

Chapter 2

Voice Capacity Analysis

2.1 Introduction

The voice and video streaming are the most common real-time wireless services in the modern day wireless networks [1]. The demand for video streaming is growing rapidly, and it is predicted to be the most common wireless service in the future [2]. As an upcoming networking approach, it is necessary to support the voice and video services over CRNs.

The cellular and satellite networks have dedicated capacity allocation to voice traffic flows, and dedicated channel time is available for each flow. Therefore, the delay of a voice packet is a deterministic quantity, which depends on the propagation delay through the network. However, in the networks such as wireless local area networks (WLANs), voice traffic flows are sharing the network resources with other services such as video streaming and interactive data services. Therefore, fixed capacity allocation is not always feasible. Due to the random nature of the channel access, the packet delay is not deterministic. When the voice service is operating in such a network, the service quality requirement is given in terms of a stochastic delay requirement [3], such as $P(D > D_{\max}) \leq \epsilon$, where D is the end-to-end delay of a voice packet, D_{\max} and ϵ are the delay bound (or the maximum delay) and maximum delay bound violation probability allowed, respectively. Further, the service requirement is relaxed in some studies by giving the QoS requirement in terms of mean of the packet delay. Voice traffic sources are categorized into constant-rate and on-off traffic sources [4] based on the output of the voice coder. The voice service support is extensively studied over the conventional wireless networks, and as an upcoming networking approach, it is an important research area in the context of CRNs.

In an overlay CRN, the channel availability for the secondary network exclusively depends on the behavior of the PUs. Therefore, the amount of spectrum resources available for the secondary network is limited. In order to provide the required service levels using limited available resources in CRNs, capacity analysis and call admission control are essential. The voice service support over CRNs

has been studied in the literature with respect to the capacity analysis [5–8], call admission control [9, 10], and developing channel access schemes [5, 6, 11]. The voice capacity is defined as the maximum number of voice calls that can be supported by the network satisfying their service requirements. In [7, 8, 11], the network under consideration is a centralized CRN, and the service quality is measured in terms of the packet dropping probability due to the buffer overflow at the base station [7, 8] and the end-to-end delay of voice packets [9], respectively. The QoS requirement is given in terms of the mean end-to-end voice packet delay in [9]. However, in [5, 6], a stochastic delay requirement is considered in the capacity analysis. The voice sources are treated as constant-rate traffic sources in [5, 6, 11], and on-off traffic sources in [7, 8, 10]. In [9], three different CAC strategies used for voice over IP (VoIP) traffic in wireless networks are mathematically analyzed with respect to centralized CRNs. The authors study the impact of the primary user information on the CAC algorithms in [10]. Different CAC algorithms based on the number of SUs in the system and the total number of users (both the PUs and SUs), respectively, are compared. The call blocking probability and the packet loss probability are considered as the QoS parameters. Developing channel access schemes for voice service support over distributed fully-connected CRNs and the capacities of the developed schemes are presented in [5, 6]. In almost all of the works, the primary network under consideration is time-slotted, and the channel availability for the secondary users can be either independent or dependent in adjacent time-slots, respectively. Most existing studies assume perfect channel sensing by the secondary users, except in [8] which extends the capacity analysis to incorporate the effect of imperfect channel sensing. Different from the previous studies, the voice service support over TV bands is studied in [11]. The minimum number of channels required to be sensed by each SU to support a given traffic load is analyzed using an optimization technique. The channel sensing errors are also incorporated into the optimization problem. Further, deployment of long term evolution (LTE)-Advanced in the industrial, scientific, and medical (ISM) band to resolve some deployment issues is studied in [12]. The authors propose a statistical traffic control scheme to tackle critical challenges of the packet transmission coordination and the radio resources allocation in the network.

The voice capacity studies in the existing works are carried out for single voice packet transmission per time slot per frequency channel. Transmitting multiple voice packets as a composite packet over a time-slot (per channel) improves the network capacity without the requirement of a proportional increment in the channel transmission rate. However, transmitting multiple voice packets as a composite packet can increase the delay jitter, which is not desirable in the context of voice service. There is a trade-off between reducing the packet delay and delay jitter. Furthermore, the existing works are limited to the case that all the voice traffic flows require the same service quality. For example, in [5, 6], all the voice traffic flows have a stochastic delay requirement with the same delay bound and 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 service cost and the service quality. By incorporating different delay requirement parameters (D_{\max} and ϵ), 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 [6, 7] over CRNs, and much attention has not been paid to the generic channel access schemes in supporting voice traffic.

2.1.1 Motivation and Objectives

Treating the voice source as an on-off traffic source leads to higher system capacities. Further, stochastic delay guarantees are provided in shared wireless networks to provide higher service satisfaction. However, time varying nature of the on-off voice traffic flows and the random nature of the channel availability of the CRNs make the service satisfaction challenging. Therefore, on-off voice capacity has not been studied in the literature providing a stochastic delay guarantee. In this chapter, we study the on-off and constant-rate voice capacity of a centralized CRN with a stochastic delay requirement. A base station may not be available in a CRN due to the cost of installation. Therefore, the node coordination is difficult, and interruptions by the primary users makes it more challenging. In distributed CRNs, efficient channel access schemes are required for the network coordination. The system capacity depends on the efficiency of the channel access schemes, and benchmarks are important to compare the efficiency of the developed schemes. Readily available legacy channel access schemes such as round-robin and slot-ALOHA schemes can be used as the benchmarks, and studying the voice capacity of CRNs with legacy channel schemes is useful for the CR research community. In this chapter, we study the capability of legacy channel access schemes in supporting voice traffic over fully-connected CRNs. In non-fully-connected networks, the larger the number of neighbors associated with target user, the lower the channel time available per user, leading to lower service quality. Therefore, CAC plays an important role in keeping the number of neighbors per user to an acceptable level. In this chapter, we develop CAC algorithms for non-fully-connected slot-ALOHA based CRNs in supporting voice traffic.

2.1.2 Contributions

The contribution of this chapter is five fold: (1) We analyze the on-off and constant-rate voice capacity of a single-channel centralized CRN with FCFS service discipline; (2) We analyze the constant-rate 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; (3) 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. Further, possible extensions to the analytical models to incorporate sensing errors are discussed; (4) 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; (5) 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 (1) is presented in [13] and (2)–(5) are presented in [14, 15].

2.2 System Model

The system architecture, channel model, voice traffic model, and channel access schemes under consideration are described in this section.

2.2.1 System Architecture

In this study, we consider a centralized CRN with a base station and a distributed fully-connected CRN as illustrated in Fig. 2.1a, b, respectively. The secondary network operates over a time-slotted single-channel primary network, and all the SUs see the same spectrum opportunities¹ (spectrum homogeneous). The secondary network is an overlay CRN in which the SUs access the channel (transmit or receive) only when the PUs are not present. Each SU is equipped with a single transceiver to sense the channel and transmit information packets.

2.2.2 Channel Availability Model

The channel time is partitioned into slots of constant duration T_S . The channel state of each time-slot 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,

¹The primary user activities are consistent throughout the network. Therefore, the coverage area of the secondary network should be smaller than the one hop coverage area of the primary network.

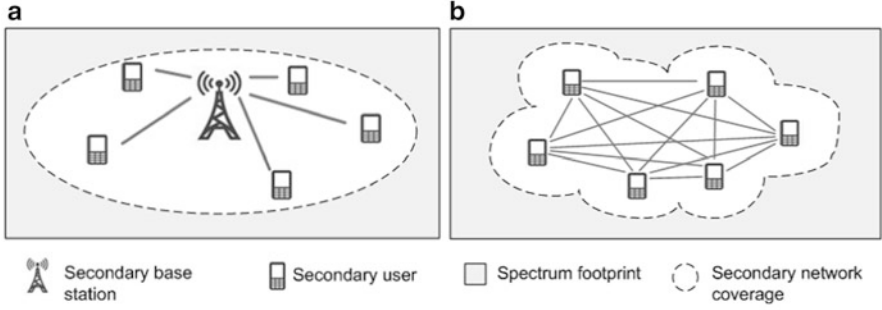
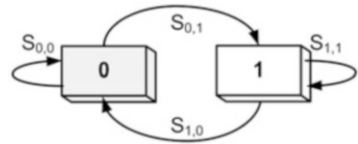


Fig. 2.1 (a) The centralized CRN with a BS. (b) The distributed fully-connected CRN

and is 1 otherwise. The state transition of the channel among adjacent time-slots can be given using a Markov chain as illustrated in Fig. 2.2, where $S_{i,j}$ denotes the transition probability from state i to state j ($i, j \in \{0, 1\}$). This is a widely used method to model the behavior of primary users [5–8, 16] due to its simplicity. The channel state can be independent or dependent among adjacent time-slots. In the independent case, $S_{0,1}=S_{1,1}=p_1$ and $S_{1,0}=S_{0,0}=p_0=1-p_1$. For both the independent and dependent cases $S_{0,1}+S_{0,0}=1$ and $S_{1,0}+S_{1,1}=1$. The channel state is identified by an SU by spectrum sensing. Different sensing techniques are listed in [17] and references there in. For the simplicity of analysis, we assume that the final sensing decisions of the SUs and the BS are free of errors.² A time-slot is mainly divided into sensing and transmission phases, and in addition, distributed networks need a contention phase before the transmission phases. An SU transmits only when the channel is at State 1 (available for SUs), and the sensing and transmissions are free of errors.

Fig. 2.2 The channel state transition diagram



2.2.3 Voice Traffic Model

All the SUs are voice nodes, and each voice call is associated with an SU and the BS or two SUs. Each of the two SUs (or an SU and the BS) initiates an independent

²In reality, sensing errors are inevitable. Under the future extensions of this work, we discuss the effects of the sensing errors on the secondary network and the primary network, respectively.

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. Widely accepted voice traffic models, namely, the constant-rate voice model [18, 19] and on-off voice model [20, 21] are considered.

2.2.3.1 Constant-Rate Voice Model

A voice node generates a constant-rate traffic flow with a packet inter-arrival time of T_I (normalized to the time-slot duration T_S), and the output of a voice codec is illustrated in Fig. 2.3.

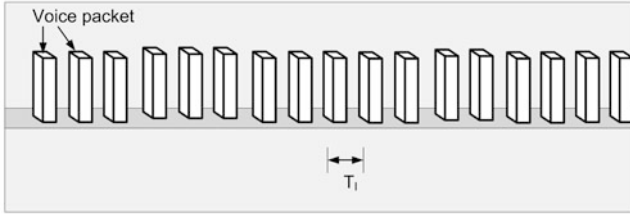


Fig. 2.3 The codec output of a constant-rate voice source

2.2.3.2 On-Off Voice Model

The on-off model is a common voice traffic model used in the VoIP applications [7–10]. The state transition diagram of an on-off voice node is illustrated in Fig. 2.4a, where $Q_{i,j}$ denotes the transition rate from state i to state j ($i, j \in \{0, 1\}$).

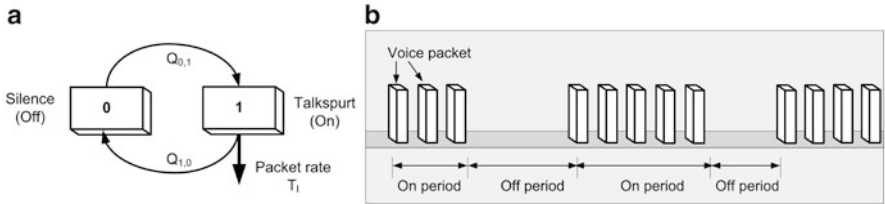


Fig. 2.4 (a) The state transition, (b) the codec output of an on-off voice source [20]

When a voice node is at the off state, it does not generate any packet; when at the on state (talk spurt), it generates voice packets at a rate of $1/T_I$. An output of a voice codec is illustrated in Fig. 2.4b. When an on-off voice node is at the off state

with an empty buffer, the voice node is at its inactive state, and when it is in the on state or off state with a non-empty buffer, it is in its active state.

2.2.3.3 Voice Buffer Management

Each voice source buffers the voice packets until it gets an spectrum (transmission) opportunity. The service requirement for the voice traffic flows in the secondary network is characterized by the end-to-end delay of a voice packet (i.e., from the time that a packet is generated at the source node to the time that it is received at the receiver node). As the packet propagation delay is negligible when compared with the time that a packet spend at the source buffer, the service requirement is given in terms of the queuing 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. The stochastic delay requirement is given by [3, 6, 22]

$$P(D > D_{\max}) \leq \epsilon \quad (2.1)$$

where D_{\max} (normalized to T_S) and ϵ are the delay bound and maximum delay bound violation probability allowed, 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} .

2.2.3.4 Voice Capacity

The voice capacity is defined as the maximum number of simultaneous voice calls that can be supported by the system, without violating the delay requirement given in (2.1) for all the admitted calls. In the analysis, it is important to note that the number of voice calls refers to the number of one-way voice calls.

2.2.3.5 Service Disciplines for Voice Traffic

For the centralized network, we consider an ideal scenario with the BS having the queue head waiting times of all the nodes.

- **FCFS service discipline:** The BS schedules packet transmission of the nodes in the available time slots based on the maximum queue head waiting time first principle.

For the distributed (fully-connected) CRN, two legacy channel access schemes, namely, the slot-ALOHA scheme and the round-robin scheme are considered, and the random allocation scheme is used as a benchmark for the performance comparison.

- **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 [23] or a mini-slot based scheme [6] 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.
- **Slot-ALOHA scheme:** All the nodes with a non-empty buffer will transmit with a probability q during an idle time-slot. If a collision occurs, each node will re-transmit at the next available time-slot with the same probability.
- **Random allocation:** One node act as a controller and assigns the channel access right to the other nodes randomly.

As the first step, we analyze the capacity of a centralized CRN with ideal information availability, and we extend the study to a distributed CRN with less information for the transmission decision. For simplicity of the analysis, we will only consider networks with stationary nodes.

2.3 Voice Capacity of the Centralized CRN

We analyze the voice capacity of a centralized CRN with a BS to schedule the channel access of each user, as illustrated in Fig. 2.1a. Only single-hop voice communication occurs between voice nodes, and all the voice flows have the same delay requirement. The arrival process and the service process of the centralized system are illustrated in Fig. 2.5, where the outputs of all the voice codecs are either on-off or constant-rate.

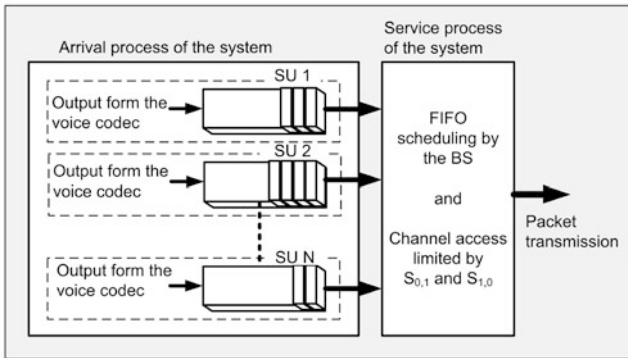


Fig. 2.5 The arrival process and the service process of the centralized system

2.3.1 Service Process Analysis

Since the channel is time-slotted as discussed in Sect. 2.2.2, its service process is a discrete-time process. The service process, $\mu_S(n)$, is defined as the number of packets that can be transmitted in the time-slot n , and is given by $\mu_S(n) = X_S(n) \cdot n_{pk}$, where $X_S(n) (\in \{0, 1\})$ is the channel availability index of the time-slot, and n_{pk} is the maximum number of packets that can be transmitted in a time-slot. Service process analysis with respect to the QoS requirement can be carried out using the theory of effective capacity (EC), as discussed in [24–29]. The EC provides the constant arrival rate that can be supported by the system (service process), without violating the required service quality. The EC analysis of a block fading channel [28, 30] can be adopted to analyze our channel by modeling the channel as a single block fading channel with two fading amplitudes (0 and 1). Therefore, the EC of the secondary network with dependent channel availability in adjacent time-slots is given by

$$\zeta_c(\theta) = -\frac{1}{\theta} \ln \left[\frac{S_{0,0} + S_{1,1}e^{-\theta n_{pk}}}{2} + \sqrt{\left(\frac{S_{0,0} - S_{1,1}e^{-\theta n_{pk}}}{2} \right)^2 + S_{0,1}S_{1,0}e^{-\theta n_{pk}}} \right] \quad (2.2)$$

where, θ depends on the QoS requirement, and it is shown in [25] that $P(D \geq D_{\max}) \approx e^{-\theta D_{\max}}$. In order to satisfy the condition (2.1), the parameter θ should satisfy the condition $\theta \geq \frac{1}{D_{\max}} \ln \left(\frac{1}{\epsilon} \right)$. In order to support a constant arrival rate r with the given delay requirement (2.1), $\delta^* = r \zeta_c^{-1}(r)$ should satisfy the condition

$$\delta^* \geq \frac{1}{D_{\max}} \ln \left(\frac{1}{\epsilon} \right). \quad (2.3)$$

The EC of the secondary network with independent channel availability scenario can be obtained by setting $S_{0,1} = S_{1,1} = p_1$ and $S_{1,0} = S_{0,0} = p_0 = 1 - p_1$.

2.3.2 Arrival Process Analysis

The capacity requirement of the constant-rate voice traffic sources remains $1/T_I$ packets/time-slot throughout the duration of the call. However, the capacity requirement of an on-off voice source varies with time. An on-off traffic source can be characterized by the mean, m , variance, ϑ , auto covariance time coefficient, ς , and peak-to-mean ratio, ν , of the traffic flow [31]. The four parameters are given by $m = Q_{0,1}/T_I(Q_{0,1} + Q_{1,0})$, $\vartheta = m(\frac{1}{T_I} - m)$, $\varsigma = 1/(Q_{0,1} + Q_{1,0})$, and $\nu = (Q_{0,1} + Q_{1,0})/Q_{0,1}$. The four parameters corresponding to the aggregate traffic from N independent sources are given by $m_a = N \cdot m$, $\vartheta_a = N \cdot \vartheta$, $\varsigma_a = \varsigma$, and $\nu_a = \nu$, respectively. The aggregate traffic flow can be characterized by a two-state Markov

modulated Poisson process (MMPP). The MMPP can be characterized by four parameters R_1 , R_2 , ϕ_1 , and ϕ_2 , where R_i is the mean rate of the Poisson process in state i , and ϕ_i is the transition rate from state i ($i \in \{1, 2\}$). The four parameters are given by, $R_1 = m_a + \sqrt{v_a \vartheta_a}$, $R_2 = m_a - \sqrt{\vartheta_a / v_a}$, $\phi_1 = v_a / \zeta_a (1 + v_a)$, and $\phi_2 = 1 / \zeta_a (1 + v_a)$ [31]. The capacity requirement of a time varying arrival process considering its service requirements is carried out using the theory of effective bandwidth (EB), as explained in [32,33]. The EB of an arrival process is the required constant service rate in order to satisfy the service quality requirement of the arrival process. The effective bandwidth of the two-state MMPP is given by [31]

$$\zeta_b(\theta) = \frac{\mathcal{E}(\Phi + (e^\theta - 1)\hat{R})}{\theta} \quad (2.4)$$

where Φ is the transition rate matrix of the two-state Markov chain of the aggregated traffic flow, $\hat{R} = \text{diag}(R_1, R_2)$, and $\mathcal{E}(\cdot)$ gives the largest real eigen value. In order to satisfy the given delay requirement (2.1) using a constant service rate u , $\delta^* = u \zeta_b^{-1}(u)$ should satisfy the condition (2.3) [24]. When both the arrival and service processes are time varying, in order to satisfy the delay requirement (2.1), $\delta^* = \theta^* \zeta_c(\theta^*)$ should satisfy the condition (2.3), where θ^* is the solution to the equation $\zeta_b(\theta) = \zeta_c(\theta)$ [25]. Note that, in the case of N constant-rate voice traffic sources, $\zeta_b(\delta) = N / T_l$. In order to determine the maximum number of voice sessions that can be supported by the system while satisfying the stochastic delay requirement (2.1), we have to find maximum N which satisfies (2.3).

2.4 Voice Capacity of the Distributed CRN

The constant-rate voice traffic capacity of a distributed fully-connected CRN will be studied in this section under round-robin and slot-ALOHA channel access schemes, and it will be compared with the random allocation scheme. In all three cases, even though the voice buffer of each node acts in the FCFS manner, the system with all the voice nodes as a whole does not behave in a FCFS manner. Therefore, the theory of EB and EC cannot be directly applied to the system, but, to each node. However, due to the complexity of analyzing the service process (of each node), we resort to packet level analysis of the voice buffer of each node.

2.4.1 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 with the highest waiting time becomes the new queue-head, χ_{new} . 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, D_{new} , of the new queue-head (normalized to T_S), is given by

$$D_{new} = D_{old} - n_s \cdot T_I + 1 \quad (2.5)$$

where D_{old} (normalized to T_S) is the waiting time of χ_{old} . The term $n_s \cdot T_I$ is due to the n_s 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 (same as the probability of successful channel access), $P_{S,1}$, in an available time-slot is given by

$$P_{S,1} = \varrho (1 - \rho \cdot \varrho)^{N-1} \quad (2.6)$$

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. 2.2. 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. 2.6. 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 [34]. Furthermore, the DTMC model is similar to the approach given in [6], in analyzing the constant-rate voice capacity of two different cognitive radio MAC protocols. Different from [6], 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

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

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

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

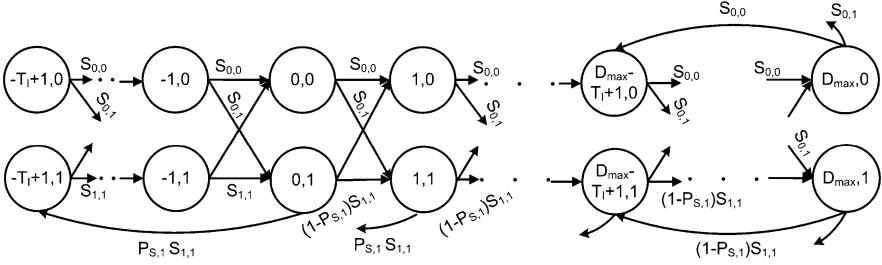


Fig. 2.6 The Markov chain for the queue-head waiting time and channel state pair

$$P_e = \frac{\sum_{j=0}^1 (1 - P_{S,j}) \cdot \pi_{D_{\max},j}}{P_{S,1} \cdot \sum_{i=0}^{D_{\max}} n_a(i) \cdot \pi_{i,1} + \sum_{j=0}^1 (1 - P_{S,j}) \cdot \pi_{D_{\max},j}} \quad (2.7)$$

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 $\sum_{j=0}^1 (1 - P_{S,j}) \cdot \pi_{D_{\max},j}$ 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 $P_e \leq \epsilon$. 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 (2.6). 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 $\pi_{i,j}$ ($i \in \{0, 1, \dots, D_{\max}\}$ 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 $P_e \leq \epsilon$ can be evaluated. The capacity analysis for the independent channel occupancy scenario can be carried out using the preceding method by substituting appropriate values for $S_{i,j}$ ($i, j \in \{0, 1\}$).

2.4.2 Random-Assignment Scheme

As the assignment is random, the probability of successful transmission in an available time-slot is given by $P_{S,1} = 1/N$, and is independent in adjacent available time-slots. Therefore, the same approach used with the slot-ALOHA scheme can be used to analyze the probability of delay bound violation and the voice capacity.

2.4.3 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 (2.5), 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

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

and

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

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. 2.7, where $P_{i,j}$ is the transition probability from state i to state j ($i, j \in \{0, 1, 2, \dots\}$).

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

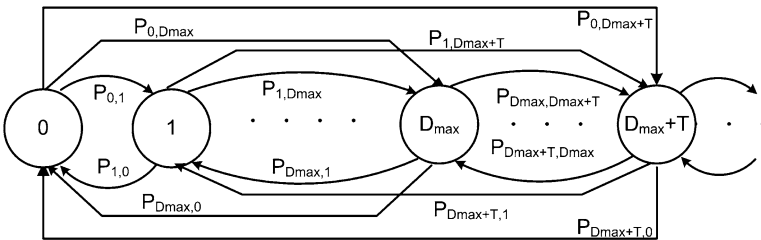


Fig. 2.7 The DTMC for the queue-head delay at the time of packet transmission with round-robin channel access

node number from Z to $Z - 1$,³ and $r = j - (i - (n_a(i) + n_a(i)) T_I)$ is the elapsed number of time-slots between adjacent channel access opportunities. The number of time-slots X_Z ($Z \in \{0, 1, \dots, N - 1\}$) 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. 2.8. 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

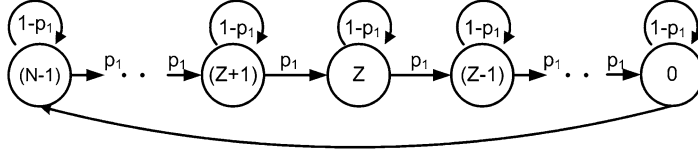


Fig. 2.8 State transition of Z for independent channel occupancy of PUs

$$P_{i,j} = \begin{cases} \binom{r-1}{r-N} p_1^N (1-p_1)^{r-N}, & \text{if } r \geq N \\ 0, & \text{otherwise.} \end{cases} \quad (2.8)$$

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. 2.9. 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

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

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

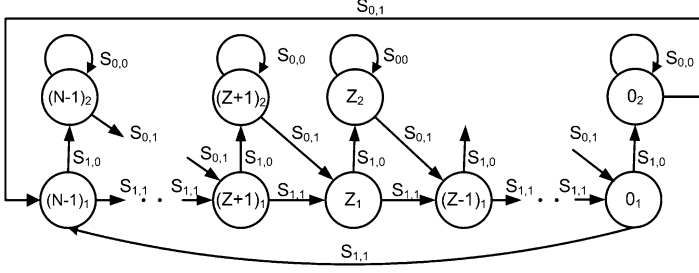


Fig. 2.9 State transition of Z for dependent channel occupancy of PUs

$$P_{i,j} = \begin{cases} \sum_{l=1}^{\min(r-N, N)} \binom{N}{N-l} S_{1,1}^{N-l} S_{1,0}^l \cdot \binom{r-N-1}{l-1} S_{0,1}^l S_{0,0}^{r-N-l}, & \text{if } r > N \\ S_{1,1}^N, & \text{if } r = N \\ 0, & \text{otherwise.} \end{cases} \quad (2.9)$$

In (2.9), 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

$$P_e \simeq \frac{\sum_{i=0}^{D_{\max}+k \cdot T_I} n_d(i) \cdot \pi_i}{\sum_{i=0}^{D_{\max}+k \cdot T_I} (n_d(i) + n_a(i)) \pi_i} \quad (2.10)$$

where π_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 capacity N_{\max} is the maximum N which satisfies the relation $P_e \leq \epsilon$. The larger the N , the larger the P_e . The minimum P_e , P_e^* , that can be analyzed by (2.10) is for the minimum N , N^* . As $N \geq T_I$, $N^* = T_I$. Thus, the capacity can be evaluated for an 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} .

2.5 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 longer 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.

2.5.1 CAC for Homogeneous Voice Traffic

In Sect. 2.4.1, 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 the set of neighboring receiver nodes of ω_s and source nodes of ω_r by $\in G_{\omega_s}$ and G_{ω_r} , respectively. 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. However, 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.

2.5.2 CAC 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 Little's theorem. Further, 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 [3, 22, 35]. 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 for non-fully-connected CRNs 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 Sect. 2.3.1 to analyze the effective capacity of the CRN can be adopted to analyze that of the service process of each node.

The packet buffer of each source node act in the FCFS 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 of the discrete-time service process for independent and dependent channel availability scenarios can be obtained by (2.2). In a particular time-slot (**irrespective of its availability**), define the state of the service process 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 for independent and dependent channel availability cases are given by

$$\zeta_c(\theta) = -\frac{1}{\theta} \ln \left[\frac{F_{0,0} + F_{1,1}e^{-\theta \cdot n_{pk}}}{2} + \sqrt{\left(\frac{F_{0,0} - F_{1,1}e^{-\theta \cdot n_{pk}}}{2} \right)^2 + F_{0,1}F_{1,0}e^{-\theta \cdot n_{pk}}} \right].$$

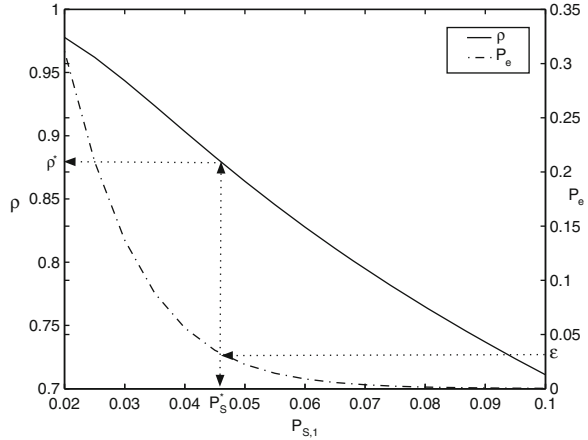
where, $F_{i,j}$ ($i, j \in \{0, 1\}$) is the transition probability of a node from state i to state j . In an **available time-slot** define the state of transmission (transmission state) as follows; If a transmission is successful, the source node is in state 1, and state 0 otherwise. Consider a Markov chain in which the state is represented by the channel state and transmission state pair which consist of three states (1,1), (1,0), and (0,0). Denote the state transition probability matrix of the Markov chain by \bar{F} . The node is at state 1 if both the transmission state and the channel state are 1, and state 0 otherwise. The state transition probability matrix F of the service process a source node can be obtained using the state transition probability matrix \bar{F} . The condition to satisfy the delay requirement of a voice call i is given by $\delta^* \zeta_c(\delta^*) \geq \frac{1}{D_{\max}} \log\left(\frac{1}{\epsilon}\right)$, where δ^* is the solution to the equation $\zeta_c(\delta) = \frac{1}{T_I}$.

This condition can be given in the form $\zeta_c(\delta_{\min}^*) \geq \frac{1}{T_I}$, 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

$$P_{S,1} = \prod_{i_s \in G_{\omega_r}} Q_{\omega}(1 - \rho_i \cdot Q_i) \quad (2.11)$$

where Q_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. 2.6, it can be seen 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. 2.10. As long as the existing source nodes satisfy the delay

Fig. 2.10 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 $\epsilon=0.01$



requirement $P_e \leq \epsilon$, the probability ρ is upper bounded by ρ^* . Therefore, instead of using $P_{S,1}$, we substitute $P_{S,1}^* = \prod_{i_s \in G_{\omega_r}} Q_{\omega}(1 - \rho_i^* \cdot Q_i) (\leq P_{S,1})$ in (2.11). 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 Q_i of $P_{S,1}^*$ should be replaced by their respective values of the traffic class c_i as $\rho^*(c_i)$ and $Q(c_i)$, where $Q(c_i)$ is the default Q 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 $1/T_l$. The benchmark CAC algorithm based on the effective capacity is given in algorithm A1.

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

```

1  $\hat{C}_i \leftarrow \emptyset$ ;
2 repeat
3   randomly select  $c_k$  from  $C$ ;
4    $C \leftarrow C - \{c_k\}$ ;
5   if  $i = \omega$  then
6      $P_{S,1} \leftarrow \varrho(c_k) \prod_{j \in G_{\omega_r}} (1 - \rho^*(c_j) \cdot \varrho(c_j))$ ;
7   else
8      $P_{S,1} \leftarrow \varrho(c_i) \prod_{j \in G_{i_r}} (1 - \rho^*(c_j) \cdot \varrho(c_j)) (1 - \rho^*(c_k) \cdot \varrho(c_k))$ ;
9   end
10   $\delta_{\min}^* \leftarrow \frac{T_l}{D_{\max}(c_i)} \log\left(\frac{1}{\epsilon}\right)$ ;
11  if  $\zeta(\delta_{\min}^*) \geq \frac{1}{T_l}$  then
12     $\hat{C}_i \leftarrow \hat{C}_i \cup c_k$ ;
13  end
14 until  $C = \emptyset$ ;
15 Exit;
```

Algorithm 1: CAC algorithm based on the effective capacity

Each receiver node in 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}_ω 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 $C_\omega = \bigcap_{i_r \in \{G_{\omega_s} \cup \omega_r\}} \hat{C}_i$ that can admit call ω . If $C_\omega = \emptyset$, 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 [24, 25] 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. 2.10, 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 $\varrho_\omega \prod_{i_s \in G_{\omega_r}} (1 - \rho_{i_s} \cdot \varrho_{i_s}) \geq \varrho_\omega \prod_{i_s \in G_{\omega_r}} (1 - \rho^*(c_i) \cdot \varrho_{i_s})$ 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 $\varrho_\omega \prod_{i_s \in G_{\omega_r}} (1 - \rho^*(c_i) \cdot \varrho_{i_s}) \geq P_S^*(c_\omega)$. However, as discussed in Sect. 2.5.1, 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

$$\varrho_\omega \cdot \prod_{i_s \in G_{\omega_r}} (1 - \rho^*(c_i) \cdot \varrho_i) \geq P_S^*(c_\omega) \quad (2.12)$$

where the LHSs of (2.12) are always less than or equal to $P_{S,1}$. The expressions of the LHSs of (2.12) can be evaluated using $\gamma_j (= \varrho_j \cdot \prod_{i \in G_{j_r}} (1 - \rho^*(c_i) \cdot \varrho_i))$ and c_j obtained from the neighboring receiver nodes of the new source node, and $\gamma_\omega (= \prod_{i \in G_{\omega_r}} (1 - \rho^*(c_i) \cdot \varrho_i))$ 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, parameter ϱ_{\min} is the minimal ϱ value which satisfies the first inequality in (2.12), ϱ_{\max} is the maximal ϱ value which satisfies the second inequality in (2.12), and $\hat{\beta}$ is the transmission probability selection parameter. Algorithm 2 searches for ϱ_{\min} and ϱ_{\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 $\varrho_{\min}\varrho_{\max}$ 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 $\hat{\beta}$ value should be selected for the network to obtain a ϱ_ω ($= \varrho_{\min} + \hat{\beta}(\varrho_{\max} - \varrho_{\min})$) value, such that the network capacity is maximized. This can be carried out by trial and error method off-line.

2.6 Numerical Results

Computer simulations are carried out to evaluate the accuracy of the capacity analysis of the given 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. The voice traffic classes used in the analysis are given in Table 2.1 Note that all the time durations are normalized to T_S . The typical values of the on and off durations of an on-off voice source are around 320 ms and 640 ms, respectively [20, 36]. However in [36], it is shown that these durations are dependent on the factors such as conversation topics and situations of voice calls. The probability of delay bound violation, P_e , is obtained by the ratio of the

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

```

1  $\kappa \leftarrow 0$ ;
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    $\kappa \leftarrow 1$ ;
18    $\varrho_{\max} \leftarrow \varrho_\omega$ ;
19    $\varrho_\omega \leftarrow \varrho_\omega + \varrho_s$ ;
20 end
21 if  $\kappa == 1$  then
22    $\varrho_\omega \leftarrow \varrho_{\min} + \hat{\beta}(\varrho_{\max} - \varrho_{\min})$ ;
23   Admit the call;
24 else
25   Block the call;
26 end
27 Exit;

```

Algorithm 2: CAC algorithm based on the successful transmission probability

Table 2.1 The voice traffic classes used in the simulations

Voice traffic class	Notation	Traffic type	Mean on duration	Mean off duration	T_I	D_{max}
Class 0	c_0	On-off	320	640	10	30
Class 1	c_1	Constant-rate	—	—	10	30
Class 2	c_2	Constant-rate	—	—	10	100
Class 3	c_3	Constant-rate	—	—	10	250

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. Each simulation result is obtained by averaging the results of 10 simulation runs each having a duration of 10^6 time-slots.

2.6.1 Capacity Analysis

Consider homogeneous voice traffic flows of class c_i ($i \in \{0, 1, 2, 3\}$). 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 $P_e \leq \epsilon$.

2.6.1.1 Centralized Network

Figure 2.11a, b show the variation of N_{\max} with n_{pk} in a centralized network with FCFS service discipline and different channel availability statistics for on-off (class c_0) and constant-rate (class c_1) voice traffic, respectively. The results demon-

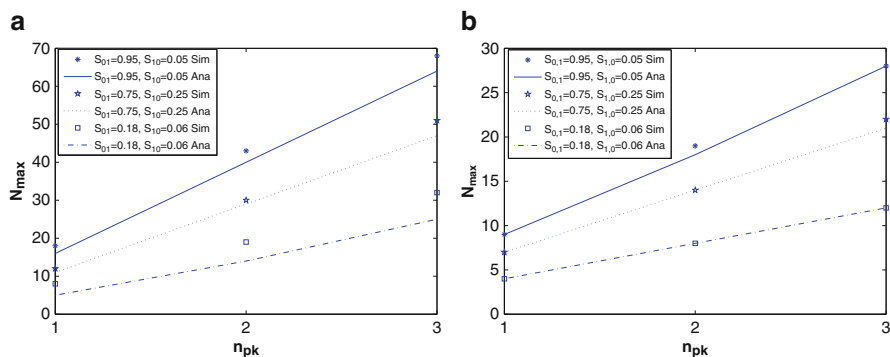


Fig. 2.11 Variation of N_{\max} with n_{pk} for on-off voice traffic in a (a) centralized, (b) distributed network with FCFS service discipline

strate that the analytical results match closely with the simulation results. However, the analytical results stay slightly below the simulation results due to the conservative nature of the theory of effective bandwidth and its dual, effective capacity. The capacity of the system increases with the number of voice packets that can be transmitted in a single time-slot (per channel), n_{pk} . However, a proportional increment in the channel rate is not required to increase the n_{pk} . Therefore, capability of transmitting multiple packets in a single-time-slot can have a considerable impact on the system capacity. In the on-off voice sources under consideration, the duration of talkspurts are only one third of the call duration. The number of voice packets generated in an on-off voice traffic flow is approximately one third of that of a constant-rate voice traffic flow. Therefore, with the given FCFS service discipline (the ideal scenario), the on-off voice traffic provides more than twice the capacity of the constant-rate voice traffic.

2.6.1.2 Distributed Network

Consider homogeneous voice traffic flows of class c_2 . Figure 2.12a–c show the variation of P_e with N obtained from numerical analysis and simulations with slot-ALOHA, random allocation, and round-robin channel access schemes, respectively, for different channel availability statistics with $n_{pk}=4$. The results demonstrate that the P_e obtained from simulation match well with the analytical results in all three scenarios. The system capacity N_{\max} is given by the maximum N having P_e less than ϵ (0.01 in our simulation). Further, it can be observed that the higher the mean channel availability, $p_1=S_{0,1}/(S_{0,1}+S_{1,0})$, the higher the system capacity in all three cases.

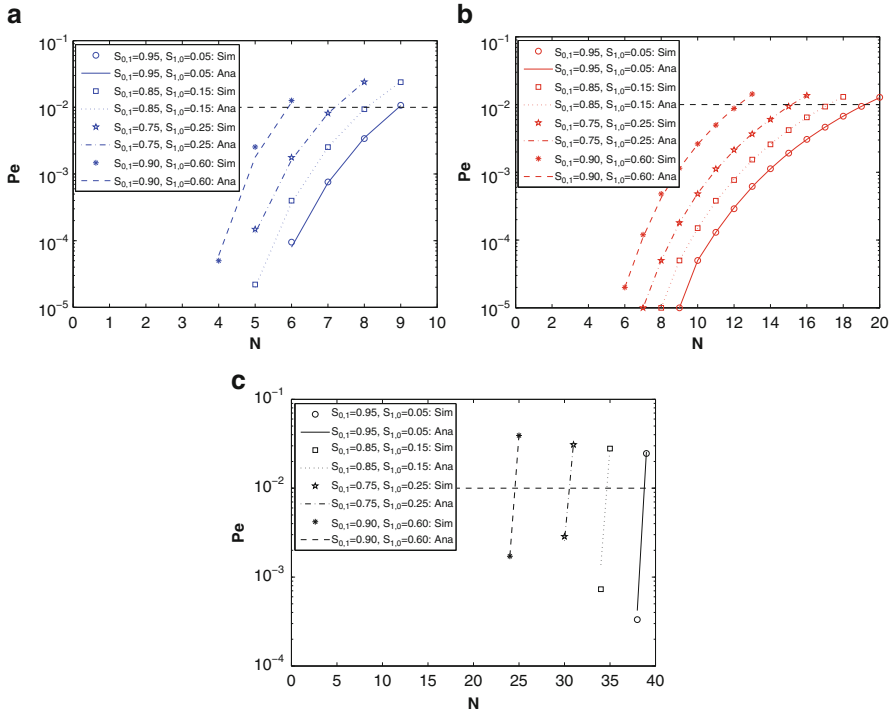
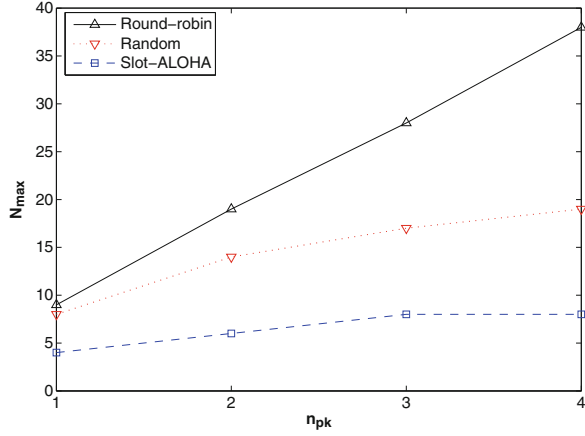


Fig. 2.12 Variation of P_e with N for a fully-connected network with (a) slot-ALOHA, (b) random allocation, (c) round-robin channel access

Figure 2.13 shows the variation of N_{\max} with n_{pk} for all three channel access schemes having p_1 constant in 0.95.

It is observed that the higher the mean channel availability and n_{pk} , the higher the capacity in all three channel access schemes. Further, the round-robin scheme provides the highest system capacity and slot-ALOHA provides the worst system capacity, when compared with the other two. Note that the overhead required

Fig. 2.13 Variation of N_{\max} with n_{pk} for a fully-connected network for all three channel access schemes

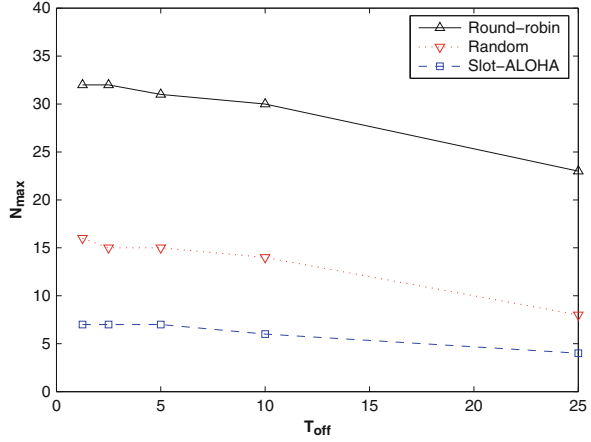


for the establishment of the round-robin scheme is much higher than that of the slot-ALOHA scheme, as explained in Sect. 2.2.3, which is neglected in the simulation. Furthermore, it is observed that the rate of increment of system capacity with n_{pk} in the round-robin scheme is higher than the random allocation and slot-ALOHA schemes. In the slot-ALOHA and random allocation schemes, 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 other two schemes. Therefore, the mean number of packets transmitted during a channel access opportunity is smaller in the slot-ALOHA and random allocation schemes than that in the round-robin scheme, which explains that the latter has a higher rate of capacity improvement with n_{pk} . The system capacity with the round-robin scheme is similar to that of centralized network with FCFS service discipline. Since, the round-robin scheme does not need the packet timing information, it can be considered as a promising candidate for voice service support over CRNs.

Figure 2.14 shows the variation of N_{\max} with T_{off} for all three channel access schemes having p_1 constant in 0.8 and $n_{pk} = 4$.

The results demonstrate that the longer the T_{off} , the lower the system capacity even though the mean channel availability remains constant. The longer the T_{off} , the longer the duration of the busy periods of the channel from the viewpoint of the SUs, leading to longer the durations of packet waiting times. This increases the probability of delay bound violation of the voice packets. Even if the channel available duration is longer (corresponding to longer T_{on}) it is not possible to transmit the packets which have already violated the delay bound. Therefore, the longer the T_{off} , the lower the system capacity. This shows the importance of considering the state transition probabilities of the channel in analyzing the system capacity rather than considering the mean channel availability.

Fig. 2.14 Variation of N_{\max} with T_{off} for a fully-connected network for all three channel access schemes



2.6.1.3 Discussion

In our work, we assume ideal channel sensing (i.e., error free detection of primary activities). However, sensing errors are inevitable in practical scenarios. There are two types of sensing errors, namely, missed detection (an SU or the BS does not detect the presence of a PU) and false alarm (an SU or the BS detects presence of a PU while the PUs is not present) [37]. The missed detections lead SUs to transmit simultaneously with the PUs, causing interference. In order to establish the CRN, the probability, P_{md} , of missed detection has to be controlled below a certain threshold to minimize the interference with the PUs. Despite the interference, the SU transmission can be successful. The false alarms reduce the channel utilization of the SUs. Therefore, the probability, P_{fa} , of false alarm has to be minimized to improve channel utilization of the SUs. From the viewpoint of an SU (or the BS), the channel availability differs from the true state of the channel due to the presence of the sensing errors. Denote the channel state from the viewpoint of an SU (or the BS) as the virtual channel state. The virtual channel state transition probabilities can be obtained using the channel state transition probability matrix and the error probabilities P_{md} and P_{fa} , respectively. We can incorporate the effect of the sensing errors into our capacity analysis after some efforts on modification of the Markov chains.

When the sensing errors are present, packet transmissions of the SUs in the centralized network depend on the virtual channel state of the BS. In order to incorporate the sensing errors into the capacity analysis, the channel state transition probabilities in (2.2) should be replaced by the virtual channel state transition probabilities. In the fully-connected slot-ALOHA network, the packet transmission of an SU depends on the channel sensing errors. Therefore, the state transition probabilities of the DTMC in Fig. 2.6 should be modified to incorporate the sensing errors. The modifications to the state transitions are illustrated in Fig. 2.15, where $P_{S,md}$ is the probability of successful transmission given the occurrence of a

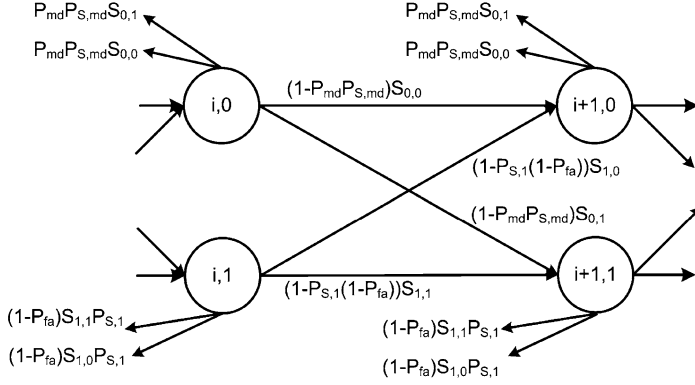


Fig. 2.15 The modifications to the state transitions of the DTMC in Fig. 2.6 to incorporate sensing errors

missed detection and $P_{S,1} = \rho(1 - \rho\varrho(1 - P_{fa}))^{N-1}$ is the successful transmission probability given the channel is available. The transition probabilities $P_{md} P_{S,md} S_{0,j}$ and $(1 - P_{fa}) P_{S,1} S_{1,j}$ are due to successful transmissions given the channel is not available (transmission being successful due to missed detection) and available (when there is no false alarm), respectively, where $j \in \{0, 1\}$. The transition probability $(1 - (1 - P_{fa}) P_{S,1}) S_{1,j}$ and $(1 - P_{md} P_{S,md}) S_{0,j}$ are due to non occurrence of packet transmissions when the channel is available and not available, respectively, where $j \in \{0, 1\}$. Similar to that given in Sect. 2.4.1, the delay bound violation probability can be evaluated by finding the steady state probabilities of the DTMC. The sensing errors can be incorporated into the capacity analysis of the round-robin scheme by modifying the state transition probabilities of the DTMC in Fig. 2.9, and the modification depends on the mechanism used to establish the round-robin scheme. With the above, it is clear that the sensing errors can be incorporated in our capacity analysis with slight modifications in the state transition probabilities of the DTMCs.

In a multiple channel network, an SU will either select a channel and sense for availability or sense all the channels and select an available channel. Two approaches lead to different successful transmission probabilities. We can extend our capacity analysis approach for a multiple channel CRN by evaluating the corresponding successful transmission probabilities. The service quality is given in terms of queuing delay D . The QoS requirement can be relaxed by defining the service quality in terms of the mean of queuing delay, $E[D]$.

As mobile video is predicted to generate most of the mobile traffic growth through 2005 [1], it is important to study video streaming over the CRNs. As given in [38], video frames are generated in burst according to a coding and compression algorithm, and each video burst consists of a number of video packets (with a pdf given by negative binomial distribution). The video clips are grouped into a small number of shot classes depending on the burst size, and a video traffic flow is

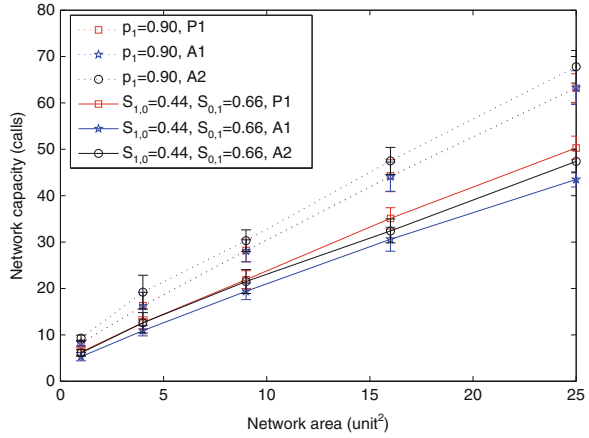
modeled my a Markov modulated Gamma process in [39]. The author also analyze the EB of the video traffic flow for a maximum data loss rate of 10^{-2} . Therefore, the number of video traffic flows that can be supported by the centralized FCFS system can be studied using the EC evaluated in Sect. 2.3.1 and the EB approach in [39]. For the distributed networks, the possibility to carry out a packet level analysis of the source buffer can be studied, given the statistics of the video bursts and the probability of channel access. Further research is necessary to model the source buffer state using a Markov chain, and to analyze the packet dropping probability using the steady state probabilities as given in Sect. 2.4.1.

2.6.2 Call Admission Control

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_2 or c_3 . 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 8,000 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.

Figure 2.16 shows the comparison of the network capacity (with the 95% confidence interval) of class c_2 voice calls using procedure P1 and algorithms A1 and A2. The CAC procedure P1 outperforms the algorithms A1 and A2 when the mean channel availability is lower, and the algorithm A2 outperforms the other two when the mean channel availability is higher. The algorithm A2 opportunistically chooses the probability q at the instance of call admission whereas P1 has a fixed q value. Therefore the opportunistic q selection may choose different q values for different calls leading to a probability $P_{S,1}$ which is just enough to satisfy the admission criterion (q can vary from q_{min} to q_{max}). The lower the channel availability, the larger the P_S^* . The larger 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 q 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

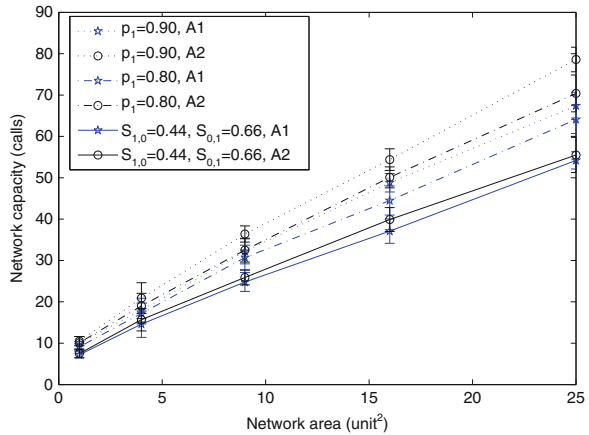
Fig. 2.16 Variation of the network capacity with the network area for procedure P1 and algorithms A1 and A2



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.

Figure 2.17 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.

Fig. 2.17 Variation of the network capacity with the network area for algorithms A1 and A2 for a network with two voice traffic classes



The results demonstrate that 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_2 . The relaxed QoS requirement of voice traffic class c_3 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.

2.7 Summary

In this chapter, we have studied the voice capacity and call admission control for CRNs. A stochastic delay requirement, both independent and dependent channel availability statistics, and different channel access schemes are considered. The on-off voice capacity of a centralized CRN is studied for the FCFS service discipline using the theories of effective bandwidth and its dual effective capacity. The analytical results appear to be slightly lower than the simulation results due to the conservative nature of the theory of effective bandwidth. Further, it is observed that the silent-suppressed (on-off) voice sources provides more than twice the capacity over that of constant-rate voice traffic. The existing DTMC model is modified to analyze the capacity of slot-ALOHA scheme, and a new DTMC model is developed to analyze the capacity of round-robin scheme in supporting constant-rate voice traffic over distributed fully-connects CRNs. It is shown that the round-robin scheme performs better than the other two schemes, and the capacity is very close to that of the FCFS scheme used in the centralized network. Therefore, the round-robin scheme is a better choice in fully-connected networks to support voice traffic, and it can be established using a token based scheme or a mini-slot based scheme as explained in [6, 23]. Further, the maximum number of voice packets that can be transmitted in a time-slot and the mean channel unavailable duration have a significant impact on the system capacity. The longer the mean channel unavailable durations the lower the system capacity, even when the mean channel availability remains unchanged. In order to relax the assumption of perfect channel sensing, possible extensions to the analytical models to incorporate the sensing errors are discussed.

We use the capacity analysis results of the fully-connected network to limit the number of neighboring users of each target voice user in non-fully-connected CRNs with slot-ALOHA network coordination. It is only applicable for constant-rate voice sources with the same delay requirement. However, having long delay bounds and large delay bound violation probabilities can increase the system capacity. Therefore, we develop two new CAC algorithms to support voice sources with different delay requirements (different delay bound and maximum delay bound violation probability). It is shown that the longer the delay bound, the larger the system capacity. In other words, the lower the required service quality, the higher the system capacity. A low quality service can be priced at a lower cost than a high quality service. Giving the users an option to choose the required service quality can increase the level of user satisfaction.

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Radio Networks

Gunawardena, S.; Zhuang, W.

2014, VIII, 91 p. 49 illus., Softcover

ISBN: 978-3-319-04644-0