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Performance Evaluation of the VoIP Services of the Cognitive Radio System, Based on DTMC

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Abstract—In recent literature on traffic scheduling, the combination of the two-dimensional discrete-time Markov chain (DTMC) and the Markov modulated Poisson process (MMPP) is used to analyze the capacity of VoIP traffic in the cognitive radio system. The performance of the cognitive radio system solely depends on the accuracy of spectrum sensing techniques, the minimization of false alarms, and the scheduling of traffic channels. In this paper, we only emphasize the scheduling of traffic channels (i.e., traffic handling techniques for the primary user [PU] and the secondary user [SU]). We consider the following three different traffic models: the cross-layer analytical model, M/G/1(m) traffic, and the IEEE 802.16e/m scheduling approach to evaluate the performance of the VoIP services of the cognitive radio system from the context of blocking probability and throughput.

Keywords—Cognitive Radio, VoIP, DTMC, Cross-layer Analytical Model, M/G/1(m) Traffic, IEEE 802.16e/m

1. INTRODUCTION

The increasing demand for high data rate wireless voice services over the Internet has led to numerous research initiatives in the field of wireless communication systems. The desired communication system should have enhanced computational intelligence along with higher access speeds, quality of service assurance, and the ability to successfully handle multiple users simultaneously. The growing demand of users has also brought the need for the flexible and efficient allocation of available spectrum resources. Spectrum regulatory entities have found that spectrum usage is currently concentrated over a certain portion of the spectrum and that most of the licensed radio frequency spectrum is inadequately utilized [1]. Improved utilization of the spectrum is one of the key drivers for the development of the cognitive radio (CR), which was first proposed by Mitola and Magure [2]. A CR can use an unused licensed spectrum as a secondary user (SU), which ensures that licensed primary users (PUs) are not interfered with by its transmission. For the successful transmission of secondary users, it is very important to determine the vacant radio frequency spectrum of the PU, which is also referred to as the spectrum hole. While determining the spectrum hole, the probability of a false alarm should be minimal and the level of interference towards the PUs should be at the acceptable level. If the CR detects that the level of interference towards the PUs has exceeded the threshold level, it releases the spectrum and switches to another spectrum for transmission. A CR system can support the multiuser environment efficiently with a limited number of spectrum holes. To

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overcome the challenge of the multiple access control in the CR system, an effective modulation strategy and traffic model that can adapt to the time varying conditions of the radio environment assuring the reliable communication [3] need to be chosen. For future communication systems, VoIP (Voice over IP) is one of the essential technologies that can provide users with satisfactory voice services at a lower cost. VoIP can support as many users as possible with a satisfactory quality of service. Therefore, the combination of VoIP technology with the CR system has the potential to provide reliable multimedia wireless services with the efficient use of the limited radio spectrum. In this case, the CR system should be able to support real time traffic with service satisfaction. The voice capacity of the CR system and the performance of the VoIP services also needs to be investigated. Hence, in this paper we have concentrated on the performance analysis of VoIP services within a CR system by using different types of traffic models. In the CR system, VoIP services can be modeled by choosing an efficient traffic model and modulation strategy. In [4], the VoIP capacity and the performance of VoIP services have been analyzed using the queuing model for the discrete-time Markov chain (DTMC) framework based on a Markov modulated Poisson process (MMPP) traffic model. However, the authors have concentrated more on finding the minimum target-detection and false alarm probabilities. A cross-layer analytical model has been proposed in [5] to ensure the quality of service for the secondary CRs.

In this paper, we consider the cross-layer analytical model for DTMC based VoIP services in the CR system. Furthermore, the traffic control schemes for M/G/1(m) traffic discussed in [6, 7] have been analyzed in the context of a CR system. Here, we also investigate VoIP performance in the CR system using the scheduling algorithm with an adaptive multi rate (AMR) speech codec in IEEE 802.16e/m[8]. In this paper, we have made an effort to evaluate the performance of the VoIP services in the CR system for three different traffic models and to determine an efficient traffic model that can be implemented for the DTMC based secondary VoIP users of the CR system [9, 10].

The remainder of the paper is organized as follows: Section 2 outlines the VoIP traffic and the system model along with the details of the three traffic models that we consider for the CR system, which are the cross-layer analytical model, M/G/1(m) traffic, and the IEEE802.16e/m traffic scheduling model. The results and analysis are presented in Section 3. Finally, Section 4 concludes the paper.

2. SYSTEM MODEL

2.1 VoIP Traffic using DTMC

VoIP traffic can be modeled using the discrete-time Markov chain (DTMC) by considering the following two parameters: (a) change in the number of queuing packets in two adjacent frames and (b) change in the number of unoccupied channels between two adjacent frames. Changes in the above two parameters are expressed by a transition matrix as [4]:

$$\mathbf{P} = \begin{bmatrix} \mathbf{A}_{0,0} & \mathbf{A}_{0,1} & \cdots & \mathbf{A}_{0,Q_{max}} \\ \vdots & \vdots & \ddots & \vdots \\ \mathbf{A}_{Q_{max},0} & \mathbf{A}_{Q_{max},1} & \cdots & \mathbf{A}_{Q_{max},Q_{max}} \end{bmatrix}, \tag{1}$$

where Q_{max} is the maximum length of queue. Each element $\mathbf{A}_{i,j}$ of the matrix $\mathbf{P}$ is another matrix that represents that there are i packets in the queue in the duration of the current frame and it will change to j packets in the next frame. The matrix $\mathbf{A}_{i,j}$ is expressed in a generalized form as:

$$\mathbf{A}_{i,j} = \begin{bmatrix} \mathbf{B}_{(i,0)(j,0)} & \mathbf{B}_{(i,0)(j,1)} & \cdots & \mathbf{B}_{(i,0)(j,N)} \\ \vdots & \vdots & \ddots & \vdots \\ \mathbf{B}_{(i,N)(j,0)} & \mathbf{B}_{(i,N)(j,1)} & \cdots & \mathbf{B}_{(i,N)(j,N)} \end{bmatrix}, \tag{2}$$

where each element $\mathbf{B}_{(i,m)(j,n)}$ of the matrix $\mathbf{A}_{i,j}$ is another matrix of size 2×2 representing the change in the number of unoccupied channels from m to n when the number of packets in the queue changes from i to j. Here, N is the total number of channels.

Any element $\mathbf{B}_{(i,m)(j,n)}$ of the matrix $\mathbf{A}_{i,j}$ is evaluated as:

$$\mathbf{B}_{(i,m)(j,n)} = \sum_{k=0}^{N} \mathbf{U}.\mathbf{D}\big(j - max(i-k,0)\big) P_s\big(k \big| x_c = m\big)\, P_{m,n}\,, \tag{3}$$

where:

$$P_{m,n} = \sum_{x'=max(0,\,m-n)}^{min(m,\,N-n)} \binom{m}{x'} P_{01}^{x'} P_{00}^{m-x'} \binom{N-m}{y'} P_{10}^{y'} P_{00}^{N-m-y'}$$

is the transition probability that indicates the number of unoccupied channels changes from m (current frame) to n (next frame), which are derived from the two-state MMPP model. Here, $P_s\big(k \big| x_c = m\big)$ represents the probability that a node serves k packets when the number of unoccupied channels is m and $y' = n - m + x'$. From the two-state discrete time MMPP model, the diagonal probability matrix $\mathbf{D}(k)$ and the transition probability matrix $\mathbf{U}$ are expressed respectively as [11]:

$$\mathbf{D}(k) = \begin{bmatrix} \dfrac{\left(\lambda_1 T_f\right)^k e^{-\lambda_1 T_f}}{k!} & 0 \\ 0 & \dfrac{\left(\lambda_2 T_f\right)^k e^{-\lambda_2 T_f}}{k!} \end{bmatrix} \tag{4}$$

and:

$$\mathbf{U} = \left(\mathbf{\Lambda} - \mathbf{R}\right)^{-1} \mathbf{\Lambda}\,, \tag{5}$$

where the transition rate matrix $\mathbf{R}$ and the Poisson arrival rate matrix $\mathbf{\Lambda}$ are expressed as follows:

$$\mathbf{R} = \begin{bmatrix} -r_1 & r_1 \\ r_2 & -r_2 \end{bmatrix}, \tag{6}$$

and:

$$\mathbf{\Lambda} = \begin{bmatrix} \lambda_1 & 0 \\ 0 & \lambda_2 \end{bmatrix}. \tag{7}$$

Here, λ_1 and λ_2 are the mean arrival rates, r_1^{-1} and r_2^{-1} are the mean sojourn times of the two-state MMPP, and T_f is the frame duration. Each element of the $\mathbf{D}(k)$ matrix provides the probability of the arrival of k packets over a frame duration of T_f.

Now, the steady probability vector $\mathbf{\Pi}$ can be obtained using the relations $\mathbf{\Pi}_p\mathbf{P} = \mathbf{\Pi}_p$ and $\mathbf{\Pi}_p\mathbf{e} = 1$; where $\mathbf{e}$ is a vector where all elements are 1. The dimension of the vector $\mathbf{\Pi}_p$ is 1 by $2\times(N+1)\times(Q_{max}+1)$.

The general expression of the relation $\mathbf{\Pi}_p\mathbf{P} = \mathbf{\Pi}_p$ with $\mathbf{\Pi}_p\mathbf{e} = 1$ is derived as:

$$\begin{bmatrix} 0 & \{P_{21}-(P_{11}-1)\} & \{P_{31}-(P_{11}-1)\} & \cdots & \{P_{N1}-(P_{11}-1)\} \\ \{P_{12}-(P_{21}-1)\} & 0 & \{P_{32}-(P_{22}-1)\} & \cdots & \{P_{N2}-(P_{22}-1)\} \\ \vdots & \vdots & \vdots & & \vdots \\ \{P_{1N}-(P_{NN}-1)\} & \{P_{2N}-(P_{NN}-1)\} & \{P_{3N}-(P_{NN}-1)\} & \cdots & 0 \end{bmatrix} \cdot \begin{bmatrix} \pi_1 \\ \pi_2 \\ \vdots \\ \pi_N \end{bmatrix} = \begin{bmatrix} 1-P_{11} \\ 1-P_{22} \\ \vdots \\ 1-P_{NN} \end{bmatrix}.$$

The probability of k packets waiting in queue is found as:

$$\pi(k) = \sum_{i=0}^{2(N+1)-1} \pi_p(2k(N+1)+i). \tag{8}$$

The steady probability vector of size 1 by $(Q_{max}+1)$ is

$$\mathbf{\Pi} = \left[\pi(0) \quad \pi(1) \quad \ldots \quad \pi(Q_{max})\right]. \tag{9}$$

The mean queue length is:

$$Q_{avg} = \sum_{i=0}^{Q_{max}} i\pi(i). \tag{10}$$

The average arrival rate is:

$$\rho_{av} = \mathbf{s}\left\{\sum_{i=0}^{N.A_{max}} i\,\mathbf{D}(i)\right\}\mathbf{e}, \tag{11}$$

where A_{max} is the maximum number of packets that have arrived from a node during the interval $[0, T_f]$ and the vector $\mathbf{s} = [s_1, s_2]$ is obtained from the relations $\mathbf{sU} = \mathbf{s}$ and $s_1 + s_2 = 1$, where U is given by equation (5).

The average number of served VoIP packets under spectrum sensing statistics $T(x)$ is:

$$k_{av} = \sum_{i=0}^{N}\sum_{j=0}^{N}\sum_{k=0}^{Q_{max}} min(j,k)\pi(k)p_s(j \mid x_c = i)R(p_d, p_f), \tag{12}$$

where $R(p_d, p_f)$ is the ratio of the number of successfully transmitted packets and the number of scheduled packets.

The average throughput is calculated as:

$$S = k_{av} l_{VoIP}, \tag{13}$$

where l_{VoIP} is the length of the packet data unit (PDU) of the VoIP. Finally, packet-blocking probability is calculated as:

$$\beta = 1 - \frac{k_{av}}{\rho}. \tag{14}$$

2.2 VoIP Traffic under the Cross-Layer Analytical Model

A cross-layer analytical model of the Cognitive Radio is proposed in [5]. The traffic parameters of the PUs are the call arrival rate λ_p, and the call termination rate μ_p. The blocking probability of PU is evaluated using the M/M/N model.

Now, the probability of a channel is kept unoccupied by the PUs that are sensed by the SU, which is expressed as:

$$B = (1-\eta)(1-P_f), \tag{15}$$

where P_f is the probability of false alarms sensed by the SU and η is the channel utilization factor of the PUs, which is expressed as:

$$\eta = \frac{A_p(1-B_p)}{n}, \tag{16}$$

where $A_p = \lambda_p / \mu_p$ is the offered traffic and B_p is the blocking probability of the PUs.

The probability of z channels being unoccupied and correctly determined as idle and remaining unoccupied during the spectrum-sensing period can be derived as:

$$P(z \mid m) = \binom{m}{z}(1-B)^{m-z} e^{-\lambda_p T_p}, \tag{17}$$

where T_p is the spectrum-sensing period.

Now, the probability $P(k \mid x_c = m)$ that appears as in Equation (3) can be expressed as:

$$P(k \mid x_c = m) = \{1 - P(N, A_s, k)\} Y(m), \tag{18}$$

where:

$$P(N, A_s, k) = \frac{\binom{N}{k} A_s}{\sum_{r=0}^{k} A_s^r \binom{N}{r}},$$

and:

$$Y(m) = P(m \mid n).$$

2.3 Connection Oriented Packet Switching of M/G/1(m) Traffic

Let λ_j, h_j, V_j, and y_j be the call arrival rate, mean call holding (connection) time, data speed, and the data activity rate for class j∈J, respectively. The input traffic load a and the utilization ρ_{co} of the transmission line are then given by:

$$a = \sum_{j \in J} \lambda_j h_j, \tag{19}$$

and:

$$\rho_{co} = (1 - B) \frac{L_c}{L_p} \frac{1}{c} \sum_{j \in J} \lambda_j h_j V_j y_j, \tag{20}$$

respectively, where B is the call blocking probability, L_c is the cell length (53 bytes for ATM), L_p is the payload length (48 bytes for ATM), and c is the transmission speed.

We have Erlang's loss formula:

$$B = \frac{a^s}{s!} \Big/ \sum_{i=0}^{s} \frac{a^i}{i!}, \tag{21}$$

where s is the number of virtual channels (VCs) assigned to the transmission line.

According to [6,7] for the M/G/1 system, we have:

$$\Pi_j^* = p_j \Pi_0^* + \sum_{k=1}^{j+1} p_{j-k+1} \Pi_k^*; \quad j = 0,1,2,\ldots, \tag{22}$$

where p_j is the probability that j calls arrive during the service time.

The M/G/1(m) system, $\mathbf{\Pi}_j^* = 0$ for $j > m+1$ and the equation (22) take the following recurrence form:

$$\Pi_{j+1}^{*} = \left(\Pi_{j}^{*} - p_j \Pi_0^{*} - \sum_{k=1}^{j} p_{j-k+1} \Pi_k^{*} \right) p_0^{-1} \,;\; j = 0,1,2,...,m-1 \,. \tag{23}$$

Taking, $C_j = \Pi_j^{*} \big/ \Pi_0^{*}$ we get:

$$C_{j+1} = \left(C_j - p_j - \sum_{k=1}^{j} p_{j-k+1} C_k \right) p_0^{-1} \,;\; j = 0,1,2,...,m-1 \,. \tag{24}$$

Let, $C = \sum_{j=0}^{m} C_j$ with $C_0 = 1$ and combining the above equations, we obtain:

$$\Pi_j = \Pi_j^{*} = P_j = \frac{C_j}{1 + aC} \,. \tag{25}$$

The cell loss rate is then:

$$B = P_{m+1} = 1 - \sum_{j=0}^{m} P_j = 1 - \sum_{j=0}^{m} \frac{C_j}{1 + aC} = 1 - \frac{C}{1 + \rho_{co} C} \,.$$

Here, the probability $P(k \mid x_c = m)$ can be expressed as:

$$P(k \mid x_c = m) = \Pi(k) Y(m) \,, \tag{26}$$

with $Y(m) = P(m \mid n)$.

2.4 VoIP Traffic under IEEE 802.16e/m

In VoIP traffic, there is always a tradeoff between the efficient utilization of radio resources and the quality of service. In the IEEE 802.16e/m model, different algorithms are prevalent for the scheduling of resources to support the talk-spurt and silent-period of a user. One of the popular VoIP scheduling services under IEEE 802.16e/m is unsolicited grant service (UGS), where a base station (BS) grants a constant bandwidth (BW) to a subscriber station (SS) during the entire service time [11,12]. Here, BS periodically assigns a grant to a SS, irrespective of the state of the SS (i.e., its silent and talk-spurt states). During the silent-period any allocation of BW is simply a waste of the utilization of a channel. Therefore, the UGS algorithm is inefficient for the case of VoIP scheduling. The adaptive multi-rate (AMR) speech codec under the IEEE 802.16e/m system has overcome the above-mentioned waste of channel utilization. Here, a speech frame of 95-477 bits is generated periodically on every 20ms during the talk-spurt, while 40 bits of silence descriptor (SID) frames are generated every 160ms during the silent-period. For further improvement in the utilization of a channel for VoIP service over AMR speech codec, a new algorithm is proposed in [8]. According to that algorithm, SID is sent in a random access method, instead of via a periodic transmission during the silent-period. This is done to save the BW of the uplink. When the state of a SS changes from the silent-period to a talk-spurt

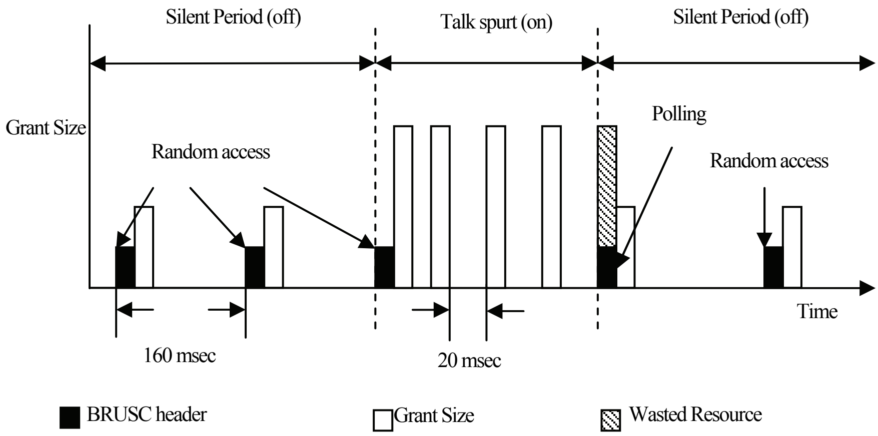

Fig. 1. VoIP scheduling algorithm

state, the access technique is still maintained as random during the transition time since the duration of this transition time is unpredictable. Only during the talk-spurt time does the BS provide an allocation (grant) to the SS, where the grant size can vary according to the required data rate. Any SS can use the reserved bit and bandwidth-request (BR) field of the "bandwidth request and uplink sleep control" (BRUSC) to apprise the BS about the state of the SS and the required bytes. In this case, a SS need not to use a generic-MAC header. The above VoIP scheduling algorithm is depicted in Fig. 1. The AMR speech codec and the corresponding queuing method can be modeled by a one dimensional Markov chain, as shown in Fig. 2.

Here, we consider the voice traffic to be exponentially distributed with a mean on-time of $1/\lambda$ and a mean off-time of $1/\mu$. Taking the length of the queue to be k, and by applying cut equations on Fig. 2, we derive the steady probability states, blocking probability, and the probability of entering the queue.

The probability state is expressed as:

$$P_x = \begin{cases} P_0 \sum_{x=0}^{m} \binom{M}{x} a^x; \quad 0 \le x \le m \\ P_0 \binom{M}{m} (N-m)!\, a^m \sum_{x=1}^{M} \frac{\left(\frac{a}{m}\right) x}{(M-m-x+2)!} \quad ; \quad m < x \le M \end{cases} \tag{27}$$

where:

$$P_0 = \left[\sum_{x=0}^{m} \binom{M}{x} a^x + (M-m)! \binom{M}{m} a^m \sum_{y=1}^{M} \frac{\left(\frac{a}{m}\right)^y}{(M-m-y+2)!} \right]^{-1}.$$

The probability of entering queue is:

$$Q(a,M,m,k) = \sum_{y=1}^{k} \frac{(M-m)!}{(M-m-k)!} \binom{M}{m} \left(\frac{a}{m}\right)^{y} P_0 a^m \ . \tag{28}$$

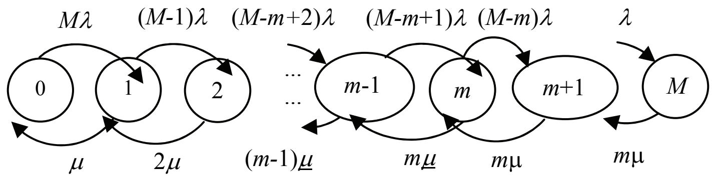

Fig. 2. Markov chain of VoIP users for the on-off system

The mean queue length is:

$$\overline{Q} = \sum_{r=1}^{k} rQ(a,M,m,r) \tag{29}$$

and the blocking probability is expressed as:

$$B(a,M,m,k) = \frac{(M-m)!}{(M-m-k)!} \binom{M}{m} \left(\frac{a}{m}\right)^{k} P_0 a^m \ . \tag{30}$$

Based on the equations (27)-(30), the probability $P(k \mid x_c = m)$ of Equation (3) can be expressed as:

$$P(k \mid x_c = m) = \{1 - P(m,k)S(k)\}, \tag{31}$$

where:

$$P(x,y) = a^x \binom{M}{x} P_0(y),$$

and:

$$S(k) = \frac{(M-m)!}{(M-m-k)!} \binom{M}{m} . \left(\frac{a}{m}\right)^{k} P_0(k) a^m \ .$$

The first term indicates the probability of occupancy of m channels by the PU and the second term indicates the probability that k packets exist in the queue. Finally, we have:

$$P_0(k) = \left[\sum_{r=0}^{m} a^r \binom{M}{r} + (M-m) \binom{M}{m} a^m \sum_{r=1}^{k} \left(\frac{a}{m}\right)^{r} \frac{1}{M-m-r} \right]^{-1} . \tag{32}$$

3. RESULTS

For each of the traffic models, we have taken two sets of the matrix $\mathbf{P}$: one for the $Q_{max} = 3$ and another is for $Q_{max} = 10$. To form the matrix $\mathbf{P}$ for $Q_{max} = 3$, we take 3 matrices of $\mathbf{A}_{i,j}$ where the size of each of $\mathbf{A}_{i,j}$ matrix is 8×8 . Thus, the size of the final $\mathbf{P}$ matrix becomes 24×24 . For $Q_{max} = 10$, the size of the matrix $\mathbf{P}$ becomes 80×80 with 10 matrices of $\mathbf{A}_{i,j}$. Considering the total number of channels $N = 3$, we obtain the 2×2 matrix $\mathbf{B}_{(i,n)(j,m)}$ and take some typical values of overloaded network like: $P_{00} = 0.6$, $P_{01} = 0.15$, $P_{10} = 0.75$ and $P_{11} = 0.24$. To get the diagonal matrix $\mathbf{D}(k)$ and the transition probability matrix $\mathbf{U}$, we assume that $T_f = 1$, $\lambda_1 = 0.656$, $\lambda_2 = 0.842$, $r_1 = 0.21$, and $r_2 = 0.12$ (considering an adverse condition of the network that stays in an underloaded and overloaded state at 36% and 64% of the observation time, therefore, $0.36 = r_1/(r_1+r_2)$ and $0.64 = r_2/(r_1+r_2)$ hence $r_1 = 0.21$ and $r_2 = 0.12$) for the VoIP traffic under the cross-layer analytical model and IEEE 802.16e/m. For the VoIP traffic under the cross-layer analytical model and M/G/1(m) traffic, we chose $P_f = 0.12$, $\lambda_p = 2$ and $T_p = 1$.

For the VoIP traffic under cross-layer analytical model, we took $A_p = 5$ and $A_s = 0.04$. In the case of M/G/1(m) traffic, we assumed that $A_p = 50$, $a = 135$ Erls, $s = 144$, $\lambda_1 = 1$, $\lambda_2 = 0.5$, $y_1 = 0.5$, $y_2 = 0.1$, $V_1 = 1.5$, $V_2 = 10$, $h_1 = 100$, $h_2 = 50$, $L_c = 53$, and $L_p = 48$. For VoIP traffic under IEEE 8012.16e/m, we took the number of users, $M = 125$, $a = 0.03$, $P_d = 0.85$, and $m = 6$. The steady probability state vector $\mathbf{\Pi}$ obtained from the $\mathbf{P}$ matrix of the size 24×24 for three different traffic models of Section II is plotted in Fig. 3. The profile of the probability states resembles to Poisson's pdf in a repetitive manner in three different regions of channels. Among the three cases, the IEEE 802.16e/m traffic model shows the maximum variation of states.

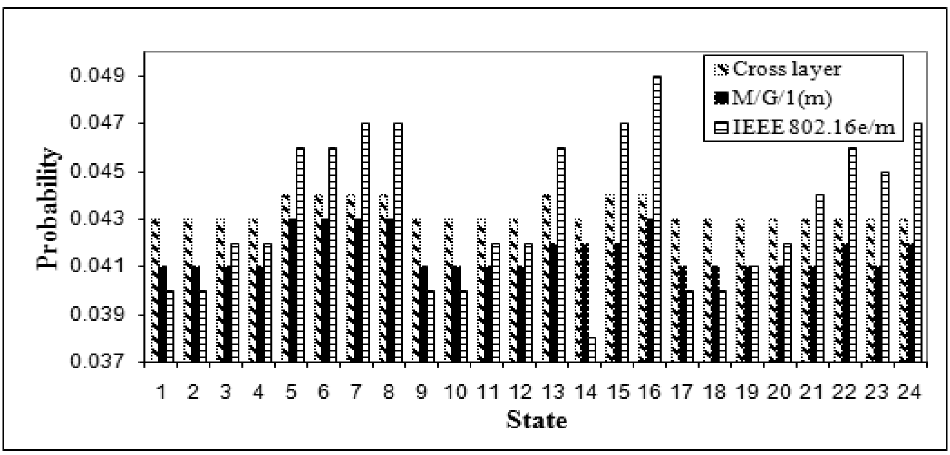

Fig. 3. The steady probability states of three traffic models

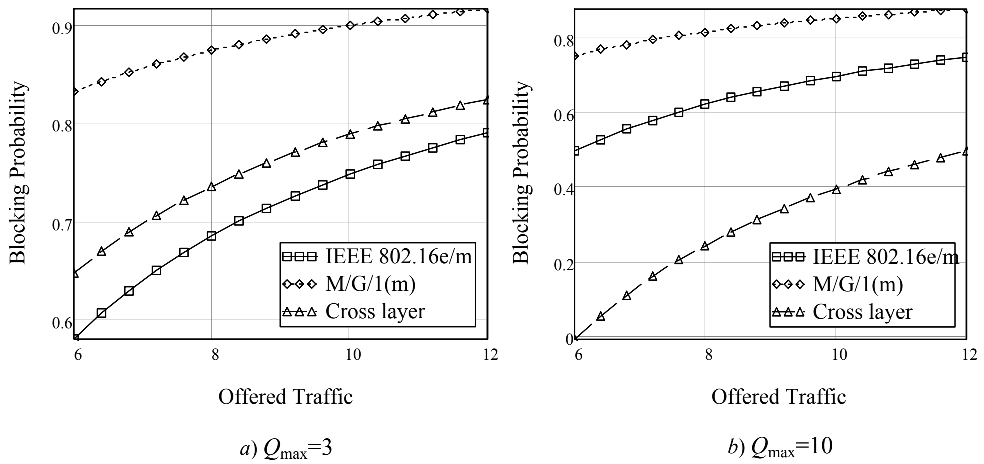

Fig. 4. Variation of blocking probability against offered traffic: a) $Q_{max} = 3$, b) $Q_{max} = 10$

Considering the above parameters, the variation of the blocking probability against offered traffic is plotted in Fig. 4(a). The IEEE 802.16e/m model reveals the best performance that satisfies the basic concept of packet traffic. A similar analysis is also shown in Fig. 4(b) considering $Q_{max} = 10$, which can support more offered traffic, as visualized in Fig. 4(b). The performance of the cross-layer analytical model has improved with a wider margin.

Finally, considering $Q_{max} = 3$ the throughput is plotted against the length of the PDU for three cases in Fig. 5(a). Among these three cases, throughput is found maximum for the IEEE 802.16e/m case. Fig. 5(b) shows the profile for $Q_{max} = 10$, where the throughput is increased significantly for all of the traffic cases with an increment in the queue length.

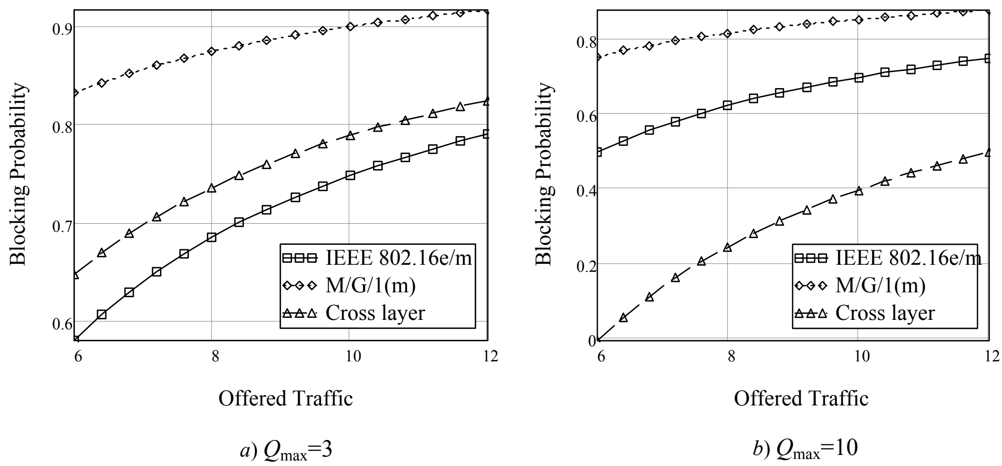

Fig. 5. Variation of the throughput against the length of PDU: a) $Q_{max} = 3$, b) $Q_{max} = 10$

4. CONCLUSION

To meet the demand of wireless voice services, more dynamic and flexible technologies need to be designed to ensure the efficient utilization of spectrum resources. One such system model can be designed based on the DTMC traffic model to analyze the VoIP traffic on the CR system. In this paper, we have analyzed the performance of VoIP services using three different traffic models in terms of throughput, steady probability states, and blocking probability. By comparing the VoIP performance under the cross-layer analytical model, M/G/1(m), and the IEEE802.16e/m traffic model, we showed that the IEEE802.16e/m traffic model outperforms other cases in terms of blocking probability and throughput when we take $Q_{max} = 3$ and the cross-layer model yields the best if we consider $Q_{max} = 10$. For all three cases throughput is increased with the increment of the queue length. Therefore, we can conclude that the relative performance of the three traffic models is sensitive to the length of the queue (i.e., Q_{max}). It should be mentioned here that the proper detection of PU and the minimization of false alarms, which is another aspect of the CR system and that is related to the fading condition of the wireless channel and the detection techniques, is beyond the scope of this paper.

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