AN ACCESS CONTROL PROTOCOL BASED ON ESTIMATION OF MULTIMEDIA TRAFFIC WITH AN ADAPTIVE ALGORITHM IN CDMA PACKET NETWORK

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Abstract – In multimedia wireless communications, various types of traffic are handled. Generally, each of these traffic has a different stochastic property. An access control protocol for multimedia wireless communications, therefore, are required to adopt a stochastic property of multimedia traffic. Conventional access control protocols, however, do not consider such characteristics of multimedia traffic. In this paper, we propose an access control protocol for multimedia CDMA packet communications. The proposed protocol controls the data packet transmissions based on the estimation of the instantaneous number of simultaneously transmitted voice and video packets. This estimation exploits the stochastic property of multimedia traffic. To carry out such estimation, we employ an adaptive algorithm as an estimation method. We evaluate the performance of the proposed protocol by comparing it with the performance in the case where a stochastic property of traffic is not considered.

Keywords – CDMA packet communications, multimedia traffic, access control protocol, adaptive algorithm.

I. INTRODUCTION

With the development of wireless communication technique, various types of traffic, such as voice, video, and data, have been handled in wireless communication networks. Generally, each traffic of these media has a different stochastic property and an inherent requirement for quality of service. In order to adopt such a property and satisfy such a requirement, it is important to design an effective access control protocol for multimedia traffic[1].

A code division multiple access (CDMA) has become more popular for wireless communication networks. Various access control protocols for CDMA packet communication systems have been proposed in the previous works [2]–[6]. In a CDMA packet communication, fluctuation of the interference among packets which are transmitted simultaneously may affect the performance of the system. Access control protocols for CDMA packet communications therefore have to not only handle various types of traffic but also control variation of the interference among packets.

Conventional access control protocols proposed in [2], [3] control fluctuation of packets based on the average number of transmitted packets or the number of users which connect to the base station (BS). Moreover, access control protocols proposed in [4]–[6] control all packet transmissions based on transmission requests preceded by each packet transmission. The reason for using such control schemes is that conventional access control protocols do not consider stochastic properties of each traffic.

It is reported that the bit rate of voice or video traffic is correlated [7], [8]. If access protocols consider such a stochastic property of voice or video traffic, it might be able to estimate the instantaneous number of simultaneously transmitted packets. It would bring us the possibility of improvement of the performance.

In this paper, we propose an access control protocol for CDMA packet communications. The proposed protocol controls packet transmissions of data users based on the estimation of the instantaneous number of simultaneously transmitted packets from voice and video users. The estimation is performed by exploiting a stochastic property of voice and video packet traffic, and carried out using an adaptive algorithm. We evaluate the performance of the proposed protocol with Monte Carlo simulation. We also compare the performance of the proposed protocol with that in the case where a stochastic property of traffic is not considered.

II. SYSTEM DESCRIPTION

We consider the uplink of a code division multiple access (CDMA) packet network. We assume that there are voice, video, and data users, and each of users generates voice, video and data packets, respectively. By using some connecting protocol, the number of connections of the voice and the video users to a base station (BS) is maintained at a fixed number K_{vo} and K_{vd}, respectively. A data user transmits his packet according to the proposed random access control protocol. We assume that the BS knows the number of received packets from all users and the type of media of each packet.

Packets generated by each user are transmitted to the BS after multiplied the spreading sequence which is assigned uniquely to each user. Time is divided into slots of length T_p=L/R, where L[bits] is the packet length and R[bps] is the transmission rate. Each of users can transmit a packet at the beginning of each slot.

The MAC protocol proposed in this paper only controls the packet transmissions of the data users. The BS broadcasts the control signal that conveys information used for the transmission control. The data users transmit the packets according to the information carried by the control signal from the BS. In contrast, the voice or the video users can transmit their packets immediately.

III. TRAFFIC MODELS

A. Voice Traffic Model

Voice packets generated by a single voice user are modeled as the two-state Markov process [7]. The generation process of voice packets is illustrated in Figure 1. The period of talkspurts and silences is exponentially distributed with mean kbps$\alpha^{-1} = 352$[ms] and $\beta^{-1} = 650$[ms], respectively. The coding interval is defined as T_d. Voice packets are generated every $T_d = 16$[ms] during talkspurts only, and are not generated during silences. The coding interval is represented by $\tau_d = T_d/T_p$[slots]. Voice packets therefore generate every τ_d[slots] during talkspurts.

We define $k_{vo}(n)$ as the number of voice packets which are simultaneously transmitted by K_{vo} voice users in the nth slot . The BS can know the number of simultaneously transmitted voice packets in the past slots by counting the number of received packets. But it can't know the number of voice packets that will be generated in the future slots. This is because the length of talkspurt is a random variable.

B. Video Traffic Model

Bit rate of a single video user during a video frame is modeled as a first order autoregressive process [8]. The generation process of the video packets is illustrated in Figure 2. The bit rate $r(v)$ in the vth video frame is expressed as by

$$r(v) = ar(v-1) + bw(m, \sigma_v), \tag{1}$$

where $w(m, \sigma_v)$ is sequence of Gaussian random variables with mean m and standard deviation σ_v. The length of video frame is T_f. Thus, the number of bits in the vth video frame is represented as $r(v)T_f$[bits]. The video message generated in one frame by a video user is divided into fixed length packets, then these packets are transmitted continuously. We set the parameters as $a = 0.99$, $b = 0.14$, $m = 5$[kbps] and $\sigma_v = 20$[kbps] [9]. Similarly to voice traffic, the BS can know the number of simultaneously transmitted packets $k_{vd}(n)$ in the past slots and can not know the number of packets that will be generate in the future slots.

C. Data Traffic Model

We assume that the number of data users K_{dt} is equal to infinite. The generation process of data packets is modeled as a Poisson process with the packet generation rate of λ. Moreover, we define the offered load of data packets G as the average number of packet generated from K_{dt} data users during one slot. Then, $G = \lambda T_p$. If we don't employ an access control, the numbers of packets generated in each slot are independent each other.

IV. FLUCTUATION OF NUMBER OF SIMULTANEOUSLY TRANSMITTED VOICE AND VIDEO PACKETS AND ITS AUTOCORRELATION FUNCTION

Figure 3 shows the fluctuation of the number of simultaneously transmitted voice and video packets $k_{vo+vd}(n) = k_{vo}(n) + k_{vd}(n)$ during 160 slots. In this example, we assume $K_{vo} = 100$ and $K_{vd} = 10$. To evaluate dependence of the numbers of simultaneously transmitted packets among slots, we derive the autocorrelation function $\phi(l)$ of $k_{vo+vd}(n)$.

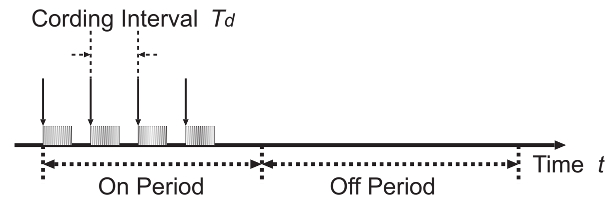

Fig. 1. Packet generations from a single voice user.

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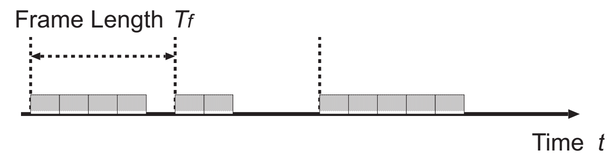

Fig. 2. Packet generations from a single video user.

Figure 4 shows the autocorrelation function of $k_{vo+vd}(n)$, where $\phi(l)$ is normalized by $\phi(0)$. This shows that the autocorrelation of $k_{vo+vd}(n)$ has the peaks that appear periodically. Moreover, the interval of their appearance equals to the coding period τ_d[slots] of voice packet traffic. This periodic property brings us the possibility to estimate the instantaneous number of simultaneously transmitted voice and video packets in the future slots.

V. DATA PACKET ACCESS CONTROL

A. Basic Idea of Proposed Access Control Protocol

The proposed protocol controls the packet transmission of the data users in order to preserve the average number of simultaneously transmitted packets from all users below a certain value. As described in section IV, the fluctuation of the number of simultaneously transmitted voice and video packets is periodical. Based on this property, the BS estimates the number of simultaneously transmitted packets from the voice and the video users in the the future slots.

B. Estimation of the Number of Simultaneously Transmitted Packets

The BS estimates the number of simultaneously transmitted packets from the voice and the video users in the future slots. This estimation is performed based on an adaptive algorithm [10] from the number of simultaneously transmitted packets in the past slots. To acquire the number of packets in the past slots, the BS counts the number of the voice and the video packets received over the observation period τ_s[slots]. We select $\tau_s = \tau_d$, where τ_d is the coding interval of voice packet traffic. With counting the number of voice and video packets over the observation period, the BS obtains the sequence of the numbers of simultaneously transmitted voice and video packets, $k_{vo+vd}(n), k_{vo+vd}(n-1), \ldots, k_{vo+vd}(n-\tau_s+1)$ at the nth slot . The BS then estimates the sequence of the numbers of packets $k_{vo+vd}(n+\tau_s), k_{vo+vd}(n+\tau_s-1), \ldots, k_{vo+vd}(n+1)$, that is the numbers of simultaneously transmitted packets over the future τ_s slots. The estimated values at the nth slot are defined as

$$\hat{k}_{vo+vd}(n + \tau_s - i) = w_i(n)k_{vo+vd}(n - i)$$

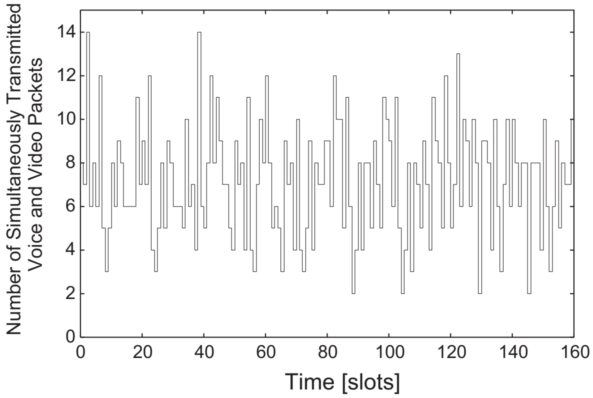

Fig. 3. Fluctuation of the number of simultaneously transmitted voice and video packets. ($K_{vo} = 100, K_{vd} = 10$)

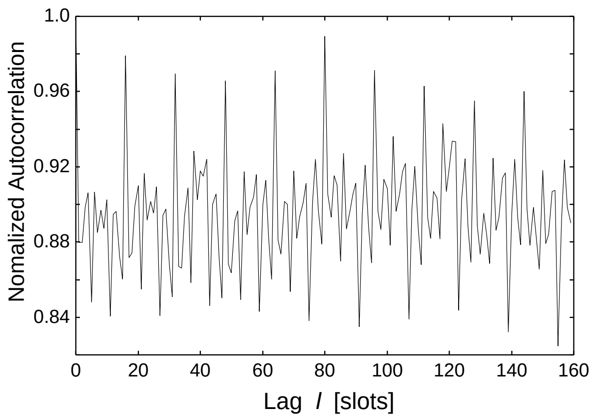

Fig. 4. Autocorrelation function of the number of simultaneously transmitted voice and video packets. ($K_{vo} = 100, K_{vd} = 10$)

$$(i = 0, 1, 2, \ldots, \tau_s - 1), \tag{2}$$

where $\hat{k}_{vo+vd}(n)$ is the estimated value of $k_{vo+vd}(n)$ and $w_i(n)$ is the weight coefficient.

The fluctuation of $k_{vo+vd}(n)$ is a stochastic process, and the estimation of $k_{vo+vd}(n)$ is accompanied by an error. This error is defined as

$$e_i(n) = k_{vo+vd}(n + \tau_s - i) - \hat{k}_{vo+vd}(n + \tau_s - i)$$
$$(i = 0, 1, 2, \ldots, \tau_s - 1). \tag{3}$$

Let us consider the problem of determining the weight coefficients to minimize the cost function J, where the cost function J is defined as the expected value of the summation of the estimation errors $e_i(n)$ from $i = 0$ to $i = \tau_s - 1$. The cost function may be expressed as

$$J = E\left[\sum_{i=0}^{\tau_s - 1} \{e_i(n)\}^2\right], \tag{4}$$

where $E[\cdot]$ denotes the expectation operator.

The weight coefficients are determined so that the cost function is minimized. At every τ_s slots, the weight coefficients are updated by

$$w_i(n + \tau_s) = w_i(n) - 2\mu k_{vo+vd}(n + \tau_s - i)e_i(n)$$
$$(i = 0, 1, 2, \ldots, \tau_s - 1), \tag{5}$$

where the parameter μ is the step-size parameter.

C. Calculation of Transmission Probability

The BS calculates the data packet transmission probability so as to preserve the total number of simultaneously transmitted packets in the slot, where the data packets is transmitted, below a certain threshold value. We define such threshold a value as α. The transmission probability in the nth slot is calculated by

$$P_{tr}(j|n) = \begin{cases} \dfrac{\alpha - \hat{k}_{vo+vd}(j)}{\hat{G}(n) \cdot \tau_s} & \text{if } \hat{k}_{vo+vd}(j) < \alpha \\ 0 & \text{otherwise} \end{cases}$$
$$(j = n+1, n+2, \ldots, n+\tau_s), \tag{6}$$

where $P_{tr}(j|n)$ denotes the probability that the packet generated in the nth slot is transmitted in the jth slot. The estimated offered load of data packets $\hat{G}(n)$ is defined as

$$\hat{G}(n) = \frac{\dfrac{k_{dt}(n)}{\tau_s}}{\dfrac{1}{\tau_s}\displaystyle\sum_{j=n-\tau_s}^{n-1} P_{tr}(n|j)} = \frac{k_{dt}(n)}{\displaystyle\sum_{j=n-\tau_s}^{n-1} P_{tr}(n|j)}, \tag{7}$$

where $k_{dt}(n)$ is the number of simultaneously transmitted data packets in the nth slot.

D. Procedure of Data Packet Transmission Control

In this subsection we explain the procedure of the proposed access control protocol for data packets. Let us consider that the BS controls the data packets generated in the nth slot.

First, the BS obtains the sequence of the numbers of simultaneously transmitted voice and video packets over the past τ_s slots, and updates the weight coefficients according to (5). The next step, the BS estimates the number of simultaneously transmitted voice and video packets over the future τ_s slots from (2). Then, the BS calculates the transmission probability $P_{tr}(j|n)$ for the data packets generated in the nth slot from (6) and (7). Finally, the BS broadcasts the transmission probability to all data users using control signals.

The data users transmit their packets generated in the nth slot according to the transmission probability $P_{tr}(j|n)$. The data packets may be transmitted in one of the τ_s slots from the $(n+1)$th slot to the $(n+\tau_s)$ slot. The probability that the packets generated in the nth slot are transmitted in the $(n+1)$th, $(n+2)$th, ..., $(n+\tau_s)$th slot is $P_{tr}(n+1|n)$, $P_{tr}(n+2|n)$, ..., $P_{tr}(n+\tau_s|n)$, respectively. The probability that the data packets generated in nth slot is not transmitted and discarded is represented by

$$\sum_{j=n+1}^{n+\tau_s} \{1 - P_{tr}(j|n)\}. \tag{8}$$

VI. Performance Evaluation

In this section, we evaluate the performance of the proposed access control protocol based on Monte Carlo simulation.

A. Simulation Settings

A direct sequence (DS)-CDMA packet communication system is considered. Perfect transmit power control is assumed so that all the packets transmitted from users are received by the BS at equal power level. We employ binary phase-shift keying(BPSK) as a modulation scheme and random signature as a spreading sequence. Then, the bit error probability $P_b(k)$ is obtain from [11] as

$$\begin{aligned} P_b(k) \;=\; & \frac{2}{3}Q\left[\left\{\frac{k-1}{3N}+\frac{N_0}{2E_b}\right\}^{-\frac{1}{2}}\right] \\ & + \frac{1}{6}Q\left[\left\{\frac{(k-1)\cdot N/3+\sqrt{3}\sigma}{N^2}+\frac{N_0}{2E_b}\right\}^{\frac{1}{2}}\right] \\ & + \frac{1}{6}Q\left[\left\{\frac{(k-1)\cdot N/3-\sqrt{3}\sigma}{N^2}+\frac{N_0}{2E_b}\right\}^{\frac{1}{2}}\right], \qquad (9) \end{aligned}$$

where

$$\sigma^2 = (k-1)\left\{N^2\frac{23}{360}+N\left(\frac{1}{20}+\frac{k-1}{36}-\frac{1}{20}-\frac{k-2}{36}\right)\right\}. \qquad (10)$$

In (10), k is the number of simultaneously transmitted packets, N is the spreading factor, E_b is the signal energy per bit, and $N_0/2$ is two sided power spectral density of the additive white Gaussian noise process. Furthermore, $Q[x] = \frac{1}{\sqrt{2\pi}}\int_x^\infty \exp(-u^2/2)du$. The packet successful probability $Q_s(k)$ is represented by

$$Q_s(k) = \{1-P_b(k)\}^L. \qquad (11)$$

In the performance evaluation, packets are generated according to the traffic models described in section III. Then, we obtain the number of simultaneously transmitted packets in each slot when we apply the proposed access control protocol. Final, we calculate the packet successful probability from (11), and decide whether the packet transmission succeeds or not. The simulation results are obtained by averaging over 5×10^3 runs where each run is performed over 2×10^6 slots.

As for the performance measures, we consider the data packet throughput under the constraint of that the voice packet loss rate is less than 10^{-3}. The data packet throughput is defined as the average number of packets correctly received per slots by the BS. The voice or the video packet loss rate is defined as the ratio of the number of packets received with errors by the BS to the total number of packets transmitted from the voice or the video users.

To evaluate the advantage of the proposed protocol, we compare the performance of the proposed protocol with that in two different cases. One is the "no estimation" case in which a periodicity of voice packet traffic is not considered, that is, the BS does not estimate the instantaneous number of simultaneously transmitted voice and video packets but calculates the average of them. In this case, (2) is substituted by

$$\hat{k}_{vo+vd}(n+\tau_s-i) = \sum_{j=0}^{\tau_s-1} k_{vo+vd}(n-j) \qquad (i=0,1,2,\ldots,\tau_s-1), \qquad (12)$$

where right-hand of this equation expresses the average number of simultaneously transmitted voice and video packets over the past τ_s slots. The other is the "perfect estimation" case in which the BS can estimate the number of simultaneously transmitted packets in the future slots perfectly. In this case, the estimation error defined by (3) is zero.

The system parameters are summarized in Table I

TABLE I

SIMULATION PARAMETERS

System Parameters		
Packet Length	L	500[bits]
tb Transmission Rate	R	500[kbps]
Spreading Factor	N	127
E_b/N_0		20[dB]
Traffic Parameters		
Mean Talkspurts Length	α^{-1}	352[ms]
Mean Silence Length	β^{-1}	650[ms]
Coding Interval	τ_d	16[slots]
Video Frame Length	τ_f	40[slots]
Access Control Parameters		
Observation Period	τ_s	16,40,80[slots]
Step-size Parameter	μ	0.0001

B. Simulation Results

Figures 5 and 6 show the packet loss rate for voice and video, respectively. In these figures, the curve "proposed protocol" shows the performance of the proposed protocol which estimates the instantaneous number of the simultaneously transmitted voice and video packets with the adaptive algorithm. In these results, the threshold α is selected under the constraint of that the voice packet loss rate is less than 10^{-3}.

In Figure 5, the voice packet loss rate is preserved below 10^{-3} as the offered load of data packet increases. Furthermore, the voice packet loss rate of the proposed protocol is smaller than that of the no estimation case when the offered load is small.

In Figure 6, the maximum video packet loss rate of both the proposed protocol and the perfect estimation case is higher than that of the no estimation case. In the case of no estimation, the data packet transmission probability is calculated based on the average number of simultaneously transmitted voice and video packets. In this situation, the maximum value of voice packet loss rate equals to that of video packet loss rate because the amount of interference with voice packets is equal to that with video packets. In the case of perfect estimation, the instantaneous number of simultaneously transmitted voice and video packets is estimated perfectly. In such situation, data packets are attempted to be transmitted in the slot where number of simultaneously transmitted voice and video packets is relatively small. Therefore, data user can transmit more packets with keeping the maximum voice packet loss rate below 10^{-3}. However, this means that the amount of interference with video packets increases relatively. This is the reason why the video packet loss rate in the perfect estimation case is higher than that in no estimation case. Furthermore, in the case of proposed protocol, the estimation of the number of simultaneously transmitted voice and video packets is accompanied with an error. Because of this fact, the video packet loss rate using the proposed protocol is the highest among the

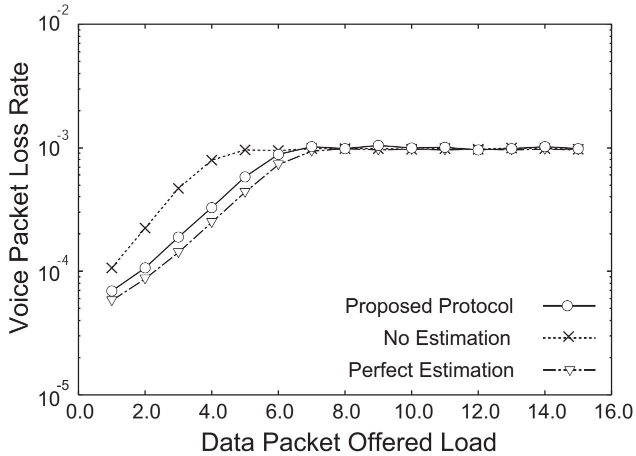

Fig. 5. Voice packet loss rate versus data packet offered load

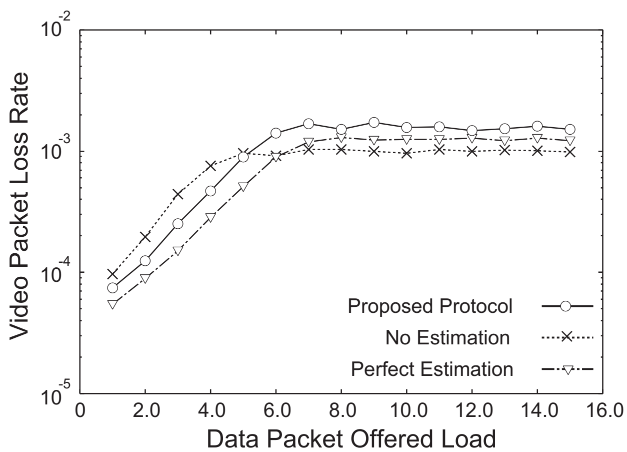

Fig. 6. Video packet loss rate versus data packet offered load

other cases.

Figure 7 depicts the data packet throughput in which the same parameters are used in Figs. 5 and 6. In this figure, the maximum data packet throughput of the proposed protocol is about 1.3 higher than that of the no estimation case. This result shows the advantage of estimating the instantaneous number of simultaneously transmitted packets. Furthermore, the performance of the proposed protocol is lower than that of the perfect estimation case. Even if the BS estimates the instantaneous number of voice and video packets based on a stochastic property of traffic, some estimation errors occur. Figure 7 indicates that this estimation error can affect the performance of our proposed system.

VII. Conclusions

In this paper, we have proposed the access control protocol based on the estimation of the instantaneous number of simultaneously transmitted packets from voice and video users. The estimation is performed based on a periodic property in fluctuation of voice packet traffic. We have evaluated the performance of the proposed protocol by comparing it with two different cases. The results show that the data packet throughput performance of the proposed protocol is higher than the case without estimation. This results indicate the advantage of estimating the instantaneous number of simultaneously transmitted packets.

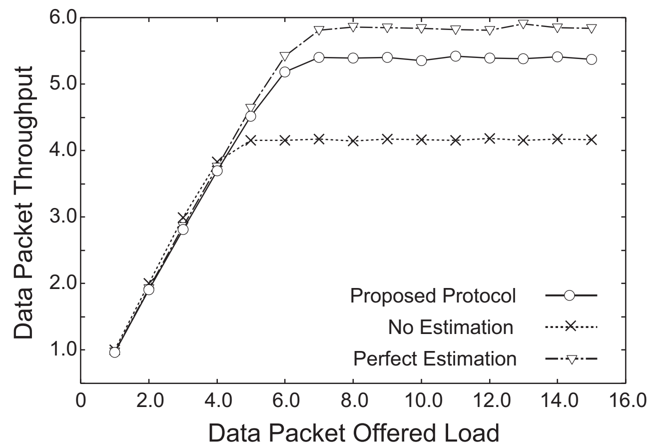

Fig. 7. Data packet throughput versus data packet offered load

We have treated the periodical statistical model for voice and video traffic in this paper. If the periodicity does not appear so clearly, we expect that the instantaneous number of voice and video packets could be estimated roughly, but the proposed protocol might improve the performance.

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References

[1] I. F. Akyildiz, et. al., “Medium access control protocols for multimedia traffic in wireless networks,” *IEEE Netw.*, vol. 13, no. 4, pp. 39–47, July/Aug. 1999.

[2] M. Soroushnejad and E. Geraniotis, “Multi-access strategies for an integrated voice/data CDMA packet radio network,” *IEEE Trans. Commun.*, vol. 43, no. 2/3/4, pp. 934–944, Feb./March/April 1995.

[3] A. Sandouk, T. Yamazato, M. Katayama, and A. Ogawa, “An access control protocol for a heterogeneous traffic with a multi-code CDMA scheme,” *IEICE Trans. Fundamentals*, vol. E83-A, no. 11, pp. 2085–2092, Nov. 2000.

[4] A. E. Brand and A. H. Aghvami, “Performance of a joint CDMA/PRMA protocol for mixed voice/data transmission for third generation mobile communication,” *IEEE J. Select. Areas Commun.*, vol. 14, no. 9, pp. 1698–1707, Dec. 1996.

[5] I. F. Akyildiz, D. A. Levine, and I. Joe, “A slotted CDMA protocol with BER scheduling for wireless multimedia networks,” *IEEE/ACM Trans. Netw.*, vol. 7, no. 2, pp. 146–158, April 1999.

[6] J-H. Wen, J-K. Lain, and Y-W. Lai, “Performance evaluation of a joint CDMA/NC-PRMA protocol for wireless multimedia communications,” *IEEE J. Select. Areas Commun.*, vol. 19, no. 1, pp. 95–105, Jan. 2001.

[7] H. Heffes and D. M. Lucantoni, ”A Markov modulated characterization of packetized voice and data traffic and related statistical multiplexer performance,” *IEEE J. Select. Areas Commun.*, vol. SAC-4, no. 6,pp. 856–868, Sep. 1986.

[8] B. Maglaris, et al, ”Performance models of statistical multiplexing in packet video communications”, *IEEE Trans. Commun.*, vol. 36, no. 7, pp. 834–843, July 1988.

[9] N. Dimitriou, G. Sfikas, and R. Tafazolli,” Call admission policies for UMTS,” *Proc. VTC 2000*, vol. 2, pp. 1420–1424, 2000.

[10] S. Haykin, *Adaptive filter theory third edition*, Prentice Hall International, 1996.

[11] J. M. Holtzman, “A Simple, accurate method to calculate spread-spectrum multiple-access error probabilities,” *IEEE Trans. Commun.*, vol. 40, no. 3, pp. 1223–1230, March 1992.