space $\boldsymbol{\Gamma}$. This can be calculated as $\boldsymbol{M}=\boldsymbol{P}_{\mathrm{c}}\otimes\boldsymbol{R}$, where $\otimes$ stands for the Kronecker product.

Let us define the random variable (V_m) as the actual packet transmission rate for the SUs of a typical group in the considered time slot, is m where the number of considered frequency channels. Here, V_m can be calculated as $V_m=\sum_{i=1}^{m}C_i r_i$, and is defined for the state space $\boldsymbol{\Omega}_m = \{V_m \mid 0\le V_m\le d.k.m\}$. The packet transmission rate can be calculated recursively as follows:

First, consider a case in which there are two frequency channels. The joint state space considering the primary user activity and the packet transmission rates for the two channels is $\boldsymbol{\Gamma}_2=\{(C_1,r_1,C_2,r_2)\mid 0\le C_1,C_2\le 1, 0\le r_1,r_2\le kd\}$.

The transition probability matrix and the steady state probability vector for $\boldsymbol{\Gamma}_2$ are denoted by $\boldsymbol{T}_2$ and $\boldsymbol{\pi}_2$, respectively. The transition probability matrix $\boldsymbol{T}_2$ is given by $\boldsymbol{T}_2=\boldsymbol{M}\otimes\boldsymbol{M}$. Let $\boldsymbol{\Omega}_2^r$ ($\boldsymbol{\Omega}_2^r\subset\boldsymbol{\Gamma}_2$) denote the set of states with the property $V_2 = r$. Let us denote the transition probability matrix for the state space $\boldsymbol{\Omega}_2$ by $\hat{\boldsymbol{R}}_2$. The transition probability from state l to l' ($l,l'\in\boldsymbol{\Omega}_2$) can then be calculated as

$$\hat{\boldsymbol{R}}_2(l,l')=\frac{\sum_{\boldsymbol{m}\in\boldsymbol{\Omega}_2^l}\sum_{\boldsymbol{m}'\in\boldsymbol{\Omega}_2^{l'}}\boldsymbol{\pi}_2(\boldsymbol{m})\boldsymbol{T}_2(\boldsymbol{m},\boldsymbol{m}')}{\sum_{\boldsymbol{m}\in\boldsymbol{\Omega}_2^l}\boldsymbol{\pi}_2(\boldsymbol{m})}. \tag{11}$$

For $k\ge 3$, we can form the combined state space $\boldsymbol{\Gamma}_k=\{(V_{k-1}, C_k, r_k)\}$ with the transition probability matrix, which is calculated recursively as $\boldsymbol{T}_k=\hat{\boldsymbol{R}}_{k-1}\otimes\boldsymbol{M}$. Thus, $\boldsymbol{T}_C$ is given as $\boldsymbol{T}_C=\hat{\boldsymbol{R}}_{C-1}\otimes\boldsymbol{M}$. We can find the steady state probability vector $\hat{\boldsymbol{\pi}}_C$ for the state space Ω_C by solving equations $\hat{\boldsymbol{\pi}}_C.\hat{\boldsymbol{R}}_C=\hat{\boldsymbol{\pi}}_C$ and $\hat{\boldsymbol{\pi}}_C.\mathbf{1}=1$, where $\hat{\boldsymbol{\pi}}_C$ is a $1\times(b+1)$ vector, which can be written as $\hat{\boldsymbol{\pi}}_C=[\hat{\pi}_0,\ldots,\hat{\pi}_b]$. In addition, $\hat{\pi}_i; 0\le i\le b$ is the probability that the actual packet transmission rate on a typical time slot equals $d.i$.

B. Imperfect Spectrum Sensing

Errors in spectrum sensing can affect the detected packet transmission rate of the SUs. A misdetection can result in a collision of the PUs and SUs. On the other hand, a false alarm results in an underutilization of the frequency channels. Let V_C' denote the detected packet transmission rate for the SUs, that is, when the effect of the spectrum sensing errors are taken into account. This can be calculated as

$$V_C'=\left\lfloor V_C-V_C P_f+\left(V_{\max}-V_C\right)P_m\right\rfloor, \tag{12}$$

where V_C is the actual packet transmission rate, and P_m and P_f denote the misdetection and false alarm probabilities, respectively. Equation (12) can be justified as follows. In each time slot, ($V_C P_f$) packets less than the actual transmission rate (V_C) are transmitted because of the underutilization of the frequency channels caused by a false alarm in the spectrum sensing. On the other hand, a misdetection on the remaining frequency channels can result in an increase in the measured packet transmission rate by an amount equal to $(V_{\max}-V_C)P_m$. Hence, the detected packet transmission rate in a time slot can be calculated using (12).

3. Markov Chain Analysis

The transition probability matrix of the DTMC $\{X(n)\}$ is calculated in this section. To achieve this goal, we should first calculate the packet transmission and packet dropping probabilities. It should be noted that m secondary packets will be transmitted in a time slot if the detected packet transmission rate for the SUs in that time slot equals m. Let $P^s(m|i)$ denote the probability that m secondary voice packets will be scheduled for transmission in the considered group in a particular time slot when the number of secondary packets equals i. Considering the packet transmission rate calculated in Section II, $P^s(m|i)$ can be expressed as

$$P^s(m \mid i) = P\left(V_C' = m \middle| N_P = i\right);\ 0 \le m \le \min(i, V_{\max}). \quad (13)$$

Applying (12), this can be written as

$$P^s(m \mid i) = \begin{cases} \hat{\boldsymbol{\pi}}_C\left(\left\lceil \frac{m - P_m V_{\max}}{d\left(1-(P_f+P_m)\right)} \right\rceil\right) & 0 \le m \le \min(i, V_{\max}), \\ 0 & \text{otherwise}, \end{cases} \quad (14)$$

where $\hat{\boldsymbol{\pi}}_C(r)$ denotes the r^{th} element of vector $\hat{\boldsymbol{\pi}}_C$ and $\lceil x \rceil$ is the ceiling function of x.

Lemma 1. Let $P^w(l|i)$ denote the probability of l packets being dropped because their waiting time has exceeded the maximum acceptable value when the number of secondary packets equals i. Then, $P^w(l|i)$ is given by

$$P^w\left(l \middle| i\right) = \begin{cases} \sum_{k=0}^{i-l} \binom{i-k}{l} \left(\frac{1}{q}\right)^l \left(1-\frac{1}{q}\right)^{(i-k-l)} P^s(k|i) & 0 \le l \le i, \\ 0 & \text{otherwise}. \end{cases} \quad (15)$$

Proof. The maximum waiting time of the voice packets is considered to be $W_{\max} = qT$, where T denotes the time-slot duration. Therefore, each packet can only wait q time slots for transmission, and will otherwise be dropped.

Let us consider a tagged packet present in the system in the n^{th} time slot. Taking the maximum waiting time of the packets into account, the tagged packet should have arrived in one of the $(n-1)^{th}$, $(n-2)^{th}$, … , $(n-q+1)^{th}$ prior time slots. Note that the number of such time slots is q. If the packet had arrived in a $(n-q)^{th}$, $(n-q-1)^{th}$, … time slot, its waiting time would have expired before the n^{th} time slot, and hence it could not be in the system in the n^{th} time slot. From the point of view of the tagged packet, there is no distinction among these q time slots. Hence, the tagged packet can arrive in each of these time slots with equal probability of $(1/q)$. If the tagged packet has to be dropped in the n^{th} time slot (that is, the waiting time of this packet has to expire at the beginning of the n^{th} time slot), it should have arrived in the $(n-q)^{th}$ time slot. In other words, the tagged packet will be dropped in the n^{th} time slot if has arrived in the $(n-q)^{th}$ time slot. This can occur with a probability of $(1/q)$. Thus, the probability that l packets will be dropped in a time slot when the number of secondary packets equals i, and the number of packets scheduled for transmission in that time slot equals k (that is, $N_P^s = k$), is given by

$$P^w\left(l \middle| N_P = i, N_P^s = k\right) = \binom{i-k}{l} \left(\frac{1}{q}\right)^l \left(1-\frac{1}{q}\right)^{(i-k-l)}. \quad (16)$$

The probability that the number of scheduled packets in a time slot will equal k when the number of secondary packets equals i is denoted by $P^s(k \mid i)$, which is given in (14). Henceforth, $P^w(l \mid i)$ can be calculated through (15). ■

Theorem 1. Let us arrange all states of $\{X(n)\}$ lexicographically. The transition probability matrix of the Markov chain $\{X(n)\}$ can then be expressed as

$$\boldsymbol{P} = \begin{bmatrix} \boldsymbol{A}_{0,0} & \boldsymbol{A}_{0,1} & \cdots & \boldsymbol{A}_{0,L} \\ \boldsymbol{A}_{1,0} & \boldsymbol{A}_{1,1} & \cdots & \boldsymbol{A}_{1,L} \\ \vdots & & \ddots & \vdots \\ \boldsymbol{A}_{L,0} & \boldsymbol{A}_{L,1} & \cdots & \boldsymbol{A}_{L,L} \end{bmatrix}, \quad (17)$$

where the sub-matrices $\boldsymbol{A}_{i,j};\ 0 \le i, j \le L$ have dimensions of $2(b+1) \times 2(b+1)$ and are given by

$$\boldsymbol{A}_{i,j} = \left(\sum_{l=\max(i-j,0)}^{i} \sum_{m=0}^{l} \boldsymbol{\Phi}\, \boldsymbol{D}_{l+j-i}\, P^s(m \mid i)\, P^{ws}(l-m \mid i) \right) \otimes \hat{\boldsymbol{R}}_C. \quad (18)$$

Proof. Each sub-matrix $\boldsymbol{A}_{i,j};\ 0 \le i,\ j \le L$ contains the transition probabilities of all states in which there are i secondary packets at t_n transitioning into states where there are j packets at t_{n+1}. In other words, $\boldsymbol{A}_{i,j}$ contains the set of transition probabilities $\Pr\left((i,*,*) \to (j,*',*')\right)$, when all of the states are arranged lexicographically.

- Case $i < j$: In this case, sub-matrices $\boldsymbol{A}_{i,j}$ correspond to situations in which more packets have arrived in the time slot than have departed. Hence, if the sum of the number of scheduled and dropped packets in the time slot is l, then $(l + j - i)$ packets should arrive in that time slot such that $\{X(n)\}$ transits into a new state in which $N_P = j$. This corresponds to the matrix $\boldsymbol{D}_{l+j-i}$. If m packets are scheduled for transmission in the time slot, then $(l-m)$ packets should be dropped because their waiting time exceeds the maximum value. The probabilities of these events are given by $P^s(m \mid i)$ and $P^w(l-m \mid i)$, respectively. The probabilities of the variations of the packet transmission rate, which is one of the state variables of $\{X(n)\}$, is given by the matrix $\hat{\boldsymbol{R}}_C$. The matrix $\boldsymbol{\Phi}$ also keeps track of the phase of the packet arrival process. Thus, $\boldsymbol{A}_{i,j}$ can be calculated as

$$\boldsymbol{A}_{i,j} = \left(\sum_{l=0}^{i} \sum_{m=0}^{l} \boldsymbol{\Phi}\, \boldsymbol{D}_{l+j-i} P^s(m \mid i)\, P^w(l-m \mid i) \right) \otimes \hat{\boldsymbol{R}}_C. \quad (19)$$

- Case $i = j$: In this case, the number of packets arriving in the time slot should be equal to the number of departed packets. If l packets arrive, which corresponds to matrix $\boldsymbol{D}_l$, then l packets should also depart. Other related terms can be explained similarly to case $i < j$. Therefore, the diagonal sub-matrices $\boldsymbol{A}_{i,i}$ can be written as

$$\boldsymbol{A}_{i,i} = \left(\sum_{l=0}^{i} \sum_{m=0}^{l} \boldsymbol{\Phi}\, \boldsymbol{D}_l P^s(m \mid i)\, P^w(l-m \mid i) \right) \otimes \hat{\boldsymbol{R}}_C. \quad (20)$$

http://dx.doi.org/10.4218/etrij.16.0115.0911

- Case $i > j$: In this case, if the sum of the number of scheduled and dropped packets for a time slot is l, then $(l + j - i)$ packets should arrive in the time slot. If the number of packets scheduled for transmission equals m, then $(l - m)$ packets should be dropped owing to the waiting time. Other related terms can be explained similarly to $i < j$. Therefore, in this case, the sub-matrices $\boldsymbol{A}_{i,i}$ are given by

$$\boldsymbol{A}_{i,i} = \left(\sum_{l=i-j}^{i} \sum_{m=0}^{l} \boldsymbol{\Phi} \boldsymbol{D}_{l+j-i} P^s(m \mid i) P^w(l-m \mid i) \right) \otimes \hat{\boldsymbol{R}}_C . \quad (21)$$

From the abovementioned cases, it can be readily seen that all sub-matrices of matrix $\boldsymbol{P}$ can be written in (18). ■

Let $\boldsymbol{\pi}$ denote the steady state probability matrix of the proposed DTMC $\{X(n)\}$. Note that $\boldsymbol{\pi}$ is 1 by the vector $(L + 1) \times (b + 1) \times 2$, and can be obtained using the transition probability matrix $\boldsymbol{P}$ by solving the set of equations $\boldsymbol{\pi}.\boldsymbol{P} = \boldsymbol{\pi}$ and $\boldsymbol{\pi}.\mathbf{1} = 1$. The vector $\boldsymbol{\pi}$ can be written as $\boldsymbol{\pi} = [\boldsymbol{\pi}_0, \dots, \boldsymbol{\pi}_L]$, where each sub-vector $\boldsymbol{\pi}_i ; 0 \le i \le L$ contains the steady state probabilities for all the possible states of the DTMC in which $N_P = i$.

4. Performance Measures

Having found the steady state probability vector $\boldsymbol{\pi}$, we are now able to obtain the performance measures of the SUs.

Average Throughput: The average number of secondary packets transmitted per time slot can be calculated as

$$\overline{T} = \sum_{i=0}^{L} \sum_{j=0}^{b} \min\left(i, V'_{C_j}\right) . \left(\boldsymbol{\pi}_i[j+1] + \boldsymbol{\pi}_i[b+j+2]\right), \quad (22)$$

where V'_{C_j} is the detected transmission rate for the SUs when its corresponding actual value equals $V_C = dj$. Using (12), it can be calculated as $V'_{C_j} = \lfloor dj(1 - P_m - P_f) + P_m V_{\max} \rfloor$. In addition, $\boldsymbol{\pi}_i[m]$ also denotes the m^{th} element of sub-vector $\boldsymbol{\pi}_i$. Equation (22) can be justified as follows. The number of transmitted packets equals $\min(i, V'_{C_j})$ when $\{X(n)\}$ is in a state in which $N_P = i$ (that is, the number of secondary packets equals i) and $V_C = dj ; j = 0,1,\dots,b$. This is true for either of the two phases of the packet arrival process (that is, when ϕ= 1 or 2). It can be noted that the terms $\boldsymbol{\pi}_i[j+1]$ and $\boldsymbol{\pi}_i[b+j+2]$ are the probabilities of states of $\{X(n)\}$ in which $(N_P = i, \phi = 1, V_C = dj)$ and $(N_P = i, \phi = 2, V_C = dj)$, respectively. Therefore, considering the lexicographical arrangement of the states of $\{X(n)\}$, the number of secondary transmitted packets can be calculated through (22). Having found $\overline{T}$, the average number of successfully transmitted packets, excluding those that are destroyed owing to a collision, is given by $\overline{T}_R = (1 - P_m)\overline{T}$. Hence, the average throughput of the system can be found as follows:

$$\tau_R = \left(\overline{T}_R / T\right) n_R \quad \text{(b/s)}, \quad (23)$$

where n_R is the voice packet size.

Average Packet Loss: The average number of voice packet arrivals is

$$\rho = \boldsymbol{\theta} . \left(\sum_k k.\boldsymbol{D}_k \right) . \mathbf{1}, \quad (24)$$

where $\mathbf{1}$ is a column vector of all 1s, and $\boldsymbol{\theta} = [\theta_1 \quad \theta_2]$ is obtained by solving the set of equations $\boldsymbol{\theta\Phi} = \boldsymbol{\theta}$ and $\theta_1 + \theta_2 = 1$. This can be written as $\boldsymbol{\theta} = [Pr(\phi = 1) \quad Pr(\phi = 2)]_{1\times 2}$. Thus, the average number of lost packets is

$$\overline{T}_l = \rho - \overline{T}_R , \quad (25)$$

where $\overline{T}_l$ is the sum of the number of packets that are not admitted to the system, the packets that are lost owing to the expiration of their maximum waiting time, and those packets that are destroyed from a collision, which in turn is the result of misdetection in the spectrum sensing. To achieve an acceptable quality for the voice connections, the average number of lost packets should not exceed a certain percentage of all generated packets (for example, 3%). This imposes a restriction on the secondary voice capacity of the system.

IV. Numerical and Simulation Results

In this section, the analytical results obtained are validated through a simulation using MATLAB. We provide the numerical results on the performance measures of voice users for the proposed CRN. For evaluation purposes and validation of the derived analysis, the input parameters of the network were selected as follows. The number of secondary voice users is considered a variable parameter of the system. The activities of the PUs in each frequency channel are modeled using the mentioned two-state DTMC. It is also assumed that $C = 20$, $T = 50$ ms, $q = 2$, $K = 3$, and $d = 1$. There is a good match between the simulation and analysis results, as shown in Figs. 4 through 9.

The performance of the spectrum sensing in the considered CRN can be evaluated using a receiver operating characteristics (ROC) curve. The ROC curves for various SNR values are provided in Fig. 4. It can be seen that the area under each ROC curve approaches unity for reasonable average SNR values. Therefore, the spectrum sensing achieves an acceptable level of performance in the considered CRN. As expected, a better performance is achieved for higher SNR values.

The packet loss probability for secondary voice users is shown in Fig. 5. In this case, the PU activity factor is considered to be 0.2. It can be seen that the packet loss

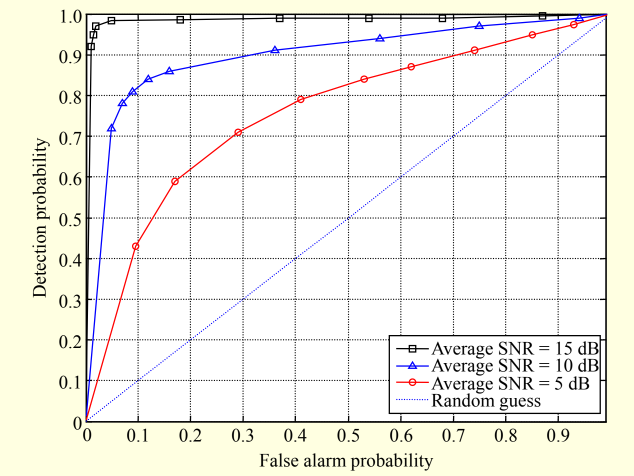

Fig. 4. ROC curves.

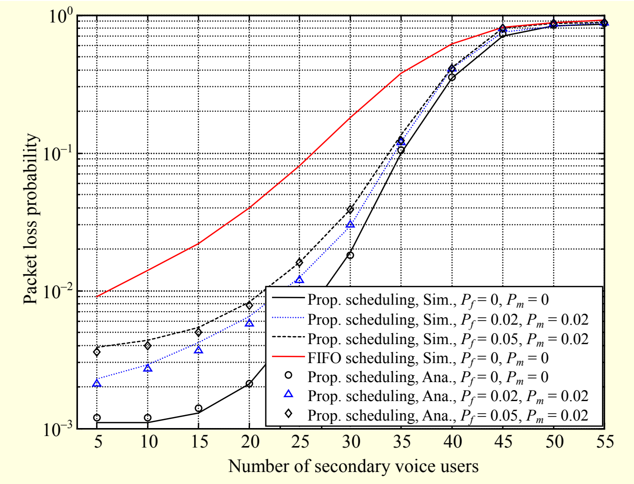

Fig. 5. Packet loss probability for various numbers of secondary voice users.

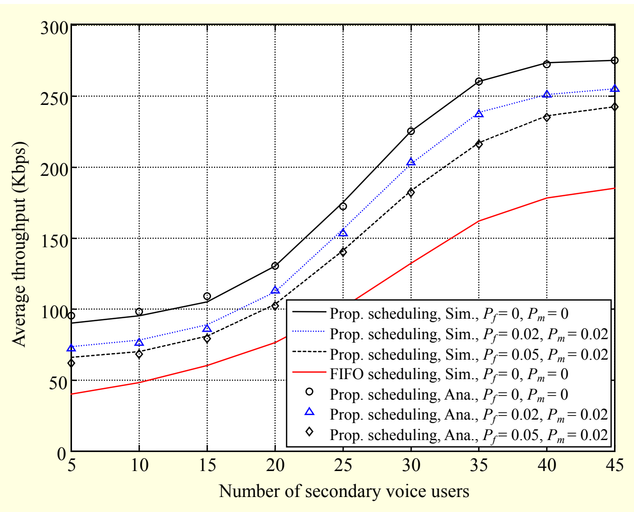

Fig. 6. Average throughput for various numbers of secondary voice users.

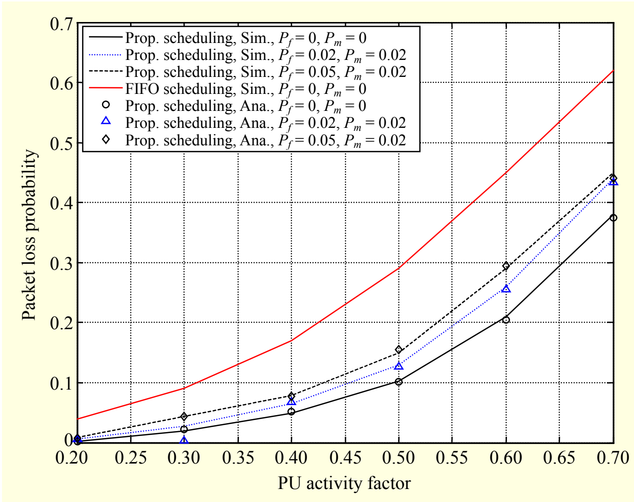

Fig. 7. Packet loss probability for SUs.

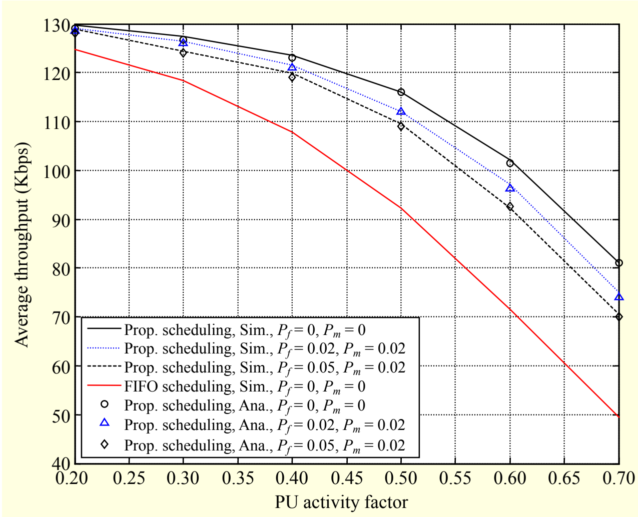

Fig. 8. Average SU throughput.

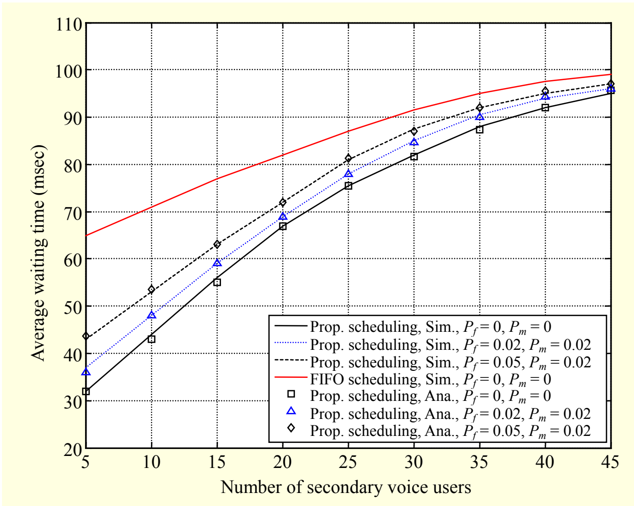

Fig. 9. Average waiting time of secondary voice packets.

ETRI Journal, Volume 38, Number 3, June 2016
http://dx.doi.org/10.4218/etrij.16.0115.0911

probability deteriorates as the number of secondary voice users increases. The reason for this is that, when the number of SUs increases, there are fewer opportunities for the secondary voice packets to be transmitted and more packets are therefore lost. The performance of the proposed scheduling scheme is also compared with first-in, first-out scheduling, which reveals the efficiency of the proposed scheme. Furthermore, it can be seen that errors in the spectrum sensing can increase the packet loss probability.

Figure 6 shows the effect of the number of SUs and the errors in the spectrum sensing on the average throughput of the system. In this case, the PU activity factor is 0.2. As can be seen, when the number of SUs increases, the increasing rate of the throughput slows down, which is the result of the limited capacity of the system for secondary voice users. A false alarm and misdetection in spectrum sensing also decrease the throughput of the system when compared with perfect spectrum sensing. This is true because a false alarm (with misdetection) results in the underutilization of the frequency channels (or packet collisions), and as a result, the throughput of the system degrades.

The effect of the PU activity factor on the packet loss probability of SUs is shown in Fig. 7. Here, the number of secondary voice users is 20. It can be seen that, as the PU activity factor increases, the packet loss probability deteriorates. The reason for this is that there are fewer spectrum opportunities for the SUs when there are more active PUs in the system. Therefore, more SU packets are lost.

Figure 8 shows the effects of the PU activity factor on the average throughput of the system. Here, the number of secondary voice users is 20. It can be seen that, when the PU activity factor increases, the average throughput of the SUs decreases. Of course, the rate of decrease is not too high for lower values of the PU activity factor. However, because the PU activity factor increases in value, the rate of decrease in the average throughput is more severe.

The average waiting time of secondary voice packets is shown in Fig. 9. In this case the PU activity factor equals 0.2. It can be observed that the average waiting time increases when there are more secondary voice users in the system.

V. Conclusion

In this paper, the problem of QoS provisioning for secondary voice users in a cognitive radio network was investigated. An efficient network structure was proposed in which the practical aspects of the system are taken into account. In addition, practical features of the voice traffic were considered in the network analysis conducted. To provide the required QoS for voice users, a novel scheduling scheme was proposed in which the effect of the quality of the frequency channels on the packet transmission rate is taken into account. Furthermore, an analytical framework was developed for our performance analysis of the system. Other practical aspects of the network, such as errors in the spectrum sensing and their effects on the performance of the SUs, were also studied. The proposed framework has an important application in the performance evaluation of secondary voice users, and can provide thorough insight into the support of voice service over cognitive radio networks.

References

[1] J. Mitola and G.Q. Maguire, "Cognitive Radio: Making Software Radios More Personal," *IEEE Pers. Commun.*, vol. 6, no. 4, Aug. 1999. pp. 13–18.

[2] A. Alshemrani, X. Shen, and L. Xie, "QoS Provisioning for Heterogeneous Services in Cooperative Cognitive Radio Networks," *IEEE J. Sel. Areas Commun.*, vol. 29, no. 4, Apr. 2011, pp. 819–830.

[3] V.K. Tumuluru, P. Wang, and D. Niyato, "A Novel Spectrum-Scheduling Scheme for Multichannel Cognitive Radio Network and Performance Analysis," *IEEE Trans. Veh. Technol.*, vol. 60, no. 4, Feb. 2011, pp. 1849–1858.

[4] H. Su and X. Zhang, "Cross-Layer Based Opportunistic MAC Protocols for QoS Provisionings over Cognitive Radio Wireless Networks," *IEEE J. Sel. Areas Commun.*, vol. 26, no. 1, Jan. 2008, pp. 118–129.

[5] I.F. Akyildiz et al., "Next Generation/Dynamic Spectrum Access/Cognitive Radio Wireless Networks: A Survey," *Comput. Netw.*, vol. 50, no. 13, Sept. 2006, pp. 2127–2159.

[6] D. Niyato and E. Hossain, "Medium Access Control Protocols for Dynamic Spectrum Access in Cognitive Radio Networks: A Survey," in *Cognitive Radio Networks*, Florida, USA: CRC Press, 2008, pp. 179–213.

[7] Q. Zhao et al., "Decentralized Cognitive MAC for Opportunistic Spectrum Access in Ad Hoc Networks: A POMDP Framework," *IEEE J. Sel. Areas Commun.*, vol. 25, no. 3, 2007, pp. 589–600.

[8] L. Lee and E. Hossain, "MAC Protocol for Opportunistic Spectrum Access in Cognitive Wireless Networks," *IEEE Wireless Commun. Netw.*, *Conf.*, Las Vegas, NV, USA, Mar. 31–Apr. 3, 2008, pp. 1426–1430.

[9] V. Brik et al., "DSAP: A protocol for Coordinated Spectrum Access," *IEEE Int. Symp. New Frontiers Dynamic Spectr. Access Netw.*, Baltimore, MD, USA, Nov. 8–11, 2005, pp. 611–614.

[10] J. Marinho and E. Monteiro, "Cognitive Radio: Survey on Communication Protocols, Spectrum Decision Issues, and Future Research Directions," *Wireless Netw.*, vol. 18, no. 2, Feb. 2011, pp. 147–164.

[11] R. Urgaonkar and M.J. Neely, "Opportunistic Scheduling with

Reliability Guarantees in Cognitive Radio Networks," *IEEE Trans. Mobile Comput*., vol. 8, no. 6, Feb. 2009, pp. 766–777.

[12] S.M. Kannappa and M. Saquib, "Performance Analysis of a Cognitive Network with Dynamic Spectrum Assignment to Secondary Users," *IEEE Int. Conf. Commun.*, Cape Town, South Africa, May 23–27, 2010, pp. 1–5.

[13] S. Tang and B.L. Mark, "Performance Analysis of a Wireless Network with Opportunistic Spectrum Sharing," *IEEE GLOBECOM*, Washington, DC, USA, Nov. 26–27, 2007, pp. 4636–4640.

[14] S. Tang and B.L. Mark, "Analysis of Opportunistic Spectrum Sharing with Markovian Arrivals and Phase-Type Service," *IEEE Trans. Wireless Commun*., vol. 8, no. 6, June 2009, pp. 3142–3150.

[15] P. Wang, D. Niyato, and H. Jiang, "Voice Service Support over Cognitive Radio Networks," *IEEE Int. Conf, Commun.*, Dresden, Germany, June 14–18, 2009, pp. 164–168.

[16] S. McBeath et al., "VoIP Support Using Group Resource Allocation Based on the UMB System," *IEEE Commun. Mag*., vol. 46, no. 1, Jan. 2008, pp. 114–120.

[17] S. Haykin, "Cognitive Radio: Brain-Empowered Wireless Communications," *IEEE J. Sel. Areas Commun*., vol. 23, no. 2, Feb. 2005, pp. 201–220.

[18] W. Lee, K. Srisomboon, and A. Prayote, "Fast Spectrum Sensing with Coordinate System in Cognitive Radio Networks," *ETRI J.* vol. 37, no. 3, June 2015, pp. 491–501.

[19] X. Gelabert et al., "Spectrum Sharing in Cognitive Radio Networks with Imperfect Sensing: A Discrete-Time Markov Model," *Comput. Netw.*, vol. 54, no. 4, 2010, pp. 2519–2536.

[20] M.M. Rashid et al., "Opportunistic Spectrum Scheduling for Multiuser Cognitive Radio: A Queueing Analysis," *IEEE Trans. Wireless Commun*., vol. 10, no. 10, Oct. 2009, pp. 5259–5269.

[21] P. Wang, D. Niyato, and H. Jiang, "Voice-Service Capacity Analysis for Cognitive Radio Networks," *IEEE Trans. Veh. Technol*., vol. 59, no. 4, May 2010, pp. 1779–1790.

[22] H. Lee and D. Cho, "Capacity Improvement and Analysis of VoIP Service in Cognitive Radio System," *IEEE Trans. Veh. Technol*., vol. 59, no. 4, Dec. 2009, pp. 1646–1651.

[23] S.L. Castellanos-López et al., "Joint Connection Level and Packet Level Analysis of Cognitive Radio Networks with VoIP Traffic," *IEEE J. Sel. Areas Commun*., vol. 32, no. 3, 2014, pp. 601–614.

[24] R. Doost-Mohammady, M.Y. Naderi, and K.R. Chowdhury, "Spectrum Allocation and QoS Provisioning Framework for Cognitive Radio with Heterogeneous Service Classes," *IEEE Trans. Wireless Commun*., vol. 13, no. 7, 2014, pp. 3938–3950.

[25] A. Homayounzadeh and M. Mahdavi, "Quality of Service Provisioning for Real-Time Traffic in Cognitive Radio Networks," *IEEE Commun. Lett*., vol. 19, no. 3, Mar. 2015, pp. 467–470.

[26] J.-W. So, "Performance Analysis of VoIP Services in the IEEE 802.16e OFDMA System with Inband Signaling," *IEEE Trans. Veh. Technol*., vol. 57, no. 3, May 2008, pp. 1876–1886.

[27] H. Heffes and D. Lucantoni, "Markov Modulated Characterization of Packetized Voice and Data Traffic and Related Statistical Multiplexer Performance," *IEEE J. Sel. Areas Commun*., vol. 4, no. 6, Sept. 1986, pp. 856–868.

[28] P. Salvador, R. Valadas, and A. Pacheco, "Multiscale Fitting Procedure Using Markov Modulated Poisson Processes," *Telecommunication Syst.*, vol. 23, no. 1, June 2003, pp. 123–148.

[29] H.S. Wang and M. Moayeri, "Finite-State Markov Channel-a Useful Model for Radio Communication Channels." *IEEE Trans. Veh. Technol*., vol. 44, no. 1, Feb. 1995, 163–171.

[30] P. Sadeghi et al., "Finite-state Markov Modeling of Fading Channels- a Survey of Principles and Applications," *IEEE Signal Processing Mag*., vol. 25, no. 5, Sept. 2008, pp. 57–80.

Alireza Homayounzadeh received his BS and MS degrees in electrical engineering from Shiraz University, Iran, in 2006 and 2009, respectively. He is currently pursuing his PhD in the Department of Electrical and Computer Engineering, Isfahan University of Technology, Iran. His research interests are mainly focused on the performance analysis of telecommunication systems, cognitive radio networks, and resource allocation in wireless networks.

Mehdi Mahdavi received his BS degree in electrical engineering from Isfahan University of Technology (IUT), Iran, his MS degree in electrical engineering from Amir-Kabir University of Technology, Tehran, Iran, and his PhD in electrical engineering from the University of Sheffield, UK, From 2004 to 2007, he was a member of the ICT faculty, Ministry of ICT, Tehran, Iran. In 2007, he joined the Electrical and Computer Engineering Department of IUT, where he is currently an Associate Professor. From 2008 to 2011, he was a member of the NGN committee of the Telecommunication Company of Iran, Tehran, Iran. His research interests include capacity planning of wireless mesh networks; cognitive radio including MAC protocols, design, and analysis; QoS analysis; cooperative spectrum sensing; teletraffic theory; network performance evaluations; network calculus; and telecommunications.

http://dx.doi.org/10.4218/etrij.16.0115.0911