Performance Evaluation of Voice-Data Integrated Traffic in IEEE 802.11 and IEEE 802.16e WLAN

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Abstract: - With the advent of 4G mobile communication system the traffic of wired and wireless network becomes voice/video- data integrated service. In this paper traffic model of Markov Modulated Poisson Process (MMPP) is applied for bursty traffic of Wireless LAN (WLAN), especially in IEEE 802.11 WLAN, known as WiFi, and IEEE 802.16e WLAN is known as WiMAX. Traffic of both networks is heavily affected by the fading of wireless channel. The traffic parameters of IEEE 802.11 WLAN are evaluated using Giuseppe Bianchi state transition chain. The profile of probability of idle condition, the probability of one successful transmission and the probability of collision is shown against the number of users. The packet blocking probability and throughput of WLAN are observed varying packet arrival rate. In this paper, a mathematical model of VoIP traffic over wireless channel under IEEE 802.16e WLAN is also analyzed under Rayleigh and Nakagami-m fading cases with the help of MMPP and discrete time Markov chain (DTMC) model. Finally, mean delay of both wired and wireless LAN are compared.

Key-Words: - Voice-data integrated service, MMPP, DTMC, Rayleigh fading, Nakagami-m fading, VoIP.

1 Introduction

To cope with the constant growth of user demand, service of a voice-data network is growing rapidly. To combat network congestion, limited number of available resource must be allocated among users in an optimum way. Traffic is the flow of information/messages through a communication system. The simplest way of evaluating the performance of a network is the use of state transition chain of birth-death process called Markov chain. One of the major drawbacks of Markov chain lies in the incorporation of large number of probability states which complicates the analysis of the traffic parameters of a network. Markov arrival process (MAP) provides equivalent state transition chain of few probability states with some assumption. MAP can be defined as a process 0 on the state space (N(t), J(t)) for $t \ge$ $\{(i, j); i \ge 0, 1 \le j \le m\}$, where N(t) is a counting process of "arrivals", indicates the number of arrival in (0, t] and J(t) is a Markov process with a finite state space, $1 \le j(t) \le n$ of the underlying Markov Chain. Teletraffic engineering adopts three most widely used cases of MAP and these are [1-3]: PH Markov renewal process (PH-MRP), Markov Modulated Poisson Process (MMPP) and Batch Markovian Arrival Process (BMAP). In this paper we adopt MMPP model in the analysis of bursty traffic of voice.

Voice-data integrated traffic is widely used in both wired and wireless networks. The traffic of packet switching network is bursty in nature. Hence they are analyzed based on MMPP. Again in M/D/1/K model (suitable for ATM traffic), the probability density function (pdf) of arrival of packet is exponential but that of service time is deterministic, where K is the length of queue. Therefore, the combination of MMPP and traffic of deterministic service time can be used to support voice/video- data integrated traffic. In wired communication, simple ON-OFF traffic model of [4] can be used quite comfortably to evaluate performance of such networks since wired network is not affected by any type of fading [5-6] like wireless network.

In case of wireless networks, the traffic performance is evaluated incorporating small scale

fading of wireless channel. Small scale fading or simply fading takes place in wireless environment where multi-path propagation occurs due to unguided nature of the channel. Fading is the rapid fluctuations of the amplitudes, phases or multi-path delays of a radio signal over a short period of time or travel distance. In this paper we consider Rayleigh and Nakagami-m fading [7-8] in both IEEE 802.11 and IEEE 802.16 standard networks. IEEE 802.11 standard (known as WiFi) is widely used as wireless local area network. Recent literatures find the performance of such networks using state transition diagram of Back off window. Such an analysis is found in [9-11]. In all of the papers, the state transition chain is solved to determine the probability of one successful transmission, probability of collision and packet transmission probability. The papers show the profile of packet delay and throughput against utilization factor of a channel.

With the development of 3G mobile communication system and beyond, Voice over Internet Protocol (VoIP) gets importance over wireless channel. The different protocols are found for VoIP services over fading channels. In [11], the authors proposed the session initiated protocol (SIP); where seven sessions were included to set up a transmission control protocol (TCP) connection. The condition of the channel is modeled using 2state Markov Chain. Finally, the authors show the profile of session set up delay against frame error rate. Initially IEEE 802.16 standard was developed to support high speed data communication over wireless channel based on MAC protocol. The standard is now upgraded to IEEE 802.16e to support mobile wireless service. The performance of an uplink VoIP system is proposed in [12] using two-dimensional Discrete Time Markov Chain (DTMC). In that paper the authors derived a steady state transition matrix from the two-dimensional DTMC. Finally, packet dropping probability is evaluated against the number of users. Similar analysis is also found in [13]. In [14] and [15], performance of both uplink and downlink of IEEE 802.16e are analyzed using different MCS (Modulation and Coding Schemes) levels.

The paper is organized as follows: Sec. II provides the complete analysis of IEEE 802.11 traffic under Binary Backoff algorithm, Sec. III derives the VoIP traffic parameters using IEEE 802.16e mode under wireless fading condition, while Sec. IV provides the results pertinent to Sec. II and Sec. III. Finally, Sec. V concludes the entire analysis.

2 IEEE 802.11 Traffic

The traffic parameters of IEEE 802.11 WLAN are evaluated using Giuseppe Bianchi state transition chain under Binary Backoff algorithm [16]. In this section the Binary Backoff algorithm and the corresponding traffic parameters are derived based on the twodimensional Markov chain of Giuseppe Bianchi's model shown in [16] and [17].

2.1 Binary Backoff algorithm

To reduce the packet dropping probability or to enhance throughput of wireless LAN, exponential binary backoff algorithm is widely used. The access method of MAC protocol of IEEE 802.11 based on exponential binary backoff algorithm can be explained with the following steps.

Step 1: The transmitting node first senses the status of the channel. If the channel is found busy then the *Tx* node continues to monitor the channel.

Step 2: If the channel is found idle for a fixed duration known as DIFS (Distributed Inter-frame Space), the Tx chooses a random number according to the binary exponential back off algorithm. The random number is used as a back off timer.

Step 3: Time immediately after the DIFS is slotted known as idle slots, where the duration of a slot is considered as the sum of the time required to sense a station and to switch the *Tx* from sensing / listening mode to transmitting mode.

Step 4: Elapsing of each idle slot the back off timer is decreased by one. If the channel is found busy before the back off timer reaches to zero then repeat the steps 1 to 3. The transmission of data begins only if the back off timer reaches to zero.

2.2 Two-Dimensional Markov Chain of Giuseppe Bianchi

For simplicity of analysis, let us consider the twodimensional Markov chain of Giuseppe Bianchi, where any state { x(t), y(t) } is defined as: y(t) be the size of the Backoff window and x(t) be the backoff stage of the station at time *t*. Figure 1 shows the state transition chain of the Backoff algorithm.



Fig. 1 Two dimensional Markov chain of backoff algorithm.

Solving the linear equations from nodes and application of normalization technique we get [9]:

$$P_{00} = \frac{2(1-2p)(1-p)}{(1-2p)(W+1) + pW(1-(2p)^{m})},$$
 (1)

where p is the probability that, in a time slot at least one of the N-1 remaining stations transmits, the Backoff stage of the station at time t may be (0, 1, 2, ...m) and W is the minimum size of Backoff window. The probability that a transmission occurs when the Backoff window is equal to zero, regardless of the Backoff stage, is found as

$$\tau = \sum_{j=0}^{m} P_{j,0}$$

$$= \frac{P_{00}}{1-p}$$

$$= \frac{2(1-2p)}{(1-2p)(W+1) + pW(1-(2p)^{m})},$$
(2)

where

$$p = 1 - (1 - \tau)^{N - 1}.$$
 (3)

For traffic analysis of IEEE 802.11 we need three important parameters: P_i , the probability that a channel found idle, P_s , the probability of successful

transmission and P_c , the probability of collision, i.e., when two more users transmit messages simultaneously.

Now, the probability of idle condition, one successful transmission and probability of collision are given respectively by $P_i = (1-\tau)^N$, $P_s = N\tau (1-\tau)^{N-1}$ and $P_c = 1-P_s - P_i$. To solve the probability τ let us define a function

$$f(\tau) = \frac{2(1-2p)}{(1-2p)(W+1) + pW(1-(2p)^m)} - \tau, \quad (4)$$

where the zero crossing point of above function gives the value of τ .

Six functions are plotted in Fig. 2 taking M = 5, W = 32, N = 20, 25, 30, 35, 40 and 45. The zero crossing points are 0.0265, 0.0235, 0.0205, 0.0195, 0.0175, 0.0165 found from Fig. 2.



Fig. 2 Profile of function $f(\tau)$.

2.3 Throughput and Delay of Packet Traffic

To evaluate the mean delay of packet and throughput, let us consider a 2-state MMPP mode as [18]:

$$\mathbf{Q} = \begin{bmatrix} -r_1 & r_1 \\ r_2 & -r_2 \end{bmatrix},$$

$$\Lambda = \begin{bmatrix} \lambda_1 & 0\\ 0 & \lambda_2 \end{bmatrix},$$
$$\lambda = \frac{r_2 \lambda_1 + r_1 \lambda_2}{r_1 + r_2}$$

and

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

The steady probability state is given by

$$\pi_j = \frac{1}{j!} Lt_{z \to 0} \frac{dj}{dz^j} b^* [\lambda(1-z)], \qquad (6)$$

where $b^*(x)$ is the pdf of service time. The steady state probability vector is then by using Eq. (6):

$$\theta = [\pi_0 \quad \pi_1 \quad \pi_2 \quad \dots \quad \pi_{k-1}]. \tag{7}$$

Let $E[T_B] = 0.2$ ms, which is the mean service time at the MAC layer. In [11], authors deal with IEEE 802.11 MAC protocol of WLAN to estimate the traffic parameters using three probability states: P_i (probability of idle state) P_s (probability of successful transmission and P_c (the probability of collision). In [10], authors determined throughput of IEEE 802.11 MAC, based on a MMPP/G/1/m queuing model. A DTMC is also proposed to determine the probability states. The probability states are evaluated like [28] as follows:

$$x_0 = \gamma^T \left\{ \pi_0 \left(\mathbf{\Lambda} - \mathbf{Q} \right)^{-1} \right\} E[T_B]^{-1}, \tag{8}$$

$$x_n = \gamma^T \left\{ \pi_n + \sum_{k=0}^{n-1} \pi_k \mathbf{U}^{n-1-k} \left(\mathbf{U} - \mathbf{I} \right) \right\}$$
(9)

×
$$(\mathbf{\Lambda} - \mathbf{Q})^{-1} E[T_B]^{-1}$$
;
where $n = 1, 2, ..., m$ -1, and

$$x_m = \theta - \sum_{n=0}^{m-1} x_n .$$
 (10)

The matrices Λ and \mathbf{Q} are given in [2] and the parameter γ is given by

$$\gamma = \frac{E[T_B]}{E[T_B] + \theta_0 (\mathbf{\Lambda} - \mathbf{Q})^{-1} \mathbf{e}},$$

where $E[T_B]$ is the mean values of size of packets and θ is the steady-state probability vector. All the above parameters along with the packet waiting time *W* and throughput *S* are found in [10, 15, 19]. In particular, *W* and *S* are given respectively by

$$\lambda (1 - P_B) W = \sum_{k=0}^{K} k X_k e \tag{11a}$$

and

$$S = \lambda (1 - P_B) (1 - P_c^{m+1}).$$
 (11b)

3 VoIP and Traffic Using IEEE 802.16e Model

The main objective of wireless communication is to provide high data rate with guaranteed quality of service (QoS). The throughput prediction for Mobile WiMAX is done based on IEEE Std 802.16-2009. With the advent of OFDMA based IEEE 802.16e, research is now going on to implement VoIP service with adaptive modulation and channel coding scheme. To enhance the throughput of the wireless system the modulation and coding scheme of the transmitter is changed according to the fading condition of the channel. Such dynamic modulation and coding scheme is called MCS (Modulation Coding Scheme). Therefore, the service becomes a variable bit rate service where the bit rate depends on the fading of the wireless channel. VoIP traffic over wireless channel requires the same QoS as the constant bit rate service of wired network.

To resolve the situation mentioned above, many packet scheduling algorithms are proposed [20, 21]. Recent literature shows the VoIP traffic model based on the combination of two-dimensional DTMC and MMPP. Such model is applicable for wireless VoIP services, cognitive radio system and IEEE 802.16 OFDMA/WiMax traffic. The traffic models of above services are depicted in [22, 23]. The on-off traffic model of VoIP considering exponential distribution of mean value of on and off periods are summarized in [19, 22]. The voice traffic of exponentially distributed on-off system is also analyzed using one-dimensional Markov chain of limited user traffic model in [20] the chain is extended with incorporation of queue in [21]. The four transition rates and formation of transition rate matrix and arrival rate matrix of MMPP are explained in [2]. The down link packet scheduling technique of VoIP (using MMPP and probability of using MCSs levels) is analyzed in [24], the similar job is enhanced for both up and down link in [15]. The MCSs levels are shown in tabular form in [25, 26] with derivation of probability of choosing a particular MCS level.

3.1 IEEE 802.16e Based on MCS Level

The probability of a VoIP packet being modulated with m^{th} MCS level in the uplink is expressed as

$$P_m = \int_{\gamma_m}^{\gamma_{m+1}} P_{\gamma}(\gamma) \, d\gamma \,, \tag{12}$$

where $P_{\gamma}(\gamma)$ is the pdf of fading. For Nakamagi-m fading,

$$P_{\gamma}(\gamma) = \frac{1}{\Gamma(m_f)} \frac{m_f^{m_f} \gamma^{m_f-1}}{\gamma_{av}^{m_f}} e^{\frac{-m_f \gamma}{\gamma_{av}}}.$$
 (13)

where m_f is the Nakagami fading parameter, γ_{av} is the average SNR and γ_m is the minimum SNR that can support MCS level *m*.

In this paper we consider a different approach. Let us calculate the symbol error rate P_s using the relation

$$P_{s}(\gamma_{m}) = \int_{\gamma_{m}}^{\gamma_{m+1}} P_{\gamma}(\gamma) Q(a,b) d\gamma, \qquad (14)$$

where Q(a, b) is Gaussian Q-function and the parameters *a* and *b* are selected according to the particular modulation scheme. Considering Rayleigh fading and $\gamma_{av} = 12$ dB, the probability of symbol error rate of QPSK, 16-QAM and 64-QAM are found as in Fig. 3. If the threshold symbol error rate (SER) is 0.01, then the minimum SNR (γ_m) that can be supported by QPSK, 16-QAM and 64-QAM are 3, 8 and 11dB respectively. If γ_m falls 2dB below the threshold value of a particular modulation scheme, then coding rate will be increased by 2 times.



Fig. 3 SER under fading condition.

If a base station (BS) is scheduling b VoIP packets from uplink queue then its probability is expressed as

$$P_{s}^{u}(b) = \Pr\left\{X_{b} \in \psi_{b} \text{ and } X_{b+1} \notin \psi_{b+1}\right\}$$
$$= \sum_{\forall X_{b} \in \psi_{b}} \left(b! \prod_{m=1}^{M} \frac{p_{m}^{x_{m}}}{x_{m}!}\right) \left(1 - \sum_{m=1}^{M} p_{m} I_{m}(X_{b})\right),$$
(15)

where *b* is the maximum number of packets in the uplink schedule within the range of the bandwidth, ψ_b is the possible set of scheduling for successful transmission, X_b is a member of ψ_b and the index probability $I_m(X_b)$ is given by

$$I_m(X_b) = \begin{cases} 1, & \text{if } (X_b + \Delta) \in \psi_b + 1\\ 0, & \text{otherwise.} \end{cases}$$

The complete explanation of the above parameters is available in [15, 24].

Let us now consider an Algorithm for determination of the index probability $I_m(X_b)$: Select *M*, a positive integers (positive no of MCS level) and b = 1 as the minimum value.

Select the vector,

$$X_b = (x_1, x_2, x_3, \dots, x_M);$$

 $X_b \in \{\text{positive integer}\}$ such that

$$X_b \in \{\text{positive integer}\} \text{ such that } \sum_{i=1}^M x_i = b.$$

If $\sum_{i=1}^M x_i l_i \le N_{slot,u}$ and $\sum_{i=1}^M x_i l_i + l_j \le N_{slot,u}$,

then $I_i(X_b) = 0$ else $I_i(X_b) = 1$.

Repeat steps 1 and 2 for j = 1, 2... M.

Set b = b+1 and repeat steps 2 to 4 to evaluate all possible $I_i(X_{b+1})$.

Now the transition matrix of the uplink queue is

$$\mathbf{P} = \begin{vmatrix} p_{0,0} & p_{0,1} & \cdots & p_{0,K_u} \\ p_{1,0} & p_{1,1} & \cdots & p_{1,K_u} \\ \cdots & \cdots & \cdots & \cdots \\ \mathbf{P}_{W_u,0} & \mathbf{P}_{W_u,1} & \cdots & \mathbf{P}_{W_u,W_u} \end{vmatrix} .$$
(16)

 $\begin{bmatrix} P_{K_u}, 0 & P_{K_u}, 1 & \cdots & P_{K_u}, K_u \end{bmatrix}$ Elements of the matrix **P** is expressed as

$$p_{i,j} = \sum_{b=b_1}^{b_2} \mathbf{UD}(j - \max(i - b, 0)P_S^u(b)), \quad (17)$$

where $\mathbf{U} = (\mathbf{\Lambda} - \mathbf{R})^{-1} \mathbf{\Lambda}$, and **D** is the diagonal probability matrix and is given by

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

 T_f is the duration of a frame, **R** and **A** are the transition rate matrix and arrival rate matrix of MMPP respectively.

The steady probability vector $\boldsymbol{\pi}$ can be found from,

$$\boldsymbol{\pi}.\mathbf{P} = \boldsymbol{\pi} \text{ and } \boldsymbol{\pi}.\mathbf{1} = 1. \tag{18}$$

Two numerical examples of this section are given in Sec. 4.

3.2 IEEE 802.16 e/m using AMR Speech Coding

In VoIP traffic, there is always a tradeoff between efficient utilization of radio resources and QoS. In IEEE 802.16e model, different algorithms are prevalent for scheduling of resources to support in talk-spurt and silence period of a user, as summarized in [20]. In [21], a VoIP scheduling algorithm is proposed to enhance the performance of a WLAN. According to the algorithm, the BS allocates small bandwidth as a grand in a random access manner instead of periodically allocated grand. The random access scheme continued till the instant of change of silent to talk-spurt state. Once the state of the user is talk-spurt, the BS grants wide band in periodic manner as shown in Fig. 4 and the corresponding scheduling can be modeled by a onedimensional Markov chain of Fig. 5.

Here we consider the voice traffic is exponentially distributed with mean On time of $1/\lambda$ and mean Off time of $1/\mu$. Let the capacity of the link is *m* and the number of user is *N*. The capacity of the link is expressed as [21]:

$$m = \frac{T_{GI}}{T_{MF}} \frac{S_{TOT}}{\overline{S_{scheduler}}},$$
(19)

where, T_{GI} is the time separation between two frames during talk-spurt, T_{MF} is the duration of the frame, S_{TOT} is the total number of time slots in the frame and $S_{scheduler}$ is the average number of time slots required during grand interval. Taking the length of queue k, we derive the initial state P_0 , probability of entering queue Q(a, N, m, k), blocking probability B(a, N, m, k), and the steady probability states. The initial state probability is

$$P_{0} = \left[\sum_{x=0}^{m} \binom{N}{x} a^{x} + (N-m)! \binom{N}{m} a^{m} \sum_{y=1}^{N} \frac{\left(\frac{a}{m}\right)^{y}}{(N-m-y+2)!}\right]^{-1},$$
(20)

The probability of entering queue and the blocking probability are given respectively by

$$Q(a, N, m, k) = \sum_{y=1}^{k} \frac{(N-m)!}{(N-m-k)!} {\binom{N}{m}} {\binom{a}{m}}^{y} P_0 a^{m},$$
(21)



Fig. 4 VoIP scheduling algorithm of [25].



Fig. 5 Markov chain of VoIP users for on-off system case.

and

$$B(a, N, m, k) = \frac{(N-m)!}{(N-m-k)!} {\binom{N}{m}} {\binom{a}{m}}^{y} P_0 a^{m}.$$
(22)

Steady probability states are given by the following formula,

$$P_{x} = \begin{cases} P_{0} \sum_{x=0}^{m} {N \choose x} a^{x}, & 0 \le x \le m; \\ P_{0} {N \choose m} (N-m)! a^{m} \sum_{x=1}^{N} \frac{(a/m)^{x}}{(N-m-x+2)!}, & m < x \le N. \end{cases}$$
(23)

The average access delay is expressed as

$$\overline{D} = T_{MF} \times \sum_{k=1}^{\infty} k \times p_N \left[k \le m \right] \times p_N \left[k > m \right]^{k-1}.$$
(24)

4 Results and Discussions

Please, follow our instructions faithfully, otherwise you have to resubmit your full paper. This will enable us to maintain uniformity in the journal. Thank you for your cooperation and contribution.

First of all let us concentrate on Backoff algorithm of IEEE 802.11. Taking the number of users, N = 20, 25, 30, 35, 40 and 45 and we get $\tau = 0.0265 \ 0.0235 \ 0.0205 \ 0.0195 \ 0.0175$ and 0.0165. The probability of idle condition P_i , the probability of one successful transmission P_s , and the probability of collision P_c are plotted against the number of users N in Fig. 6.

The variation of the parameters are not so rapid hence to observe the distinct variations we vary the number of users widely and these curves are again plotted in Fig. 7. For both Fig. 6 and Fig. 7, we take the maximum Backoff stage m = 5, the minimum contention window W = 32.

Figure 7 reveals that the probability of collision P_c and the probability of successful transmission P_s increase with the number of users but the probability of idle condition P_i decreases with the number of users. For $N \ge 80$ both P_i and P_s are almost constant but P_c is still in rising condition.

The number of collisions increases with the number of users which combat any increment of throughput of the network. Similar explanation will also be found from Fig. 8.



Fig. 6 Variation of *Pi*, *Pc* and *Ps* against the number of users.

Taking the maximum Backoff stage m = 5, the minimum contention window W = 32, number of users N = 40 and using the expression of packet blocking probability P_B and throughput S of Eq.

(11), we get the graph of Fig. 8. Both parameters are plotted against arrival rate of packet. Both the blocking probability and the throughput increase with the arrival rate of the packets. Initially, the rate of increment of throughput is rapid but the rate decreases for larger packet arrival rate, and finally the throughput becomes constant. The above observation can be explained with the phenomenon of collisions of packets at higher arrival rate. Similar profile is also found for IEEE 802.16 e/m using AMR Speech Coding case as shown in Fig. 9, where we have taken N = 20, K = 12 and m = 6.



Fig. 7 Variation of P_i , P_c and P_s against the number of users.



Fig. 8 variation of P_B and S against packet arrival rate IEEE 802.11.



Fig. 9 variation of P_B and S against packet arrival rate of IEEE 802.16e/m.

For the case of IEEE 802.16e, let us consider an example, where the types of MCS level, M = 7, and the number of available slots, $N_{slot, u} = 60$. Therefore,

$$X_b = (x_1, x_2, x_3, x_4, x_5, x_6, x_7),$$

where

$$b = \sum_{i=1}^{M=7} x_i = 10$$

and x_i is the number of packets of ith MCS level.

Now, let us consider b = 10 packets with MCS distribution of

$$X_{h} = (0, 0, 3, 1, 1, 2, 3)$$
.

Therefore,

$$b = \sum_{i=1}^{M=7} x_i = 0 + 0 + 3 + 1 + 1 + 2 + 3 = 10.$$

We know, $l_1 = 36$, $l_2 = 24$, $l_3 = 12$, $l_4 = 6$, $l_5 = 4$, $l_6 = 3$, $l_7 = 2$; where the unit of l_i is time slots. The number of slots requires transmitting the frame X_{10} :

$$\begin{split} \sum_{m=1}^{M} x_m l_m &= (0 \times 36) + (0 \times 24) + (3 \times 12) + (1 \times 6) + (1 \times 4) \\ &+ (2 \times 3) + (3 \times 2) = 58 \leq N_{slot,u} \;. \end{split}$$

If the packets of 7^{th} MCS level increased by 1 then we have

$$X_{b+1} = (0, 0, 3, 1, 1, 2, 4).$$

Now the required slots:

$$\sum_{m=1}^{M} x_m l_m = (0 \times 36) + (0 \times 24) + (3 \times 12) + (1 \times 6) + (1 \times 4)$$

 $+(2 \times 3) + (4 \times 2) = 60 \le N_{slot,u}$.

Therefore,

$$I_7(X_b) = I_7(X_{10}) = 1.$$

If the packet of 6th MCS level increased by 1, then

$$\sum_{m=1}^{M} x_m l_m = 61 > N_{slot,u}.$$

Therefore, we have

$$I_6(X_b) = I_6(X_{10}) = 0.$$

If all the packets are 1^{st} MCS level then $b_1 = 1$, which is the minimum number of packets can be sent. If all the packets are 7^{th} MCS level, then $b_2 =$ 30, i.e. the maximum 30 packets can be sent.

Let us now consider another example of finding the value of function $I_j(X_b)$ considering, M = 4, $N_{slot, u} = 50$. The analysis is shown in tabular form in

Table 1: Evaluation of function $I_i(X_b)$

b	X _b	$I_j \equiv I_j(X_b)$
	$X_1 = (1,0,0,0)$	$I_1 = I_2 = 0, I_3 = I_4 = 1$
	$X_1 = (0,1,0,0)$	$I_1 = 0, I_2 = I_3 = I_4 = 0$
<i>b</i> = 1	$X_1 = (0,0,1,0)$	$I_1 = I_2 = I_3 = I_4 = 1$
	$X_1 = (0,0,0,1)$	$I_1 = I_2 = I_3 = I_4 = 1$
	$X_2 = (1,0,1,0)$	$I_1 = I_2 = I_3 = I_4 = 0$
<i>b</i> = 2	$X_2 = (1,0,0,1)$	$I_1 = I_2 = I_3 = 0, I_4 = 1$
	$X_2 = (0,1,1,0)$	$I_1 = I_2 = 0, \ I_3 = I_4 = 1$
	$X_2 = (0,1,0,1)$	$I_1 = I_2 = 0, \ I_3 = I_4 = 1$
	$X_2 = (0,0,1,1)$	$I_1 = 0, \ I_2 = I_3 = I_4 = 1$
	$X_2 = (0,0,0,2)$	$I_1 = I_2 = I_3 = I_4 = 1$

		r
	$X_3 = (1,0,0,2)$	$I_1 = I_2 = I_3 = I_4 = 0$
<i>b</i> = 3	$X_3 = (0,1,1,1)$	$I_1 = I_2 = I_3 = 0, I_4 = 1$
	$X_3 = (0,1,0,2)$	$I_1 = I_2 = 0, \ I_3 = I_4 = 1$
	$X_3 = (0,0,1,2)$	$I_1 = 0, \ I_2 = I_3 = I_4 = 1$
	$X_3 = (0,0,0,3)$	$I_1 = 0, \ I_2 = I_3 = I_4 = 1$
	$X_4 = (0,1,0,3)$	$I_1 = I_2 = I_3 = 0, I_4 = 1$
	$X_4 = (0,1,1,2)$	$I_1 = I_2 = I_3 = I_4 = 0$
	$X_4 = (0,0,1,3)$	$I_1 = I_2 = 0, I_3 = I_4 = 1$
	$X_4 = (0,0,2,2)$	$I_1 = I_2 = 0, I_3 = I_4 = 1$
<i>b</i> = 4	$X_4 = (0,0,3,1)$	$I_1 = I_2 = I_3 = 0, I_4 = 1$
	$X_4 = (0,0,4,0)$	$I_1 = I_2 = I_3 = I_4 = 0$
	$X_4 = (0,0,0,4)$	$I_1 = 0, \ I_2 = I_3 = I_4 = 1$
	$X_4 = (0,1,0,3)$	$I_1 = I_2 = I_3 = 0, I_4 = 1$
	$X_5 = (0,1,0,4)$	$I_1 = I_2 = I_3 = I_4 = 0$
	$X_5 = (0,0,1,4)$	$I_1 = I_2 = 0, I_3 = I_4 = 1$
<i>b</i> = 5	$X_5 = (0,0,2,3)$	$I_1 = I_2 = I_3 = 0, I_4 = 1$
	$X_5 = (0,0,3,2)$	$I_1 = I_2 = I_3 = I_4 = 0$
	$X_5 = (0,0,0,5)$	$I_1 = I_2 = 0, I_3 = I_4 = 1$
	$X_6 = (0,0,1,5)$	$I_1 = I_2 = I_3 = I_4 = 0$
	$X_6 = (0,0,2,4)$	$I_1 = I_2 = I_3 = I_4 = 0$
<i>b</i> = 6	$X_6 = (0,0,0,6)$	$I_1 = I_2 = 0, I_3 = I_4 = 1$
<i>b</i> = 7	$X_7 = (0,0,0,7)$	$I_1 = I_2 = I_3 = 0, I_4 = 1$
<i>b</i> = 8	$X_8 = (0,0,0,8)$	$I_1 = I_2 = I_3 = I_4 = 0$

Taking Number of users N = 50, $T_f = 1$ ms, m = 4, $\lambda_1 = 0.566$, $\lambda_2 = 0.842$, $r_1 = 0.21$, $r_2 = 0.12$, the transition matrix of VoIP traffic under Rayleigh fading case is found like given in Table 2:

Similar transition matrices are also derived for Nakagami-2 and Nakagami-4 fading cases. The steady state vector derived from transition matrix \mathbf{P} , is plotted for two fading cases as shown in Fig. 10. The impact of fading on the probability sate is found very small but the impact will be high on blocking probability, probability of delay and throughput.

In this paper we consider the delay experience by the traffic of (a) MMPP+M/D/1 Traffic Model in video-data integrated service under ATM system of wired network (immune of fading), (b) IEEE 802.11 traffic of Sec. II, (c) VoIP Traffic Using IEEE 802.16e Model of Sec. III. The mean delay vs. utilization factor of all cases is depicted in Fig. 11.



Fig.10 The steady probability vector under Rayleigh and Nakagami-m fading.

For the cases (b) and (c), the traffic parameters are: number of users N = 50, $T_f = 5$ ms, E[TB] = 10ms, $\lambda = 0.02$ Erls/user, VoIP capacity, m = 20, $\lambda_I = 0.58$, $\lambda_2 = 0.04$, $r_I = 0.1$, $r_2 = 0.2$, pdf of service time of MMPP/E2/1 case is

$$b(\theta) = (k\mu/(\theta + k\mu))^k, \quad k = 2.$$

Let us now go for the combining scheme of M/D/1 and MMPP/D/1 of (a) known as MMPP+M/D/1 applicable to voice data integrated service through ATM network. The detailed theoretical analysis of the model is given in [18, 27]. For convenience of the readers the analysis is included in the Appendix A. Let, $h = 2.718 \times 10^{-6}$ s, $T = 3.397 \times 10^{-6}$ s, and n = 3, we get the MMPP traffic parameters in per ms as, $r_1 = 0.414$, $r_2 = 0.071$, $\lambda_{b1} = 3.657 \times 10^2$ cells/ms and $\lambda_{b2} = 6.239$ cells/ms. Therefore, we get $\lambda_b = (\lambda_{b1}r_2 + \lambda_{b2}r_1)/(r_1 + r_2) = 58.499$ cells/ms. Taking $\lambda_p = 100$ cells/ms we can evaluate: $\lambda_I = \lambda_{b1} + \lambda_p = 465.652$ cells/ms, $\lambda_2 = \lambda_{b2} + \lambda_p = 106.239$ cells/ms, $\lambda_t = (\lambda_1 r_2 + \lambda_2 r_1)/(r_1 + r_2) = 158.499$ cells/ms and $\rho = \lambda_t h = 0.431$ Erls. From the graphical solution of

$$f(z) = z - e^{-hW(z)}; z = 0.62.$$

The transition probabilities are ((A.13) and (A.14)):

$$P_{01} = \frac{W(z) - R_1(z) - r_2}{(\lambda_2 - \lambda_1)(1 - z)} (1 - \rho) = 2.93 \times 10^{-4},$$

$$P_{02} = 1 - \rho - P_{01} = 0.569;$$

where

$$u = \frac{\lambda_1 - \lambda_2}{(1 - \rho)(r_1 + r_2)^2} [r_1 P_{01} (1 - \lambda_2 h) - r_2 P_{02} (1 - \lambda_1 h)]$$

= 28.842,

where we have used Eq. (A.15).

Table 2: The 16×16 matrix P

P = [0.084]	0.019	0.055	0.016	0.018	6.887E-3	3.929E-3	1.933E-3	6.444E-4	4.069E-4	8.455E-5	6.852E-5	9.244E-6	9.616E-6	8.663E-7	1	.157E-6;	
0.01	0.08	6.834E-3	0.067	2.242E-3	0.028	4.902E-4	7.971E-3	8.039E-5	1.678E-3	1.055E-5	2.826E-4	1.153E-6	3.965E-5	1.081E-7	1.081E-7 4.77E-6;		
0.084	0.019	6.834E-3	0.067	2.242E-3	0.028	4.902E-4	7.971E-3	8.039E-5	1.678E-3	1.055E-5	2.826E-4	1.153E-6	3.965E-5	8.663E-7	8.663E-7 1.157E-6		
0.01	0.08	6.834E-3	0.067	2.242E-3	0.028	4.902E-4	7.971E-3	8.039E-5	1.678E-3	1.055E-5	2.826E-4	1.153E-6	3.965E-5	1.081E-7	081E-7 4.77E-6;		
0	0	0.078	0.019	0.047	0.014	0.015	5.858E-3	3.247E-3	1.616E-3	5.281E-4	3.366E-4	6.891E-5	5.628E-5	7.504E-6	7.504E-6 7.858E-6;		
0	0	9.673E-3	0.077	5.88E-3	0.059	1.878E-3	0.024	4.05E-4	6.664E-3	6.588E-5	1.388E-3	8.595E-6	2.321E-	4 9.36E-'	9.36E-7 3.241J		
0	0	0	0	0.077	0.019	0.047	0.014	0.015	5.826E-3	3.235E-3	1.609E-3	5.265E-4	3.353E-4	6.873E-5	6.873E-5 5		
0	0	0	0	9.576E-3	0.077	5.843E-3	0.059	1.869E-3	0.024	4.035E-4	6.634E-3	6.568E-5	1.383E-	3 8.573E	8.573E-6 2.314E-4-		
0	0	0	0	0	0	0.077	0.019	0.047	0.014	0.015	5.824E-3	3.235E-3 1.609E-3 5.265E-4		3.	353E-4;		
0	0	0	0	0	0	9.57E-3	0.077	5.841E-3	0.059	1.869E-3	0.024	4.035E-4	6.633E-3	6.567E-5	6.567E-5 1.383E-		
0	0	0	0	0	0	0	0	0.077	0.019	0.047	0.014	0.015	5.824E-3	3.235E-3	1.609E-3;		
0	0	0	0	0	0	0	0	9.57E-3	0.077	5.841E-3	0.059	1.869E-3	0.024	4.035E-	4 6.	6.633E-3;	
0	0	0	0	0	0	0	0	0	0	0.077	0.019	0.047	0.014	0.015	5.82	24E-3;	
0	0	0	0	0	0	0	0	0	0	9.5	7E-3 0.07	7 5.	.841E-3 0	.059	1.869E-3	0.024;	
0	0	0	0	0	0	0	0	0	0	0	0	C	0.077	0.019	0.047	0.014;	
1	1	1	1	1	1	1	1	1	1	1	1	1		1	1	11:	

Now the mean, virtual and actual waiting time ((A.6) to (A.8)) are $W_M = \lambda_t h^2/2(1-\rho) = 1.029$ E-3 ms, $W_a = W_M + uh/\rho(1-\rho) = 0.321$ ms and $W_v = W_M + uh/(1-\rho) = 0.139$ ms. The W_j ((A.15)) are $W_1 = P_1$ { $W_v + u(\lambda_1\lambda_t)/G$ } = 0.1ms and $W_2 = P_2$ { $W_v + u(\lambda_2, \lambda_t)/G$ } = 0.038ms. The individual waiting time are: for video, $W_b = (W_1 \lambda_{b1+} W_2 \lambda_{b2})/\lambda_b = 0.632$ ms for data $W_P = W_1 + W_2 = 0.139$ ms. We are able to plot the delay of MMPP+M/D/1 traffic against utilization factor.

The profile of delay of data and video traffic of wired LAN of MMPP+M/D/1 case are almost parallel but the delay of video traffic shows longer delay compared to that of data traffic because of bursty nature of arrival and variable length of packets of the video traffic.

The profile of IEEE 802.11 shows maximum delay compared to wired LAN case because of huge collision experienced by Backoff algorithm but the rate of increment of delay with utilization factor is found very slow since users are stationary and less affected by fading under heavy traffic condition. In case of IEEE 802.16e, delay is not so large like IEEE 802.11 case, because of selection of proper modulation technique of MCS levels. The rate of increment of delay of both the IEEE 802.16e cases is very rapid because users are mobile and heavily affected by fading under heavy traffic condition.



Fig. 11 Comparison of mean delay of different LAN.

5 Conclusion

The paper compares QoS, throughput and mean waiting time of traffic of IEEE 802.11 and IEEE 802.16e WLAN to observe the impact of fading on the networks. The expected result should be like that: IEEE 802.16e network is more affected under small scale fading condition since it supports mobility of users. The result section, Sec. IV, reveals a complete different scenario because of the use of proper MCS level in IEEE 802.16e network, hence throughput is increased because of small BER. In case of IEEE 802.11 traffic, probability of collision deteriorates the performance as explained in Sec. IV. The performance of video and data integrated traffic of wired LAN is better than both of IEEE 802.11 and IEEE 802.16e because of immunity of fading. In this paper, we have considered only uplink traffic of wireless network but work can be extended for downlink case as well. Instead of binary Backoff algorithm of IEEE 802.11 the MCS level can also be applied for it to observe its level of improvement. We have applied MMPP model for bursty traffic of voice but still we have the scope to observe the situation using other MAPs, like batch arrival process or phase type renewal process to support traffic of variable packet length.

Appendix A

When voice or video signals are sent in packetized form, it is modeled as ON-OFF pattern. In case of a single source, the spurt and silence period are assumed exponentially distributed with mean of α^{-1} and β^{-1} respectively. If the sampling period of voice/video is *T* (cells / packets are formed for a fixed duration *T* of the analog signal) then three statistical parameters of the traffic are: The packet arrival rate,

B

$$\lambda = \frac{\rho}{T(\alpha + \beta)},\tag{A.1}$$

the SCV of inter arrival time,

$$C_a^2 = \frac{1 - (1 - \alpha T)^2}{T^2 (\alpha + \beta)^2},$$
 (A.2)

skewness of service time,

$$S_{k} = \frac{2\alpha T \left(\alpha^{2} T^{2} - 3\alpha T + 3 \right)}{\left[\alpha T (2 - \alpha T) \right]^{3/2}}.$$
 (A.3)

The 2-phase MMPP parameters are determined as:

$$r_{i} = \begin{cases} D\left(1 + \frac{1}{\sqrt{1 + n\lambda E}}\right), & i = 1; \\ D\left(1 - \frac{1}{\sqrt{1 + n\lambda E}}\right), & i = 2. \end{cases}$$

$$\lambda_{i}' = \begin{cases} n\lambda + F + F\sqrt{1 + n\lambda E}, & i = 1; \\ n\lambda + F - F\sqrt{1 + n\lambda E}, & i = 2; \end{cases}$$
(A.4)
(A.5)

where

$$D = \frac{K_H \lambda_{H_1} + (1 - K_H) \lambda_{H_2} - \lambda}{C_a^2 - 1},$$

$$F = D. \frac{3C_a^4 - S_k C_a^3 - 3C_a^2 + 2}{3(C_a^2 - 1)},$$

and

$$E = \frac{K_H \lambda_{H_1} + (1 - K_H) \lambda_{H_2} - \lambda}{F^2}.$$

Equations (A.1)-(A.5) provide parameters of MMPP traffic. Again the virtual and actual waiting time of MMPP/G/1 model are respectively as

$$W_{\nu} = W_M + \frac{uh}{1-\rho}, \qquad (A.6)$$

and

$$W_a = W_M + \frac{uh}{\rho(1-\rho)}, \qquad (A.7)$$

where W_M is the mean waiting time of M/G/1 traffic expressed as

$$W_M = \frac{\lambda_t h}{\rho(1-\rho)}.$$
 (A.8)

The parameters u and λ_t are given in (A.15) and (A.9). Now for voice data integrated traffic of MMPP + M/D/1 case, let λ_p be the Poisson's arrival

rate of voice traffic. We have $\lambda_1 = \lambda_1' + \lambda_p$ and $\lambda_2 = \lambda_1' + \lambda_2$

$$\lambda_2 = \lambda_2 + \lambda_p \,.$$

To determine mean waiting time, let us determine,

$$\pi_1 = \frac{r_2}{r_1 + r_2}, \ \pi_2 = \frac{r_1}{r_1 + r_2}, \ \lambda_t = \frac{\lambda_1 r_2 + \lambda_2 r_1}{r_1 + r_2}.$$
(A.9)

Let us define a function,

$$W(z) = \frac{1}{2} \{\lambda_1(1-z) + r_1 + \lambda_2(1-z) + r_2\} + \frac{1}{2} \sqrt{(\lambda_1(1-z) + r_1 - \lambda_2(1-z) - r_2)^2 - 4r_1r_2} = \frac{1}{2} \{(1-z)(\lambda_1 + \lambda_2) + (r_1 + r_2)\} + \frac{1}{2} \sqrt{\{(1-z)(\lambda_1 - \lambda_2) + (r_1 - r_2)\}^2 - 4r_1r_2}.$$
(A.10)

For MMPP/ D/1 model,

$$z = b^{*}(W(z)) = \exp(-hW(z)),$$
 (A.11)

where *h* is the duration of a cell or packet, and W(z) is given by Eq. (A.10). By solving the above equation graphically, we can get the value of *z*. Let us evaluate the two probability of transitions P_{01} and P_{02} :

$$P_{01} = \frac{W(z) - R_1(z) - \gamma_2}{(\lambda_2 - \lambda_1)(1 - z)} (1 - \rho), \rho = \lambda_t h; \quad (12)$$

and

$$P_{02} = 1 - \rho - P_{01}, \ R_j(z) = \lambda_j(1 - z) + r_j; \ j = 1, 2.$$
(A.13)

The mean waiting time is given by the following expression

$$W_j(z) = \pi_j \left(W_v + u \cdot \frac{\lambda_j - \lambda_t}{G} \right), \quad j = 1, 2; \quad (A.14)$$

where

$$G = \sum_{j=1}^{2} \pi_{j} (\lambda_{j} - \lambda_{t})^{2}$$

and

$$u = \frac{\lambda_1 - \lambda_2}{(1 - \rho)(r_1 + r_2)^2} [r_1 P_{01}(1 - \lambda_2 h) - r_2 P_{02}(1 - \lambda_1 h)].$$

The mean waiting time of individual traffic is

$$W_{MMPP/D/1} = \frac{\lambda'_1 \ W_1 + \lambda'_2 W_2}{\lambda'_t},$$
 (A.15)

where

$$\lambda_t' = \frac{\lambda_1' r_2 + \lambda_2' r_1}{r_1 + r_2},$$

and

 $W_{voice} = W_1 + W_2$. (A.16)

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