Capacity Analysis and Call Admission Control in Distributed Cognitive Radio Networks

SubodhaGunawardena, *Student Member, IEEE*, and Weihua Zhuang, *Fellow, IEEE*

***Abstract*—In this paper, homogeneous voice traffic in a single-channel cognitive radio network (CRN) is considered. We analyze the constant-rate voice capacity of a fully-connected network with slot-ALOHA and round-robin channel access, and propose two call admission control (CAC) algorithms for a non-fully-connected network with slot-ALOHA channel access. Different from the existing work in literature, transmission of multiple packets in a single time-slot is considered. Two discrete-time Markov chain based approaches are used for the capacity analysis of the two channel access schemes, respectively. It is shown that the number of voice packets that can be transmitted in a time-slot has a significant impact on the system capacity. The capacity analysis results of the slot-ALOHA scheme is used to develop a CAC procedure when all the voice flows have an identical statistical delay requirement. Further, two CAC algorithms (A1 and A2) are developed for a network with voice traffic flows having different delay requirements in which one (A1) is based on the theory of effective capacity and is considered as a benchmark to compare with the other. Simulation results demonstrate that algorithm A2 performs better than algorithm A1, and that a relaxed delay requirement leads to an increase in the network capacity.**

***Index Terms*—Cognitive radio network, voice capacity, quality-of-service, call admission control, slot-ALOHA, round-robin.**

I. Introduction

The extensive growth of wireless networks over the recent years has increased the demand for the radio spectrum to a great extent. The static spectrum allocation regardless of its spatiotemporal usage has resulted in scarcity of the radio spectrum. As a result, satisfying the spectrum requirements of emerging wireless applications and technologies has become a challenging task. As a promising approach to solve this problem, opportunistic spectrum access using cognitive radios has been proposed [3]-[4] with the idea of enabling devices to use any available spectrum. The concept of cognitive radio networks (CRNs) has been well accepted within the wireless communications research community, which explores portions of the spectrum not being used by licensed primary users (PUs) for the benefit of unlicensed secondary users (SUs), without causing harmful interference to PUs. The spectrum availability of a secondary network depends on the activities of the PUs, leading to some randomness in resource availability from the view point of SUs. The unpredictable nature of spectrum resource availability in the secondary networks results in that the recent research on CRNs addresses only best effort services without any strict quality of service (QoS) requirements, with the main concern on realization of CRNs [5]-[8]. However, with the ever increasing demands for wireless multimedia services, supporting delay-sensitive traffic flows with service satisfaction over the CRNs is necessary [9][10]. Call admission control (CAC) is an essential approach to QoS provisioning, and the capacity analysis is a fundamental step of developing CAC policies.

As the first step of QoS provisioning in CRNs, the capacity analysis for voice traffic is carried out in [11]-[14]. The voice capacity is defined as the maximum number of voice calls that can be supported by the network with QoS guarantees. Centralized and distributed network coordination, different channel availability statistics, and different QoS requirements have been considered. The voice over IP (VoIP) capacity of a centralized CRN is analyzed in [11], considering the dependence of the channel occupancy of PUs in adjacent time-slots. The voice calls are considered as silent suppressed (on-off) traffic flows, and the base station (BS) buffer overflow probability is considered as the QoS parameter. Wong and Foh analyze the on-off voice capacity of a CRN operating over a primary voice network, considering the average packet delay as the QoS parameter [12]. The constant-rate voice service capacity of two distributed single-channel medium access control (MAC) protocols is analyzed in [13][14]. The primary network under consideration is time-slotted and, in [13] the channel occupancy of PUs are independent in adjacent time-slots. The work is extended in [14], including both cases where the channel occupancy of PUs is dependent and independent, respectively, in neighboring time-slots. Furthermore, it studies the effect of the inconsistent channel availability information on the system capacity.

The voice capacity is analyzed in existing work under the assumption of single voice packet transmission per time slot per frequency channel. However, transmitting multiple voice packets as a composite packet over a time-slot (per channel) improves the network capacity without the requirement of a proportional increase in the channel transmission rate. Furthermore, the existing analysis is limited to the case that all the voice traffic flows require the same QoS guarantees. For example, in [13][14], all the voice traffic flows have the same delay bound $D_{\max}$ with the same stochastic delay requirement, $P(D > D_{\max}) \leq \epsilon$, where ϵ is the allowed

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The authors are with the Centre for Wireless Communications, Department of Electrical and Computer Engineering, University of Waterloo, 200 University Avenue West, Waterloo, Ontario, Canada N2L 3G1 (e-mail: shgunawa, wzhuang@bbcr.uwaterloo.ca).

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maximum delay bound violation probability. In general, the larger the delay bound, the lower the probability of delay bound violation. Therefore, a larger number of users having a larger delay bound can be accommodated in the system, which gives the possibility to provide the service at a lower cost. However, the larger the delay bound, the lower the service quality. There is a trade-off between the cost and the service quality. By incorporating different delay requirements, the system provides users a choice between service quality and cost, which improves the satisfaction of the users. In the literature, there are only limited works in developing channel access schemes to support voice [9][13] over CRNs. However, much attention has not been payed on the legacy channel access schemes in supporting voice traffic.

In this paper, we analyze the capability of legacy channel access schemes in supporting voice traffic over CRNs. The contribution of this paper is four fold: (i) We analyze the voice capacity of a single-channel distributed fully-connected CRN with slot-ALOHA channel access coordination. Different from the existing work, we consider the transmission of multiple voice packets in a single time-slot; (ii) We analyze the voice capacity of a single-channel distributed fully-connected CRN for round-robin channel access coordination. As the capacity analysis approach used for the slot-ALOHA scheme cannot be used for the round-robin scheme, a new approach is introduced; (iii) We develop a CAC procedure for a distributed non-fully-connected CRN with slot-ALOHA network coordination, assuming homogeneous single-hop voice traffic flows. The capacity analysis results of the fully-connected network is used to limit the number of calls entering the system; (iv) We develop two CAC algorithms for a distributed non-fully-connected slot-ALOHA CRN when the voice traffic flows have different delay requirements. For all the above studies, both dependent and independent channel occupancies of PUs in neighboring time-slots are considered, and the end-to-end delay of voice packets is considered as the QoS parameter. Note that (i) and (ii) are presented in [2].

II. System Model

Consider a distributed single-channel CRN in which all the SUs see the same spectrum opportunities (i.e., the network is spectrum homogeneous). The CRN is fully connected if all the SUs associated with the network are connected with each other, and non-fully-connected otherwise[1]. Each SU is equipped with a simple transceiver to sense the channel and to transmit information packets.

A. Channel availability model

The channel time is partitioned into slots of constant duration T_S. In each time-slot, the channel is either idle (i.e., no primary activities) or busy (i.e., with primary activities). In a time-slot, the state is defined as 0 if the channel is busy, and is 1 otherwise. The channel state can be independent or dependent among adjacent time-slots. In the independent case, a time-slot of the channel is (idle) available to the SUs with probability p_a, and the current channel state does not depend on its previous states. In the dependent case, the state transition of the channel among adjacent time-slots can be illustrated using a Markov chain as shown in Fig.1, where $S_{i,j}$ denotes the transition probability from state i ($\in \{0,1\}$) to state j ($\in \{0,1\}$). This is a widely used method to model the behavior of primary users [11][14][15] due to its simplicity. In each time-slot, each SU senses the channel and accurately identifies the idle state[2]. The SUs transmit only during idle time-slots, and the physical channel is assumed to be an error free channel.

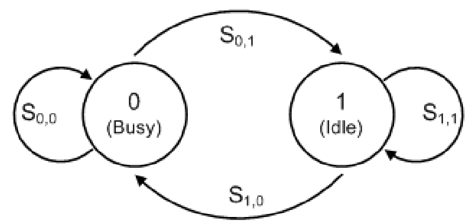

Fig. 1. The channel state transition when the channel availability is dependent in adjacent time-slots [15].

B. Traffic source characteristics

All the SUs are voice nodes, and each voice call is associated with two SUs. Each of the two SUs initiates an independent voice traffic flow to the other. For the simplicity of our analysis, we consider only one traffic flow per voice call, and each call is limited to a single-hop voice flow. In the following, the terms voice call and voice traffic flow are used interchangeably to denote a one-way single-hop packet flow of a voice call, and the term node is used to denote an SU. A voice traffic flow is a constant-rate traffic flow with a packet inter-arrival time of T_I (normalized to T_S). The service requirement for the voice traffic flows in the secondary network is characterized by the delay D (normalized to T_S), from the time that a packet is generated at the source node to the time that it is transmitted from the source node (queuing delay). The stochastic delay requirement is given by [14][16]-[18]

$$P_e \triangleq P(D > D_{\max}) \leq \epsilon \tag{1}$$

where $D_{\max}$ (normalized to T_S) and ϵ are the maximum allowable delay and delay bound violation probability, respectively, in order to provide satisfactory voice quality. If the delay bound of a voice packet is violated, the packet is dropped without being transmitted. Without loss of generality, we assume integer values for T_I, D, and $D_{\max}$.

In general, voice traffic flows can be categorized into different classes based on the packet generation process and the QoS requirements. In this work, the packet arrival processes are the same for all voice traffic flows, which may have different delay bounds. Therefore the voice traffic flows are categorized into different voice classes based on their delay bound, $D_{\max}$. The delay bound of a voice traffic flow remains constant throughout the call. The design criterion of a CAC

[1]Two SUs are connected when they are located within each other's transmission range, i.e. when the transmission of one SU can be accurately received by the other SU.

[2]There are no channel sensing errors.

algorithm is to guarantee the delay requirement (1) for all the ongoing and incoming voice calls.

C. Channel access schemes

Two legacy channel access schemes, namely the slot-ALOHA scheme and the round-robin scheme, are considered for the fully-connected network. From the view point of SUs, each time-slot consists of two sections, sensing and transmission periods. In the sensing period, all SUs sense the channel, and the transmission period is used according to the channel access scheme if the channel is idle. In the slot-ALOHA scheme, all the nodes with a non-empty buffer will transmit with a probability ϱ during an idle time-slot. If a collision occurs, each node will retransmit at the next available time-slot with the same probability. In the round-robin scheme, each node will wait for its channel access right. When a particular node receives the channel access right, it transmits if it has packets in the buffer, or forwards the opportunity to the next node otherwise. Due to the cyclic nature of getting the channel access right, each node accesses the channel in a fair manner. As an approach of realizing the round-robin channel access coordination, a token based scheme [19] or a mini-slot based scheme [14] can be used. There are no packet collisions in the round-robin scheme as a node transmits only when it has the channel access right.

When an SU decides to transmit in a time-slot, it transmits n_a $(= \min\{n_b, n_{pk}\})$ voice packets, where n_b is the number of voice packets in the buffer at the time of transmission, and n_{pk} is the maximum number of voice packets that can be transmitted by an SU during a time-slot. All n_a voice packets are encapsulated into one composite packet for transmission. The maximum number of packets is given by $n_{pk}=\lfloor \frac{(T_S-T_O-T_{sen})R_S-L_H}{L_{pk}} \rfloor$, where R_S, T_{sen}, L_H, L_{pk}, and T_O are the channel rate, sensing duration, header length of the composite packet, length of a voice packet, and the overhead duration in implementing the channel access coordination, respectively. Implementing the slot-ALOHA channel access coordination does not have any significant overhead duration ($T_O \approx 0$). However, the overhead in the round-robin scheme may not be negligible. For example, the overhead of the token-based round-robin scheme in [19] is the token passing duration, and that of the mini-slot based approach in [14] is the mini-slot durations.

In a non-fully-connected network, each node may interact with a unique set of neighbor nodes. Therefore, it is important to define the neighborhood of each node. Consider a particular voice traffic flow and denote its source and receiver nodes as the target source (ω_s) and receiver (ω_r) nodes, respectively. Consider a scenario when a non-target source node (i_s) and the ω_s transmit simultaneously in a time-slot. If the reception at ω_r fails due to the interference from i_s, i_s is denoted as a neighboring source node of ω_r, and ω_r is denoted as a neighboring receiver node of i_s. All the neighboring source (receiver) nodes associated with the target receiver (source) node constitute the neighborhood of the target receiver (source) node. The set of neighboring source (receiver) nodes of a target receiver (source) node is denoted by G_{ω_r} (G_{ω_s}), and the number of neighboring source nodes of i_r is denoted by N_{i_r}. In the non-fully-connected network, each node may have a unique neighborhood, and each node may belong to multiple neighborhoods. With the round-robin channel access, the neighborhood of ω_r should follow a particular transmission sequence such that ω_s gets the channel access right once in each round. Since a non-target neighboring source node, i_s, associated with ω_r may be associated with a few other neighborhoods, i_s should follow the transmission sequences of all the associated neighborhoods. Ultimately, i_s should be provided with the channel access right by all associated transmission sequences simultaneously. It is quite complex to achieve due to the non-connected nature of the nodes. However, in the slot-ALOHA channel access scheme, each node makes its transmission decision independently. Accomplishing slot-ALOHA channel access coordination in a non-fully-connected networking environment is comparatively simple. Therefore, we consider slot-ALOHA channel access coordination in the non-fully-connected CRN.

III. Capacity analysis

The voice capacity is defined as the maximum number of simultaneous voice traffic flows, $N_{\max}$, that can be supported by the system without violating the delay requirement (1) for all the flows. The voice capacity directly depends on the delay requirement parameters $D_{\max}$ and ϵ. The delay experienced by a voice packet depends on the number of voice traffic flows, N, channel availability statistics, and the channel access scheme. The limited spectrum opportunities in the network are shared among the voice traffic flows. The larger the N, the lower the amount of spectrum resources accessible by a voice source. Therefore, increasing N will increase the probability of delay bound violation. Spectrum availability for the SUs depends on the channel occupancy statistics of the PUs (p_a and $S_{i,j}$). The larger the value of p_a or $S_{0,1}$, the larger the amount of spectrum opportunities for the packet transmission of SUs. On the other hand, the larger the value of $1 - p_a$ or $S_{1,0}$, the lower the chances of a packet being served. The capability of an SU to exploit available spectrum opportunities is based on the channel access scheme. The higher the efficiency of the scheme, the higher the chances of packets to be transmitted and the lower the probability of delay bound violation. For the fully-connected network, we analyze $N_{\max}$ of the same class for slot-ALOHA and round-robin schemes, considering the dependent and independent channel availability statistics.

A. Slot-ALOHA scheme

With the initiation of a voice traffic flow, the first packet enters the source buffer becomes the queue-head, and the rest of the packets are buffered behind the queue-head. Whenever the queue-head is successfully transmitted, the next packet, χ_{new}, with the highest waiting time becomes the new queue-head. While awaiting for transmission, the waiting time of the queue-head increases with time. However, when a successful transmission occurs, the waiting time of χ_{new} is always lower than that of the queue-head, χ_{old}, which is just being transmitted. The waiting time of the new queue-head, D_{new} (normalized to T_S), is given by

$$D_{new} = D_{old} - n_a \cdot T_I + 1 \tag{2}$$

where D_{old} (normalized to T_S) is the waiting time of χ_{old}. The term $n_a{\cdot}T_I$ is due to the n_a inter-arrival times between the arrivals of χ_{old} and χ_{new}, and the constant 1 accounts for the time-slot taken for the transmission of χ_{old}. As the voice packets whose waiting time exceeds the delay bound are dropped, the waiting time of a queue-head stays between 0 and $D_{\max}$. When a packet (queue-head) is dropped due to violation of the delay bound (i.e., $D > D_{\max}$), the waiting time of χ_{new} is given by $D_{new}{=}(D_{\max}+1)-T_I$. The queue-head is dropped at the beginning of the time-slot when $D_{old} = D_{\max}+1$. The term T_I is due to the inter-arrival time between the χ_{old} and the χ_{new}. In each idle time-slot, a target node with a non-empty buffer transmits with probability ϱ, and a successful transmission occurs if all the other non-target nodes in the network do not transmit. The probability of successful transmission, $P_{S,1}$, in an available time-slot is given by

$$P_{S,1} = \varrho\,(1-\rho{\cdot}\varrho)^{N-1} \tag{3}$$

where ρ is the probability of a node having a non-empty buffer. The product $\varrho{\cdot}\rho$ is the probability of a node transmitting in an idle time-slot. Note that, the probability $P_{S,1}$ does not depend on D. The value of D at the next time-slot depends on the value of D, the state of the channel, and the success or failure of the transmission in the current time-slot. Furthermore, the state of the channel in the next time-slot either does not depend on that of the current time-slot for the independent channel availability scenario, or only depends on the state of the channel in the current time-slot for the two-state channel in Fig.1. Therefore, we can establish a discrete-time Markov chain (DTMC) in which the state (i, j) represents the waiting time of the queue-head and the channel state, respectively, as shown in Fig.2. Since there is no queue-head when the buffer is empty, the negative value of the time remaining until the next packet arrival is considered as the queue-head waiting time. Therefore, D varies from $-(T_I-1)$ to $D_{\max}$. Theoretical aspects of this approach is discussed in [20]. Furthermore, the DTMC model is similar to the approach given in [14], in analyzing the constant-rate voice capacity of two different cognitive radio MAC protocols. Different from [14], here we consider the transmission of possible multiple (up to n_{pk}) voice packets by a node in a time-slot. The state transition probabilities of the Markov chain are given by

$$\begin{aligned}
&P_{(k,i),(k+1,j)} = S_{i,j}, && k \in \{-T_I+1,...,-1\}\\
&P_{(k,i),(k+1,j)} = (1-P_{S,i}){\cdot}S_{i,j}, && k \in \{0,...,D_{\max}-1\}\\
&P_{(k,i),(k-T_I+1,j)} = (1-P_{S,i}){\cdot}S_{i,j}, && k = D_{\max}\\
&P_{(k,i),((k \bmod T_I)-T_I+1,j)} = P_{S,i}{\cdot}S_{i,j}, && k \in \{0,...,(n_{pk}{\cdot}T_I-1)\}\\
&P_{(k,i),(k-n_{pk}{\cdot}T_I+1,j)} = P_{S,i}{\cdot}S_{i,j}, && k \in \{n_{pk}{\cdot}T_I,...,D_{\max}\}
\end{aligned}$$

where $P_{(k,i),(l,j)}$ denotes the transition probability from state (k, i) to state (l, j) and $i, j \in \{0, 1\}$. Since the channel is not available for the SUs when it is at state 0, $P_{S,0}{=}0$. As the packets whose waiting time is larger than the delay bound are dropped, the delay bound violation probability, P_e, is equal to the packet dropping probability, given by

$$P_e = \frac{\sum_{j=0}^{1}\left(1-P_{S,j}\right){\cdot}\pi(D_{\max}, j)}{P_{S,1}{\cdot}\sum_{i=0}^{D_{\max}} n_a(i){\cdot}\pi(i,1) + \sum_{j=0}^{1}\left(1-P_{S,j}\right){\cdot}\pi(D_{\max}, j)} \tag{4}$$

where $\pi(i, j)$ is the steady state probability of state (i, j) and $n_a(i)$ is the number of packets that can be transmitted when the queue-head waiting time is i, given by

$$n_a(i) = \begin{cases} \lfloor \frac{i}{T_I} \rfloor + 1, & (\lfloor \frac{i}{T_I} \rfloor + 1) < n_{pk} \\ n_{pk}, & \text{otherwise.} \end{cases}$$

The summation $\sum_{j=0}^{1}\left(1-P_{S,j}\right){\cdot}\pi(D_{\max}, j)$ represents the mean number of dropped packets and $P_{S,1}{\cdot}\sum_{i=0}^{D_{\max}} n_a(i){\cdot}\pi(i,1)$ represents the mean number of transmitted packets at the steady state, in a time slot. The capacity analysis problem can be represented as to maximize N with the constraint $P_e \le \epsilon$. However, the relationship between the probability P_e and N is not straightforward. Therefore, we resort to numerical analysis in calculating the capacity.

We can find the probability $P_{S,1}$ for a given ρ and N by (3). Using $P_{S,1}$, the steady state probabilities of the Markov chain can be computed, and thereby the probability of buffer occupancy ρ is given by $\rho{=}\sum_{j=0}^{1}\sum_{i=0}^{D_{\max}} \pi(i, j)$. Since probabilities $\pi(i, j)$ ($i \in \{0, 1, ..., D_{\max}\}$ and $j \in \{0, 1\}$) can be represented in terms of ρ, the right hand side (RHS) of the equation also contains ρ. Denote the ρ in RHS as ρ_R and that in the left hand side (LHS) as ρ_L. The value of ρ_L can be computed for different values of ρ_R, and the solution for the equation is the one when $\rho_L{=}\rho_R$. Then, the probability of delay bound violation P_e can be obtained for a given N. Therefore, the maximum N which satisfies $P_e \le \epsilon$ can be evaluated.

When the channel occupancy of PUs in adjacent time-slots is independent, the probability of the channel state being 0 or 1 in the next time-slot is independent of the channel state of the current time-slot. Therefore, $S_{0,1} = S_{1,1} = p_a$ and $S_{1,0} = S_{0,0} = 1 - p_a$, and the capacity analysis for the independent channel occupancy scenario can be carried out using the preceding method by substituting appropriate values for $S_{i,j}$ ($i, j \in \{0, 1\}$).

B. Round-robin scheme

The round-robin scheme guarantees that each node gets a packet transmission opportunity in an orderly manner. Whenever the node under consideration (target node) transmits, its next packet transmission does not occur before each non-target node with a non-empty buffer gets an opportunity to transmit. From (2), it can be seen that the queue-head waiting time of the target node drops just after a successful transmission. The probability of the target node getting the next transmission opportunity depends on the number of non-target nodes in the network having packets to transmit, the channel availability, and the time elapsed from its previous transmission. Therefore, with the round-robin scheme, the probability of a node getting a packet transmission opportunity is not the same for all D values, and the analysis for the probability of getting a transmission opportunity at the particular D value is not straightforward. Therefore, the Markov chain approach used for the slot-ALOHA scheme cannot be applied for the capacity analysis of the round-robin scheme.

Assuming that the packets of a target node are not dropped until it gets a channel access right (i.e., the packets with the waiting time larger than $D_{\max}$ will be dropped at the time the target node gets the channel access right), the range of D is $[0, \infty)$. When the target node gets a channel access right, it will drop $n_d(D)$ and transmit $n_a(D)$ voice packets, where

$$n_d(D) = \begin{cases} 0, & D \le D_{\max} \\ \lfloor \frac{D-D_{\max}-1}{T_I} \rfloor + 1, & \text{otherwise} \end{cases}$$

and

$$n_a(D) = \begin{cases} \lfloor \frac{D-n_d(D)T_I}{T_I} \rfloor + 1, & \lfloor \frac{D-n_d(D)T_I}{T_I} \rfloor + 1 \le n_{pk} \\ n_{pk}, & \text{otherwise.} \end{cases}$$

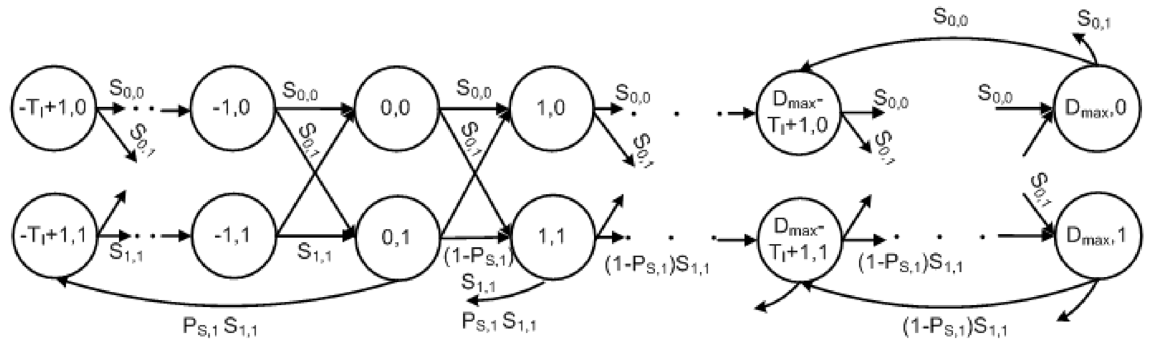

Fig. 2. The Markov chain for the queue-head waiting time and the channel state.

After transmitting the $n_a(D)$ packets, the D of the queue-head decreases by $(T_I{\cdot}n_a(D)-1)$ time-slots. Then, it increases by a random number of time-slots until the next channel access. With N voice calls in the system, for a target node, the waiting time of the queue-head at the time of packet transmission depends on the waiting time of the queue-head at the previous packet transmission and the number of time-slots required to provide a transmission opportunity to each of the $N-1$ non-target nodes. If the number of time-slots in the shortest possible round-robin cycle is larger than or equal to the number of time-slots between two successive packet arrivals, the target source buffer will always be non-empty when it receives a transmission opportunity. As the shortest possible round-robin cycle is equal to the number of nodes in the network, N, the condition to have a non-empty buffer when a source node receives a transmission opportunity can be expressed as $N \geq T_I$. Therefore, the randomness will only be due to the channel availability, not due to the number of nodes with a non-empty buffer.

As the waiting time D at the next packet transmission depends only on that of the current packet transmission, but not on the previous packet transmissions, a DTMC can be developed with the state representing the queue-head waiting time at the time of packet transmission. With the waiting time D in $[0, \infty)$, the state space of the DTMC lies in the same range, making it an infinite-state DTMC. The Markov chain is illustrated in Fig.3, where $P_{i,j}$ is the transition probability from state i to state j ($i,\ j \in \{0, 1, 2, ...\}$). For a single-channel CRN with $N{\geq}T_I$, the state transition probabilities, $P_{i,j}$, of a target node is given by $P_{i,j}{=}P\left(\sum_{z=0}^{N-1} X_z = r\right)$, if $r{\geq}N$, and 0, otherwise, where Z is the number of nodes to access the channel before the target node gets the channel access right, X_Z is the number of time-slots required to reduce the node number from Z to $Z-1$[3], and $r{=}j{-}(i-(n_d(i)+n_a(i))\,T_I)$ is the elapsed number of time-slots between adjacent channel access opportunities. The number of time-slots X_Z ($Z \in \{0, 1, ..N-1\}$) are independent and identically distributed. When the channel availability for SUs in adjacent time-slots is independent, the state (the number Z) transition for a node is illustrated in Fig.4. When there are N source nodes in the system and they all have packets to transmit, it is impossible for a target node to have its next transmission opportunity within $N-1$ adjacent time-slots from its current transmission. Therefore, $P_{i,j} = 0$ for $r < N$. In order to have $r-1$ time-slots ($r \geq N$) between two successive transmission opportunities, the target node should transmit at the r^{th} time-slot, and the rest of the $N-1$ non-target nodes should transmit during the first $r-1$ time-slots. In other words, exactly N out of the r time-slots should be idle and, out of the N idle time-slots, $N-1$ should be in the first $r-1$ time-slots. Therefore, the probability $P_{i,j}$ is given by the negative binomial distribution. The state transition probability, $P_{i,j}$, for an independent channel occupancy scenario of PUs is given by

$$P_{i,j} = \begin{cases} \binom{r-1}{r-N} p_a^N\,(1-p_a)^{r-N}\,, & \text{if } \ r{\geq}N \\ 0, & \text{otherwise.} \end{cases} \tag{5}$$

When the channel availability for SUs are dependent among adjacent time-slots, the state Z is divided into two states named Z_1 and Z_2, where a node enters state Z through state Z_1 (initial state), and enters state Z_2 if the channel is not available when it is in state Z_1. The state transition diagram of a node is illustrated in Fig.5. As explained earlier, $P_{i,j} = 0$ for $r < N$. If there are exactly N time-slots in between successive transmissions of the target node, all N time-slots should be available for the SUs[4]. Having $r > N$ time-slots between successive transmissions means that the channel has been idle for N time-slots and busy for $r-N$ time-slots. The state transition probabilities, $P_{i,j}$, for a dependent channel occupancy scenario of PUs is given by

$$P_{i,j} = \begin{cases} \sum_{l=1}^{\min(r-N,N)} \binom{N}{N-l} S_{1,1}^{N-l} S_{1,0}^{l} {\cdot} \binom{r-N-1}{l-1} S_{0,1}^{l} S_{0,0}^{r-N-l}, & \text{if } \ r > N \\ S_{1,1}^{N}, & \text{if } \ r = N \\ 0, & \text{otherwise.} \end{cases} \tag{6}$$

In (6), when $r > N$, there must be at least one transition from state 1 to state 0. The term $\binom{N}{N-l} S_{1,1}^{N-l} S_{1,0}^{l}$ represents the probability of having l state 1 to state 0 transitions out of all the transitions occur in the N idle time-slots. In order to have N idle time-slots, l state 0 to state 1 transitions are required in the remaining $r{-}N$ time-slots. The term $\binom{r-N-1}{l-1} S_{0,1}^{l} S_{0,0}^{r-N-l}$ represents the probability of having l state 0 to state 1 transitions in exactly $r-N$ time-slots. Since the DTMC has a countably infinite number of states, it is truncated to $D_{\max} + k{\cdot}T_I$ states for simplicity of analysis, where k (≥ 1) is a small integer. The delay bound violation probability, P_e, is approximately given by

$$P_e \simeq \frac{\sum_{i=0}^{D_{\max}+k\cdot T_I} n_d(i){\cdot}\pi(i)}{\sum_{i=0}^{D_{\max}+k\cdot T_I} (n_d(i)+n_a(i))\,\pi(i)} \tag{7}$$

where $\pi(i)$ is the steady state probability of state i. The terms $\sum_{i=0}^{D_{\max}+k\cdot T_I} n_d(i){\cdot}\pi(i)$ and $\sum_{i=0}^{D_{\max}+k\cdot T_I} n_a(i){\cdot}\pi(i)$ represent the mean number of dropped packets and transmitted packets, respectively, at the steady state in a time-slot. The system

[3]A non-target node with channel access right requires X_Z time-slots to obtain a channel opportunity and transmit its packets.

[4]Being in state 1 (idle state), the channel should remain in state 1 for N successive time-slots.

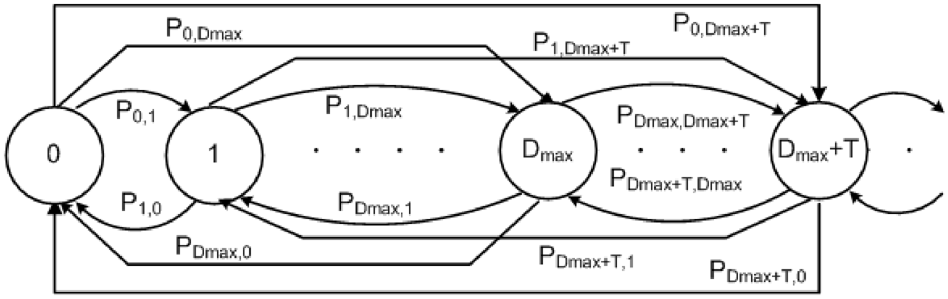

Fig. 3. The DTMC for the queue-head delay at the time of packet transmission with round-robin channel access.

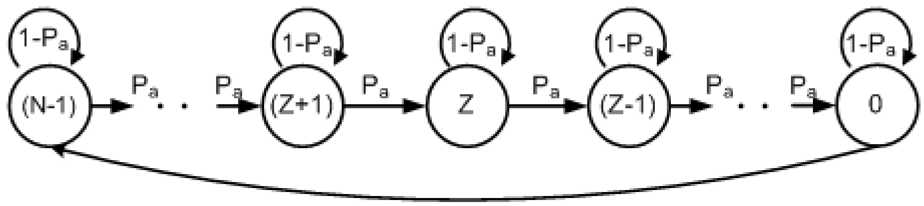

Fig. 4. State transition of Z for independent channel occupancy of PUs.

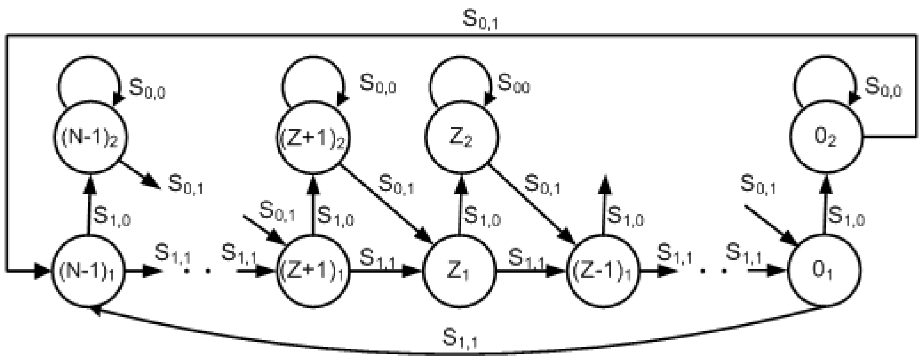

Fig. 5. State transition of Z for dependent channel occupancy of PUs.

capacity $N_{\max}$ is the maximum N which satisfies the relation $P_e \leq \epsilon$. The higher the N, the higher the P_e. The minimum P_e, P_e^*, that can be analyzed by (7) is for the minimum N, N^*. As $N \geq T_I$, $N^* = T_I$. Thus, the capacity can be evaluated for a ϵ value larger than P_e^*.

Capacity analysis of a fully-connected network is the first step of developing a call admission control algorithm. As we evaluate the maximum number $N_{\max}$ of simultaneous voice traffic flows that can be supported by the system without violating the delay requirement, the call admission control can be carried out by limiting the number of traffic flows in the network to $N_{\max}$.

IV. Call Admission Control

When the slot-ALOHA scheme is used for the channel access control, collisions occur due to simultaneous transmissions of a target source node and the neighboring source nodes associated with the target receiver node. The larger the number of neighboring source nodes associated with a target receiver, the higher the chances of collisions, which leads to a lower successful transmission probability, $P_{S,1}$, of the target source node (or traffic flow). The lower the probability $P_{S,1}$, the higher the waiting time of packets in the buffer and the probability P_e of delay bound violation. Therefore, in order to keep the probability P_e within a desired limit, the number of calls admitted to the system should be controlled.

A. CAC procedure for homogeneous voice traffic

In Subsection III-A, we analyze the maximum number, $N_{\max}$, of homogeneous voice traffic flows that can be carried out by a slot-ALOHA fully-connected network. Therefore, $N_{\max}$ is the maximum number of homogeneous voice source nodes that can be associated with a target receiver node. In a non-fully-connected network, each receiver node is associated with a number of source nodes. The packet transmission of a new source node increases the collisions at its associated receiver nodes, leading to a reduction in the successful transmission probability of the said receiver nodes. Therefore, to satisfy the delay requirement of the ongoing and incoming traffic flows, it is required to control the admission of new calls based on the number of source nodes associated with each receiver node (including that of the incoming call). A CAC procedure, $P1$, based on the number of neighboring nodes can be explained as follows. Denote the source and receiver nodes of the new call by target source (ω_s) and receiver (ω_r) nodes, respectively, and let N_{i_r} be the number of neighboring source nodes of receiver node i_r ($\in G_{\omega_s} \cup \omega_r$). It is required to limit N_{i_r} of each receiver node i_r ($\in G_{\omega_s} \cup \omega_r$) to a maximum of $N_{\max}$. Therefore, ω_s should listen to its neighbors i_r ($\in G_{\omega_s}$) and get the information N_{i_r}. At the same time, ω_r should listen to its neighbors and find N_{ω_r}. If the condition $N_{i_r} \leq N_{\max}$ can be satisfied for all i_r ($\in G_{\omega_s} \cup \omega_r$), the new call is admitted to the system, and rejected otherwise. As $N_{\max}$ is a function of ϱ, the non-fully-connected network must use the same (ϱ, $N_{\max}$) pair which used with the fully-connected network.

The capacity of a fully-connected network is under the assumption of homogeneous voice traffic. Therefore, the capacity analysis of the fully-connected network is no longer valid for non-homogeneous voice traffic. The validity of the $N_{\max}$ used in this procedure no longer holds, and a new approach is required for the CAC of non-homogeneous voice traffic over non-fully-connected CRNs.

B. CAC procedure for non-homogeneous voice traffic

Majority of the existing CAC strategies developed for non-cognitive ad hoc networks consider only the first order statistics such as average waiting time, and are based on standard queuing analysis by using the Littles theorem. However, there are some existing works on CAC in non-cognitive networks based on stochastic QoS guarantees using the theory of effective bandwidth and its dual effective capacity [17][18][21]. All of these works are for homogeneous/non-homogeneous traffic flows with the same delay requirement. Based on this idea, we can develop a CAC algorithm as a bench mark. However, analysis of the effective capacity of the service process of an SU is not straightforward as it depends on the channel access scheme. The approach used in [1] to analyze the effective capacity of the CRN can be adopted to analyze that of the service process of each node.

The effective capacity $\alpha(\delta)$ is a link layer channel model which characterizes the maximum constant traffic arrival rate that a wireless system can support as a function of its service requirement, under the assumption of first-in first-out (FIFO) service discipline [16]. The packet buffer of each source node act in the FIFO service discipline. Therefore, in order to satisfy the delay requirement of voice packets, the effective capacity of the service process of each source node should be larger than the constant arrival rate. A successful packet transmission from a target source node occurs whenever there are no collisions at the target receiver node. Therefore, the service process of the target source node is governed by the transmissions of the neighboring source nodes of the target receiver node. The effective capacities $\alpha_i(\delta)$ and $\alpha_d(\delta)$ of the discrete-time service process for independent and dependent channel availability scenarios can be obtained by (9) and (10) in [1], respectively. In a particular time-slot, define the state of a target source node as follows: If a successful transmission occurs during the time slot, the source node is in state 1, and state 0 otherwise. The effective capacities are given by

$$\alpha_i(\delta) = \frac{1}{\delta}\log\left[(1 - P_{S,1}\cdot p_a) + P_{S,1}\cdot p_a\cdot e^{-\delta\cdot n_{pk}}\right]$$

and (8)

$$\alpha_d(\delta) = \frac{1}{\delta}\log\left[sp(W\cdot\Lambda)\right]$$

where W is the 2×2 state transition probability matrix of a source node, Λ=diag$[1, e^{-\delta\cdot n_{pk}}]$, and $sp(W\cdot\Lambda)$ is the spectral radius of the matrix $W\cdot\Lambda$. Consider a Markov chain in which the state represents the channel state and node state pair which consist of three states (1,1), (1,0), and (0,1). Denote the state transition probability matrix of the Markov chain by $\acute{W}$. The state transition probability matrix W of a source node can be obtained using the state transition probability matrix $\acute{W}$. The condition to satisfy the delay requirement of a voice call i is given by $\delta^*\alpha(\delta^*) \geq \frac{1}{D_{\max}}\log\left(\frac{1}{\epsilon}\right)$, where δ^* is the solution to the equation $\alpha(\delta)$=$\frac{1}{T_I}$. This condition can be given in the form $\alpha\left(\delta^*_{\min}\right) \geq \frac{1}{T_I}$, where $\delta^*_{\min}$=$\frac{T_I}{D_{\max}}\log\left(\frac{1}{\epsilon}\right)$. In the distributed non-fully-connected network scenario, the probability $P_{S,1}$ of a target source node ω_s is given by

$$P_{S,1} = \Pi_{i_s\in G_{\omega_r}}\varrho_\omega(1-\rho_i\cdot\varrho_i) \quad (9)$$

where ϱ_j and ρ_j are the transmission probability given that the buffer is non-empty and the probability of having a non-empty buffer of source j_s ($\in \{G_{\omega_r} \cup \omega_s\}$), and ω_r is the target receiver node. However, the evaluation of ρ_i is not straightforward as it depends on the transmissions of the neighboring source nodes of receiver i_r. Therefore, rather than evaluating the exact value of ρ_i, we investigate the possibility of obtaining a close upper bound for the value of ρ_i. From the DTMC illustrated in Fig.2, it can be shown that the delay bound violation probability P_e of a constant-rate voice traffic flow and the probability ρ of a voice buffer being non-empty, monotonically decrease with the successful transmission probability $P_{S,1}$. Therefore, the delay requirement $P_e \leq \epsilon$ can be transformed to $P_{S,1} \geq P^*_S$ or $\rho \leq \rho^*$, where P^*_S is the $P_{S,1}$ value at $P_e = \epsilon$ and ρ^* is the ρ value at $P_{S,1} = P^*_S$. The variation of ρ and P_e with $P_{S,1}$, and the relationship of P^*_S, ρ^*, and ϵ are illustrated in Fig.6. As

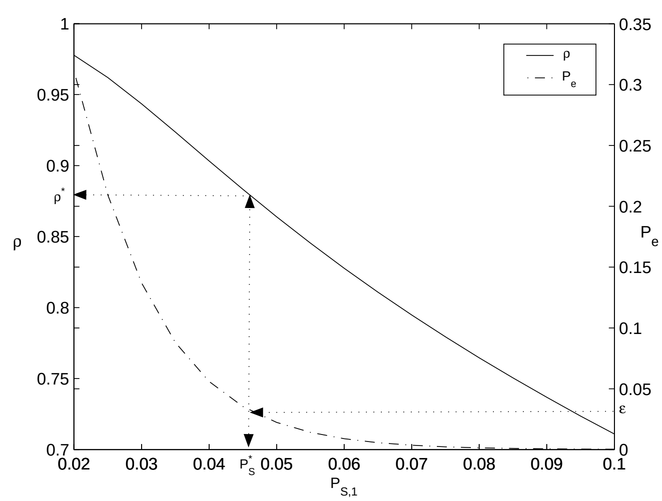

Fig. 6. Variation of ρ and P_e with $P_{S,1}$ for $D_{\max}$=100, T_I=10, n_{pk}=5, P_{01}=0.8, P_{10}=0.2, and ϵ=0.01.

long as the existing source nodes satisfy the delay requirement $P_e \leq \epsilon$, the probability ρ is upper bounded by ρ^*. Therefore, instead of using $P_{S,1}$, we substitute $P^*_{S,1}$=$\Pi_{i_s\in G_{\omega_r}}\varrho_\omega(1-\rho^*_i\cdot\varrho_i)$ ($\leq P_{S,1}$) in (8). When the system supports non-homogeneous voice traffic flows with different delay bounds, let C denote the set of all voice traffic classes in the network. Each voice traffic class c ($\in C$) has unique delay bound $D_{\max}(c)$, $P_S^*(c)$, and $\rho^*(c)$ values. Therefore, ρ_i^* and ϱ_i of $P_{S,1}^*$ should be replaced by their respective values of the traffic class c_i as $\rho^*(c_i)$ and $\varrho(c_i)$, where $\varrho(c_i)$ is the default ϱ value for the traffic class c_i. Denote the source and receiver nodes of the incoming call, ω, as the target source (ω_s) and receiver (ω_r) nodes. In order to make sure that the delay requirements of all ongoing calls and the new call are satisfied, effective capacities of each receiver node i_r ($\in \{G_{\omega_s} \cup \omega_r\}$) should be larger than the packet arrival rate $\frac{1}{T_I}$. The benchmark CAC algorithm based on the effective capacity is given in algorithm A1. Each receiver node in

Algorithm 1: CAC algorithm based on the effective capacity

Data : C_i=$\{c_j : j \in G_{i_r}\}$
Result: $\hat{C}_i$

$\hat{C}_i \leftarrow \emptyset$;
repeat
  randomly select c_k from C;
  $C \leftarrow C$-$\{c_k\}$;
  if i==ω **then**
    $P_{S,1} \leftarrow \varrho(c_k)\Pi_{j\in G_{\omega_r}}(1-\rho^*(c_j)\cdot\varrho(c_j))$;
  else
    $P_{S,1} \leftarrow$ $\varrho(c_i)\Pi_{j\in G_{i_r}}(1-\rho^*(c_j)\cdot\varrho(c_j))(1-\rho^*(c_k)\cdot\varrho(c_k))$;
  end
  if *channel availability is iid in time* **then**
    $\alpha(\delta) \leftarrow \alpha_i(\delta)$;
  else
    $\alpha(\delta) \leftarrow \alpha_d(\delta)$;
  end
  $\delta^*_{\max} \leftarrow \frac{T_I}{D_{\max}(c_i)}\log\left(\frac{1}{\epsilon}\right)$;
  if $\alpha(\delta^*_{\max}) \geq \frac{1}{T_I}$ **then**
    $\hat{C}_i \leftarrow \{\hat{C}_i \cup c_k\}$;
  end
until C==$\emptyset$;
Exit;

the network should run the algorithm and identify the set $\hat{C}_j$ ($j \in \{G_{\omega_s} \cup \omega_r\}$). The set $\hat{C}_\omega$ and $\hat{C}_i$ ($i \in G_{\omega_s}$) are the set of voice classes that can be admitted by ω_r, and to the neighborhood of an existing receiver node i_r, respectively, without violating the delay requirement of the existing and incoming voice calls. The new source and receiver nodes listen to the channel and identify the set of voice classes C_ω=$\bigcap_{i_r\in\{G_{\omega_s}\cup\omega_r\}}\hat{C}_i$ that can admit call ω. If C_ω=$\emptyset$, call ω cannot be admitted to the system. The effective bandwidth/capacity approach can be applied to different types of traffic by evaluating the effective bandwidth [16] of the source traffic and the effective capacity of the service process via modeling the source buffer occupancy at the packet level. However, this approach is computationally complex due to the requirement of calculating the effective capacity at run-time. It is possible to introduce a less complex approach for the CAC for non-homogeneous voice traffic using the relationship of $P_{S,1}$, ρ, and P_e.

Based on Fig.6, guaranteeing $P_{S,1} \geq P_S^*(c)$ guarantees $\rho \leq \rho^*(c)$. Therefore, if the probability $P_{S,1}$ of source i_s ($\in G_{\omega_r}$) satisfies $P_{S,1} \geq P_S^*(c_i)$, the inequality $\varrho_\omega\Pi_{i_s\in G_{\omega_r}}(1-\rho_i\cdot\varrho_i) \geq \varrho_\omega\Pi_{i_s\in G_{\omega_r}}(1-\rho^*(c_i)\cdot\varrho_i)$ always stands. Provided that $P_{S,1} \geq P_S^*(c_i)$ for all $i_s \in G_{\omega_r}$, the delay requirement of the incoming call can be guaranteed by choosing a proper ϱ_ω value for its source ω_s, which satisfies $\varrho_\omega\Pi_{i_s\in G_{\omega_r}}(1-\rho^*(c_i)\cdot\varrho_i) \geq P_S^*(c_\omega)$. However, as discussed in Subsection IV-A, the admission of a new source node increases the probability P_e of delay bound violation of each source i_s, where i_s is the corresponding source node of i_r ($\in G_{\omega_s}$). Therefore, it is required to guarantee that $P_{S,1}$ values of the said source nodes and the new source node are kept above their respective $P_S^*(c_j)$ ($j_r \in \{G_{\omega_s} \cup \omega_r\}$) values by making sure that the following conditions are met respectively

$$\varrho_j\cdot\prod_{i_s\in G_{j_r}}(1-\rho^*(c_i)\cdot\varrho_i) \geq P_S^*(c_j), \quad \forall\ j_r \in G_{\omega_s}$$

and (10)

$$\varrho_\omega\cdot\prod_{i_s\in G_{\omega_r}}(1-\rho^*(c_i)\cdot\varrho_i) \geq P_S^*(c_\omega)$$

where the LHSs of (10) are always less than or equal to $P_{S,1}$. The expressions of the LHSs of (10) can be evaluated using γ_j (=$\varrho_j\cdot\prod_{i\in G_{j_r}}(1-\rho^*(c_i)\cdot\varrho_i)$) and c_j obtained from the neighboring receiver nodes of the new source node, and γ_ω (=$\varrho_j\cdot\prod_{i\in G_{\omega_r}}(1-\rho^*(c_i)\cdot\varrho_i)$) obtained from the new receiver node. The CAC algorithm based on the relationship among P_e, $P_{S,1}$, and ρ is given in Algorithm 2. In the algorithm,

Algorithm 2: CAC algorithm based on the successful transmission probability

Data : Γ=$\{(\gamma_i, c_i) : i \in \{G_{\omega_s} \cup \omega_r\}\}$
Result: ϱ_ω, Admit the call or block the call

1 $\Lambda \leftarrow 0$;
2 $\varrho_\omega \leftarrow \varrho_s$;
3 **while** $\varrho_\omega \leq 1$ **do**
4   **if** $\gamma_\omega\cdot\varrho_\omega \geq P_S^*(c_\omega)$ **then**
5     $\varrho_{\min} \leftarrow \varrho_\omega$;
6   Go to 10;
7   **end**
8   $\varrho_\omega \leftarrow \varrho_\omega$+$\varrho_s$;
9 **end**
10 **while** $\varrho_\omega \leq 1$ **do**
11   $H_g \leftarrow G_{\omega_s}$;
12   **repeat**
13   randomly select j_r from H_g;
14     $H_g \leftarrow H_g$-$\{j_r\}$;
15   **if** $\gamma_j(1-\rho^*(c_\omega)\cdot\varrho_\omega) < P_S^*(c_j)$ **then** Go to 21;
16   **until** H_g==$\emptyset$;
17   $\Lambda \leftarrow 1$;
18   $\varrho_{\max} \leftarrow \varrho_\omega$;
19   $\varrho_\omega \leftarrow \varrho_\omega$+$\varrho_s$;
20 **end**
21 **if** Λ==1 **then**
22   $\varrho_\omega \leftarrow \varrho_{\min} + \beta(\varrho_{\max} - \varrho_{\min})$;
23   Admit the call;
24 **else**
25   Block the call;
26 **end**
27 Exit;

parameter $\varrho_{\min}$ is the minimal ϱ value which satisfies the first inequality in (10) and $\varrho_{\max}$ is the maximal ϱ value which satisfies the second inequality in (10). Algorithm 2 searches for $\varrho_{\min}$ and $\varrho_{\max}$ by increasing ϱ_ω from 0 to 1 in a step size ϱ_s. The smaller the ϱ_s, the higher the accuracy of $\varrho_{\min}$ and $\varrho_{\max}$ values. However, the smaller the ϱ_s, the larger the number of iterations required to get the results, leading to a larger processing time. If the algorithm outcome is to admit the call, it needs to choose a ϱ_ω value ($\varrho_{\min} \leq \varrho_\omega \leq \varrho_{\max}$) for the transmissions of the new source node. The probability $P_{S,1}$ of the new source node and corresponding source nodes of its neighboring receiver nodes will vary depending on the chosen ϱ_ω value. Therefore, a particular β value should be selected for the network to obtain a ϱ_ω ($=\varrho_{\min} + \beta(\varrho_{\max} - \varrho_{\min})$) value, such that the network capacity is maximized. This can be carried out by trial and error method off-line.

V. Numerical Results

Computer simulations are carried out to evaluate the accuracy of the capacity analysis of the two channel access schemes and to investigate the performance of the two CAC algorithms. In order to depict the primary user activities, the channel is made on and off according to the dependent and independent channel occupancy statistics of PUs. Two classes voice traffic are considered: Class 1 (c_1) with $D_{\max}(c_1) = 100$, $T_I = 10$ and $\epsilon = 0.01$; and Class 2 (c_2) with $D_{\max}(c_2) = 250$, $T_I = 10$ and $\epsilon = 0.01$. Note that $D_{\max}(\cdot)$ and T_I are normalized to T_S. The probability of delay bound violation, P_e, is obtained by the ratio of the number of dropped packets (at a source node due to the violation of delay bound) to the total number of packets generated by the source node.

A. Capacity of the fully-connected network

Consider homogeneous voice traffic flows of class c_1. While keeping N constant during a simulation run, the probability P_e is obtained for a particular channel access scheme and channel statistics. Starting from $N = 2$, we increase N by one for each simulation run and the resultant probability P_e is compared with ϵ to obtain $N_{\max}$ which satisfies $P_e \leq \epsilon$.

Figs. 7-8 show the variation of $N_{\max}$ with n_{pk} obtained from numerical analysis and simulation with round-robin channel access scheme and the slot-ALOHA channel access scheme, respectively. It can be seen that the $N_{\max}$ obtained from simulation match well with the analytical results. Even though the results show an exact match in the capacity, the P_e values from the simulation and numerical analysis can be different. Let $P_{e_s}(n)$ and $P_{e_a}(n)$ denote the P_e vales obtained from the simulation and the numerical analysis, respectively, having n calls in the system. When the system capacity is N_{max}, $P_{e_s}(N_{max}) \leq \epsilon$, $P_{e_s}(N_{max}+1) > \epsilon$, $P_{e_a}(N_{max}) \leq \epsilon$, and $P_{e_a}(N_{max}+1) > \epsilon$. This does not necessarily mean that $P_{e_s}(N_{max})=P_{e_a}(N_{max})$. Therefore, the exact match shown in N_{max} indicates a close match in the resultant P_e, but not necessarily an exact match in P_e. In both the round-robin and slot-ALOHA channel access schemes, the system capacity increases with n_{pk}. Furthermore, it is observed that the system capacity of the round-robin scheme is much higher than that of the slot-ALOHA scheme. Note that the overhead required for the establishment of the round-robin scheme is much higher than that of the slot-ALOHA scheme, as explained in Subsection II-C, which is neglected in the simulation. Furthermore, it is observed that the rate of increment of system capacity with n_{pk} in the slot-ALOHA scheme is lower than that of the round-robin scheme. In the slot-ALOHA scheme, the probability of transmission is irrespective of the buffer occupancy of packets. However, in the round-robin scheme, there is a higher probability to transmit when the waiting time of the queue-head is larger (i.e., when there are more number of packets in the buffer), which allows a node to transmit a larger number of voice packets during a transmission than in the slot-ALOHA scheme. Therefore, the mean number of packets transmitted during a channel access opportunity is larger in the round-robin scheme than that in the slot-ALOHA scheme, which explains that the latter has a lower rate of capacity improvement with n_{pk}. The analytical model based on the slot-ALOHA channel access can also be used for other

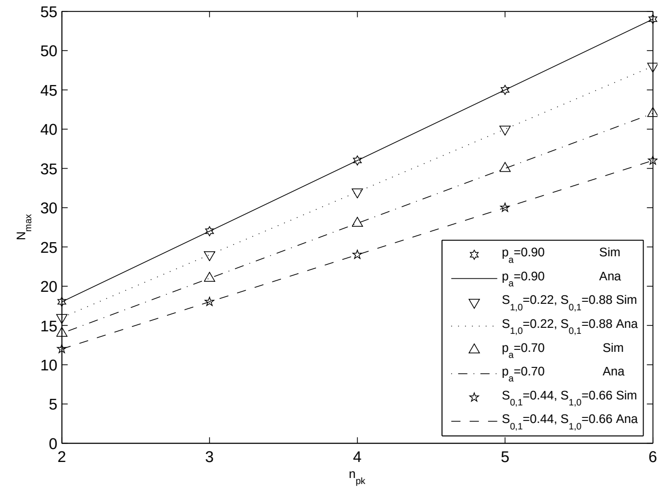

Fig. 7. Variation of $N_{\max}$ with n_{pk} for a fully-connected network with round-robin channel access.

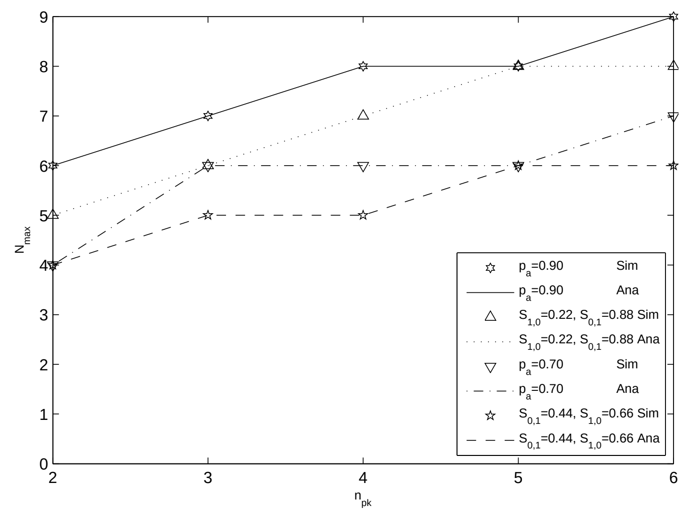

Fig. 8. Variation of $N_{\max}$ with n_{pk} for a fully-connected network with slot-ALOHA channel access.

channel access schemes in which the probability of successful transmission is independent of the queue-head waiting time (e.g. mini-slot based contention schemes). Further, this work can be extended to Markov modulated constant rate arrivals (e.g. on-off voice) given the statistics of Markov process.

B. CAC of the non-fully-connected network

For the performance comparison of the CAC procedure (P1) and two CAC algorithms (A1 and A2), we consider a CRN with homogeneous voice traffic. For the performance comparison of algorithms A1 and A2, we consider a network with both traffic classes, where new call arrivals are equally likely to be of class c_1 or c_2. The network coverage area of each voice source/receiver node is a circle with a radius of unit length. The inter-arrival time of voice calls is exponentially distributed, and the location of source nodes is uniformly distributed in a square network area. Ten different data sets are generated, each containing 8000 samples of source and receiver location and call inter arrival time. In order to compare the two algorithms, 10 different simulation runs were carried out for each algorithm using the generated data sets over a constant network area. As the network is non-fully-connected, the system capacity depends on the coverage area of the network. We saturate the network with voice calls to obtain the maximum number of voice calls that can be supported by the system, and obtained the results for different network coverage areas.

Fig.9 shows the comparison of the network capacity (with the 95% confidence interval) of class c_1 voice calls using procedure P1 and algorithms A1 and A2. The CAC procedure

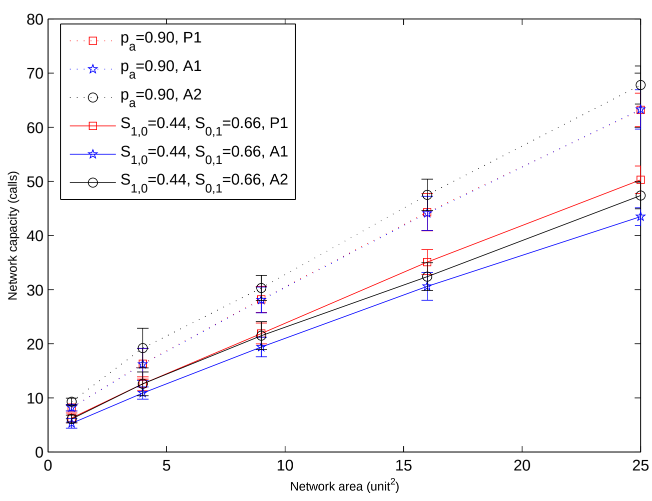

Fig. 9. Variation of the network capacity versus the network area for procedure $P1$ and algorithms A1 and A2.

P1 outperforms the algorithms A1 and A2 when the average channel availability is lower, and the algorithm A2 outperforms the other two when the average channel availability is higher. The algorithm A2 opportunistically chooses the probability ϱ at the instance of call admission whereas P1 has a fixed ϱ value. Therefore the opportunistic ϱ selection may choose different ϱ values for different calls leading to a probability $P_{S,1}$ which is just enough to satisfy the admission criterion (ϱ can vary from ϱ_{min} to ϱ_{max}). The lower the channel availability, the higher the P_S^*. The higher the P_S^*, the lower the tolerance for admitting a new call and vice versa. Therefore, lower channel availability can lead to lower capacities when the ϱ selection is opportunistic as in A2. The performance of algorithm A1 always stays below A2 due to the conservative nature of the theory of effective capacity. The required information from the neighboring nodes and the calculation complexity of P1 is less than those of A1 and A2. Therefore, the procedure P1 can be a better choice over A1 and A2 at low channel availability, and A2 can be a better choice over P1 and A1 for a network with homogeneous voice traffic at high channel availability.

Fig.10 shows the variation of the average network capacity with the network area, using algorithms A1 and A2 for the two equally likely voice traffic classes. The results show that

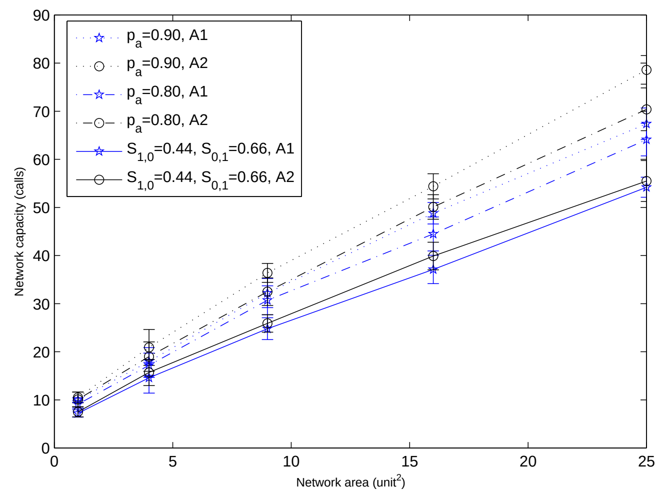

Fig. 10. Variation of the network capacity versus the network area for algorithms A1 and A2 for a network with two voice traffic classes.

algorithm A2 is a better choice over A1. The average network capacity with the mixture of two voice traffic classes is higher than that for voice class c_1. The relaxed QoS requirement of voice traffic class c_2 allows more calls to be admitted. Clearly, there is a trade-off between the number of calls in the systems and the service quality, as expected. Algorithm A2 can be extended to other contention based channel access schemes (e.g. IEEE 802.11 RTS/CTS based) and traffic types, given that P_e and ρ monotonically decrease with P_s, and the channel contention is independent over adjacent time-slots.

VI. Conclusion

In this paper, we study the constant-rate voice capacity of a synchronized single-channel fully-connected CRN using slot-ALOHA and round-robin channel access schemes, based on the delay requirement of the voice traffic. Different from the analysis given in the literature, transmission of multiple voice packets in a single time-slot is considered. A DTMC based approach is used for the capacity analysis of the slot-ALOHA scheme, and a new analytical model is introduced for the capacity analysis of round-robin scheme. Further, a procedure based on the capacity of the fully-connected networks and two algorithms, A1 and A2, based on the effective capacity and the successful transmission probability, respectively, are proposed for CAC in a non-fully-connected CRN with slot-ALOHA channel access. It is shown that the procedure P1 performs better than algorithms A1 and A2 at lower channel availabilities for homogeneous voice traffic, and the algorithm A2 outperforms A1 for both homogeneous and non-homogeneous voice traffic. Simulation results demonstrate that voice traffic flows with a relaxed delay requirement leads to an increases of the network capacity.

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Subodha Gunawardena (S'08) received the B.Sc. degree in Electronics and Telecommunications Engineering from the University of Moratuwa, Sri Lanka, and M.Eng in Telecommunications Engineering from the Asian Institute of Technology, Thailand, in 2004 and 2008, respectively. He worked at the Canon research labs in Japan form 2005 to 2006 in collaboration with Metatechno Japan Inc.. He is also a co-recipient of the Best Conference Paper Award in 2011 awarded by the IEEE Multimedia Communications Technical Committee. Subodha is currently working toward his Ph.D. degree at the Department of Electrical and Computer Engineering, University of Waterloo, Canada. His research interests include capacity and resource allocation in Cognitive Radio Networks, cooperative communication, and robotics.

Weihua Zhuang (M'93-SM'01-F'08) received the B.Sc. and M.Sc. degrees from Dalian Maritime University, China, and the Ph.D. degree from the University of New Brunswick, Canada, all in electrical engineering. Since October 1993, she has been with the Department of Electrical and Computer Engineering, University of Waterloo, Canada, where she is a Professor and a Tier I Canada Research Chair in wireless communication networks. Her current research focuses on resource allocation and QoS provisioning in wireless networks. Dr. Zhuang is a co-recipient of the Best Paper Awards from the IEEE VTC Fall-2010, IEEE WCNC 2007 and 2010, IEEE ICC 2007, and the International Conference on Heterogeneous Networking for Quality, Reliability, Security and Robustness (QShine) 2007 and 2008. She received the Outstanding Performance Award 4 times since 2005 from the University of Waterloo for outstanding achievements in teaching, research, and service, and the Premiers Research Excellence Award in 2001 from the Ontario Government for demonstrated excellence of scientific and academic contributions. Dr. Zhuang is the Editor-in-Chief of the IEEE Transactions on Vehicular Technology, and the TPC Symposia Chair of the IEEE Globecom 2011. She is a Fellow of the Canadian Academy of Engineering (CAE) and the IEEE, and an IEEE Communications Society Distinguished Lecturer.