음성/데이터 CDMA 시스템에서의 서비스 품질을 고려한 비 실시간 데이터 패킷 전송 제어 정책

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QoS-Sensitive Admission Policy for Non-Real Time Data Packets in Voice/Data CDMA Systems

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Abstract

In this paper, we propose a QoS-sensitive admission threshold method for the transmission of the non-real time data packet such that the quality of services for both voice and data traffics are maintained to a required level. By detecting the active voice traffic during the current time slot, the non-real-time data packets are transmitted up to an admission threshold level during the next time slot. We found out that the optimum admission threshold is four voice traffic resources lower from the maximum allowable threshold to maintain the outage probability within 1% when the connected voice users are 15.

I. INTRODUCTION

Direct-Sequence Code Division Multiple Access(DS-CDMA) system has been deployed recently for voice communications and considered as one of the good candidates for future multimedia wireless services because of its ability to provide high utilization of the bandwidth and to support different Grade of Service(GoS) requirements[1]. Recently, a lot of works have been widely done for accommodating voice and data traffics in the DS-CDMA systems[2,6]. Most of them are related with the design of traffic control algorithms for maximizing the throughput of data traffic while guaranteeing the pre-determined QoS of voice traffic. Sampath et al. proposed an access control method of data traffic, which schedules data packet transmissions according to the active voice load[2]. This access control method is a practical method employing a broadcasting channel while ensuring adequate quality of service. In [6], Yang proposed two models for the transmission of mixed voice/data traffics without broadcasting the persistence state to data users: one is based on thresholds and another is based on the graceful degradation of the CDMA system performance. In the first threshold model, all voice and data packets are accepted within the predefined thresholds of voice and data traffic, respectively. In the second graceful degradation model, all data users are accepted and transmit their data packets during the next slot but they are successful with a certain substantial probability. According to Yang's result, the graceful degradation model gets the higher data packet throughput and lower data packet delay. However, Yang's scheme does not guarantee the quality of the on-going voice traffics. Especially, for the graceful degradation model it is expected that increasing the data traffic degrades the quality of voice traffics severely.

In this paper, we propose a QoS-sensitive admission threshold method for the transmission of non-real time data packet while guaranteeing the QoS of voice and data traffics. We analyze the throughput, delay and the reservation packet loss probability for the data traffic and the outage probability for the voice and data traffics as a quality measurement. The proposed admission threshold method detects the active voice user during the current time slot. By using the detected activity the proposed method allows the data terminals to transmit the data packets up to an admission threshold during next time slot. Further, we search for the optimum admission threshold level for the non-real time data packet such that the quality of both voice and data traffics is maintained to a required level.

The remainders of the paper are organized as follows: In Section II, system model and call admission control criterion are given. An analytical procedure for analyzing voice/data traffic and the admission strategy for non-real time data packet are presented in Section III. In Section IV, numerical results are taken into consideration and discussions are given. This paper is concluded in Section V.

II. SYSTEM MODEL

The CDMA communication network considered is time slotted. Each time slot is equal to the duration of single data packet for data transmission. There are two kinds of switching methods to deal with different traffic transmission: one is a circuit switching method for voice traffics and the other is a reservation type packet switching method for the traffics transmitting non-real time data packets. In a reservation type packet switching method, all data users share a reservation channel and the permission signal will be assigned in advance before data transmission. In this system, we assume that voice traffic has priority over data traffic because the circuit switching method guarantees both resources and immediate transmission for voice traffic. Further, we assume that the resources required to transmit one data packet is larger than voice traffic because data packets usually require the lower BER than voice traffic. With respect to the data server equivalent to the remaining resources, we consider only the case that the resources required to transmit one data packet is k times larger than voice traffic for simplicity, where k is an integer number.

A. Model for Voice Traffic

Each voice user is modeled two-state Markov model representing voice activity. Active users transmit with rate R_v bps and inactive users do not transmit. The rate of flow from the IDLE state to the ACTIVE state is λ, and from ACTIVE to IDLE is μ. If the following condition is satisfied a new voice call is accepted, if not, rejected.

$$\gamma_v N_v(t) \le 1. \tag{1}$$

where

$$\gamma_v = \frac{a}{\frac{W}{R_v}\left(\frac{E_b}{N_o}\right)^{-1}_{v,req} + a}. \tag{2}$$

a is the voice activity factor.

B. Model for Data Traffic

Each terminal intending to transmit data packet is sending the request signal(RS) or reservation packet to the base station. This RS is stored in the buffer at the base station and error packets will be retransmitted. The RS can be blocked only due to the limitation of buffer at the base station and this blocked RS will be retransmitted. The base station can know the number of current active voice users and allocate the remaining resources, that are not used by the voice users, to the terminals having the data packet to transmit during the next time slot. Since the base station can not know the resources to be used during the next time slot(only knows the resources used during current time slot), the base station does not allocate the full remaining resources. Admission of full remaining resources may degrade the quality of both traffics. Instead, the base station allocates the resources up to the admission threshold.

III. TRAFFIC ANALYSIS AND ADMISSION STRATEGY

A. Voice Traffic Analysis

When N_v users are connected, the model for voice traffic considering voice activity can be represented discrete-time Markov chain as in Fig.2. In this Markov model, the transition probabilities are obtained easily as follows[2]:

$$p(k+1|k) = \frac{\widehat{\lambda_k}}{\widehat{\lambda_k}+\widehat{\mu_k}}(1-\exp(-(\widehat{\lambda_k}+\widehat{\mu_k})d)) \tag{3}$$

$$p(k-1|k) = \frac{\widehat{\mu_k}}{\widehat{\lambda_k}+\widehat{\mu_k}}(1-\exp(-(\widehat{\lambda_k}+\widehat{\mu_k})d)) \tag{4}$$

$$p(k|k) = \exp(-(\widehat{\lambda_k}+\widehat{\mu_k})d) \tag{5}$$

where $1/\widehat{\lambda_k} = 1/[\lambda(N_v-k)]$, $1/\widehat{\mu_k} = 1/\mu k$ and d is slot duration. The steady state probability of being in state k, $\pi_v(k)$, can be easily obtained to be

$$\pi_v(k) = \frac{\binom{N_v}{k}(\lambda/\mu)^k}{\sum_{j=0}^{N_v}\binom{N_v}{j}(\lambda/\mu)^j}. \tag{6}$$

B. Data Traffic Analysis

When N_v is constant(under the assumption that the number of connected voice users is constant during a packet duration), we can model the base station as queue and remaining data servers, where r is the number of data packet servers equivalent to the remaining resources, B is buffer size and λ_d is the reservation packet arrival rate for the data traffic.

Admission strategy for the non-real time data packets is as follows: since we can not know the number of the active voice users during the next time slot, the base station allocates the resources for the data terminals up to the admission threshold(Th) that is lower than the maximum allowable data packet during the current time slot. This procedure is depicted in Fig.1. By adjusting the admission threshold, we can find the optimum admission threshold such that the outage probability is maintained to a required level.

With the buffer whose size is B, we can model the base station as M/D/r/B queue. To get the steady state probability of the data packets in the buffer with the given threshold, Th, we search for the transition probabilities of the states of buffers from the following relationship

$$p(n_{i+1}|n_i) = \sum_{a_{i+1}=0}^{\infty}\sum_{r=0}^{\lceil Th/k \rceil} p(n_{i+1}|n_i, a_{i+1}, r)p(a_{i+1})p(r) \tag{7}$$

where n_{i+1} is the number of reservation packets at the buffer during the (i+1)th time slot, $p(a_{i+1})$ is the probability density function of the reservation packet arrival and $p(r)$ is the p.d.f. of available servers for data packets. The conditional probability in Eq.(7) is obtained by

$$p(n_{i+1}|n_i, a_{i+1}, r) = \begin{cases} 1 & \text{if } n_{i+1} = \min(B, n_i + a_{i+1} - t_{i+1}) \\ 0 & otherwise \end{cases}. \tag{8}$$

The number of reservation packets at the buffers during the (i+1)th time slot, n_{i+1}, is represented as follows:

$$n_{i+1} = n_i + a_{i+1} - t_{i+1} \tag{9}$$

where a_{i+1} is the number of arrived reservation packets during the (i+1)th time slot, n_i is the number of remaining reservation packets at the buffers during the (i)th time slot and t_{i+1} is the number of transmitted data packets during the (i+1)th time slot. $p(a_{i+1})$ follows the Poisson distribution and is given by:

$$p(a_{i+1}=a)=\frac{e^{-\lambda_d}\lambda_d^a}{a} \qquad \text{for } a=0,1,2,\cdots \tag{10}$$

When one data packet is k times larger than voice traffic, we obtain the general probability distribution of data severs as follows:

$$p(r)=\begin{cases} \sum_{j=0}^{k-1}\pi_v(Th-k\cdot r-j) & \text{if } 0< r\le \lceil \frac{Th}{k} \rceil \\ \sum_{i=0}^{R+k-1}\pi_v(N_{\max}-i) & \text{if } r=0 \\ 0 & \text{otherwise} \end{cases} \tag{11}$$

where $N_{\max}$ is the number of maximum allowable voice users and R is reservation value, that is $R=(N_{\max}-Th)$. The unit of R is the resource equivalent to the resource required to transmit one voice traffic. $\lceil x \rceil$ represents the largest value smaller than or equal to x.

The states of buffer, $n_i=0,1,\cdots,B$, follow discrete Markov chain and their steady state probability distribution, $\{\pi_d(i)\}$, is obtained by solving the following set of $(B+1)$ linear equations

$$\begin{cases} \sum_i \pi_d(i)P_{ij}=\pi_d(j) & \text{where } P_{ij}=p(n_{i+1}=j\,|\,n_i=i) \\ \sum_i \pi_d(i)=1 \end{cases} \tag{12}$$

where P_{ij} is the transition probability defined in Eq.(7). We define the average data throughput as

$$T=\sum_{r=1}^{\lceil \frac{Th}{k} \rceil}\sum_{n_i=1}^{B} t_{i+1}\pi_d(n_i)p(r). \tag{13}$$

Another important performance parameter for the system considered is the average data delay. Here, it is defined as

$$D=1+1/2+\frac{1}{T}\sum_{n_i=1}^{B} n_i\pi_d(n_i) \tag{14}$$

where the value of D is in packets(time slot), 1 is the service time of one packet, 1/2 is the average waiting time to the next time slot and last term represents average data queueing delay. It is noted that the delay due to the transmissions of reservation packet and RS are ignored. Also, we define the reservation packet loss probability as follows:

$$P_L=\frac{\lambda_d-T}{\lambda_d} \tag{15}$$

Since the buffers for storing the reservation packets at the base station are not infinite, the blocking of reservation packets could be occurred and the reservation packet loss in Eq.(15) can be considered as the reservation packet blocking.

IV. NUMERICAL RESULTS

The parameters for a numerical example are given in Table I. For the system with specific parameters in Table I, we know that the required resources for the data packet are two times larger than for the voice traffic and k=2 in Eq. (11). Now, let's find out the outage probability experienced by both voice and data traffics when we are transmitting the non-real time data packet according to the proposed admission strategy mentioned in Section III. We define the outage probability as follows:

$$P_{out}=\Pr\{\gamma_{v,active}N_v(i+1)+\gamma_d N_d(i+1)\ >\ 1\} \tag{16}$$

where

$$\gamma_{v,active}=\frac{1}{\frac{W}{R_v}\left(\frac{E_b}{N_o}\right)_{v,req}^{-1}+1} \tag{17}$$

$$\gamma_d=\frac{1}{\frac{W}{R_d}\left(\frac{E_b}{N_o}\right)_{d,req}^{-1}+1} \tag{18}$$

and $N_d(i+1)=\min(n_i,r)$. $\min(x,y)$ represents the minimum value between x and y. Index *(i+1)* represents the (i+1)th time slot and n_i is the number of remaining reservation packets during the (i)th time slot in the buffer. Note that $N_v(i+1)$, n_i, r are random variables. By conditioning on both n_i and r we get the conditional outage probability

$\Pr\{outage|n_i,r\}=$

$$\Pr\left\{N_v(i+1)>\frac{1-\gamma_d\min(x,y)}{\gamma_{v,active}}\,|n_i=x,\ r=y\right\}. \tag{19}$$

After taking expectation on n_i and r, we get the outage probability as follows:

$$P_{out}=\sum_{n_i=0}^{B}\sum_{r=0}^{Th}\sum_{i=\lfloor \frac{1-\gamma_d\min(n_i,r)}{\gamma_{v,active}} \rfloor}^{N_v} \cdot\binom{N_v}{i}p_v^i(1-p_v)^{(N_v-i)}\pi_d(n_i)p(r) \tag{20}$$

where $\lfloor x \rfloor$ represents the smallest value larger than or equal to x and $p_v=\frac{1/\mu}{1/\lambda+1/\mu}$.

Fig.3 shows the average throughput for data traffic when the number of connected voice users is 15 and the size of buffer is 30. For the higher data load, the lower is the admission threshold the lower is the average throughput. In order to find out the optimum admission threshold such that the quality of both traffic is maintained to a required level, look into the outage probability depicted in Fig.4. To satisfy the required outage probability of 1%, we should set the admission threshold to the four voice traffic resources lower from the $N_{\max}$, i.e. $R=4$. Fig.5 and Fig.6 show the average delay of the data traffic and the average reservation packet loss(or blocking) probability for the different admission thresholds, respectively. Average delay and average reservation packet loss(or blocking) probability depend on how many buffers base station have. For the system considered, the reservation packet loss means the blocking of the reservation packets due to the limitation of the buffers for storing the reservation packets at the base station.

V. CONCLUSIONS

We considered the transmission of the non-real time data packets for the system that employs a circuit switching method for voice traffic and a reservation type packet switching method for non-real time data traffic. Admission threshold method guaranteeing the quality of services for voice and data traffics is proposed and analytical procedures to find the optimum threshold are presented.

For a specific example, where the number of connected voice users(N_v) are 15, the size of buffers(B) at the base station is 30 and with the parameters in Table I, the throughput, outage, delay and reservation packet loss probability are investigated for the different admission thresholds. By investigating these results, we found out that the admission threshold should be set to the four voice traffic resources lower from the N_{max} for the transmission of the non-real time data packets such that the outage probability of both voice and data traffics is maintained to 1%.

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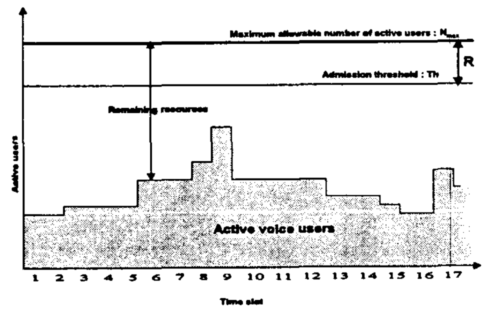

Fig.1 Admission model for data packets

Parameter	Symbol	Value
Allocated frequency bandwidth	W	1.25Mhz
Data rate for voice traffic	R_v	9.6kbps
Data rate for data traffic	R_d	9.6kbps
Required bit energy to interference power spectral ratio for voice traffic	$(E_b/N_0)_{vreq}$	7dB
Required bit energy to interference power spectral ratio for data traffic	$(E_b/N_0)_{dreq}$	10dB
Mean of ON time duration of voice user	$1/\mu$	1sec
Mean of OFF time duration of voice user	$1/\lambda$	1.5sec
Slot duration	d	0.02sec

Table I. System parameters

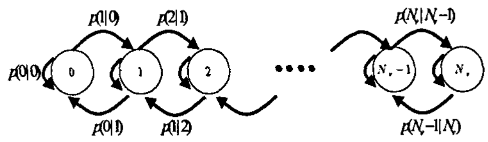

Fig.2 Queueing model for voice traffic(when N_v user is connected)

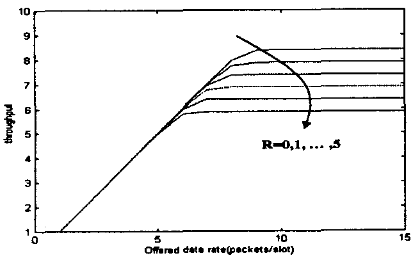

Fig.3 Average throughput(N_v=15,B=30)

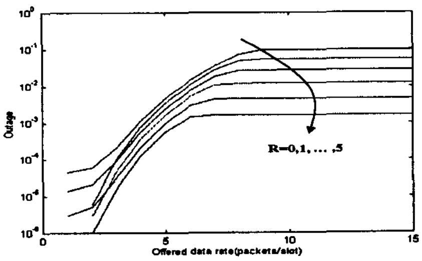

Fig.4 Outage probability(N_v=15,B=30)

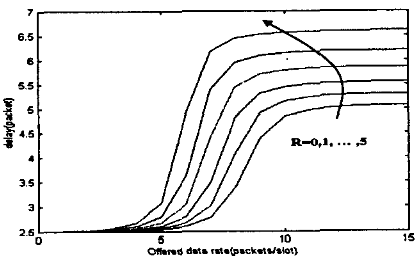

Fig.5 Average delay(N_v=15,B=30)

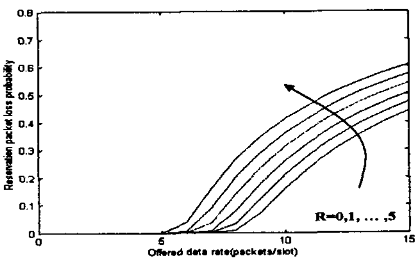

Fig.6 Reservation packet loss probability(N_v=15,B=30)