Admission Control for Variable Spreading Gain CDMA Wireless Packet Networks

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***Abstract*—A technique for variable spreading gain code-division multiple access (VSG-CDMA) with transmit power control has recently been proposed [4], [5] to provide integrated services in a wireless packet network. System capacities of VSG-CDMA networks with and without considering user activity factors were derived. The analysis gives a lower bound of system capacity because a user's transmitted power is considered part of the interference to its own signal. In this paper, we present an optimistic upper bound assuming that there is no multipath fading or the energy of a signal in all multipath components can be fully resolved. An optimum power vector assignment and a simple criterion for all users to simultaneously achieve their quality of service (QoS) requirements (in terms of bit energy-to-interference density ratios) are derived. We found from numerical examples that, compared with the lower bound given in [4] and [5], significant improvement can be obtained if multipath fading can be satisfactorily handled. Several access control schemes are also studied to guarantee delay bound requirements. Simulation results reveal that more connections can be further accommodated with access control.**

***Index Terms*—CDMA, packet switching, power control.**

I. Introduction

MULTIMEDIA applications over wireless networks have recently attracted much attention from researchers [4], [5], [7]–[16]. It is believed that packet switching can support mixed traffic such as data, voice, and video more efficiently than circuit switching. To implement packet switching in a wireless network, a multiple-access control scheme is required to coordinate medium access among active users. Time-division multiple access (TDMA) and code-division multiple access (CDMA) are two categories of medium access control techniques.

Using CDMA as the medium access control scheme eliminates the need for frequency plan revision and channel reallocation every time a new cell is introduced. Moreover, universal frequency reuse makes soft handoff possible which, when combined with proper power control, can largely improve system capacity. To support integrated services, multicode CDMA (MC-CDMA) [7], [8] and variable spreading gain CDMA (VSG-CDMA) coupled with transmitting power control [4], [5] were proposed as candidate systems. In the MC-CDMA system, K different PN codes are allocated to a user which needs K times of a basic rate. The user converts its signal stream into K parallel streams, encodes each one with a different PN code, and modulates them with a different Walsh modulator. The modulated signals are superimposed before upconverting for transmission.

In the VSG-CDMA system, users are assigned different power levels based on their data rates and quality of service (QoS) requirements. The QoS requirement is often characterized by bit energy-to-interference density ratio (or signal-to-interference ratio, SIR) because, given a transmission technology, the bit error rate can be derived from SIR. Two users having identical QoS requirements are allocated power levels proportional to their data rates; and two users having identical data rates are allocated power levels proportional to their QoS requirements. Let r_k, QR_k, and P_k denote respectively the data rate, QoS requirement, and the allocated power level of user k. As a result, the power levels allocated to user i and user j satisfy $(P_i/P_j) = (r_i/r_j)(QR_i/QR_j)$. Under such a power allocation algorithm, system capacity and admission criterion were derived in [4] and [5]. The calculations provide a lower bound of system capacity because the transmitted power of user i is considered part of the total interference to user i itself.

In this paper, we present an upper bound of system capacity for the VSG-CDMA system. In our derivations, we assume that there is no multipath fading or the energy of a signal in all multipath components can be 100% resolved (say, with the help of RAKE receivers). We prove a theorem on optimum power vector assignment and derive a simple admission criterion for all users to simultaneously achieve their QoS requirements. We found from numerical examples that, compared with the lower bound provided in [4] and [5], significant improvement can be obtained if multipath fading can be well handled. We also study with numerical examples the improvement assuming the requested QoS can be violated with a certain small probability.

In addition to requesting bounds on bit error probability, some applications may ask for guaranteed delivery of packets with time constraints. We propose several access control (AC) schemes to meet the requirements. For a CDMA system which takes advantage of activity factors [1], [2], [6], it may happen from time to time that the requested SIR's are violated. When it happens, all packets involved are destroyed, resulting in a big loss. Our proposed AC schemes can totally eliminate this possibility.

The rest of this paper is organized as follows. Section II describes the investigated system model. Optimum power assignment is derived in Section III. Admission criterion is presented in Section IV. Several AC schemes are studied in Section V. Numerical results are discussed in Section VI. Finally, the conclusion is in Section VII.

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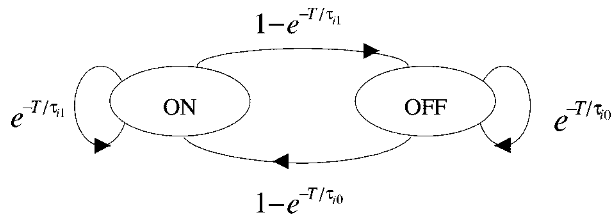

Fig. 1. Source model: two-state Markov process.

II. System Model

Consider a cellular direct-sequence (DS) CDMA system. To reduce interference and improve system capacity, we assume cells are sectorized ideally so that every cell site is equipped with three perfect directional antennas, each covers 1/3 of the cell area. In this paper, we study the reverse link, i.e., the users-to-base station direction. For each base station, let P_{oc} and P_{ic} denote, respectively, the received power for other-cell interference and the received power for all users in its area. Interference from other cells is characterized by the relative other-cell interference factor f [2], [3], [6], which is defined as

$$f \equiv \frac{P_{oc}}{P_{ic}}.$$

In [3], a tight upper bound on other-cell interference was derived, the result gives $f = 0.55$ when the propagation attenuation is proportional to the fourth power of the distance times a lognormally distributed component with 8-dB differential standard deviation.

We assume time is slotted so that the length of a slot is equal to the transmission time of a packet. Each user is modeled as a two-state Markov process. When user i is in state 0 (the OFF state), no data is generated. When it is in state 1 (the ON state), data is generated with rate r_i. The residence time in state 0 and state 1 are exponentially distributed with parameters τ_{i0} and τ_{i1}, respectively. As a result, the state transition rates from state 0 to state 1 and from state 1 to state 0 are $1/\tau_{i0}$ and $1/\tau_{i1}$, respectively. The relative amount of time for user i to stay in the ON state is called its activity factor and is denoted by α_i. For the two-state Markov process model, we have $\alpha_i = (\tau_{i1}/\tau_{i0} + \tau_{i1})$. The probabilities of state transitions during a slot time T are shown in Fig. 1.

III. Optimum Power Assignment

Assume there are M types of services. For convenience, a user of type i service is called a type i user. The processing gain for a type i user when it is in the ON state, denoted by F_i, is given by $F_i = (W/r_i)$, where W represents the spread bandwidth. Consider a particular sector and assume there are k_i type i users. We assume power control is ideal and the received power at cell site for type i users is $P_i, i = 1, 2, \cdots, M$. As in [1] and [2], we characterize the type i users' quality of service by $(E_b/I_0)_i$, the SIR, which can be determined from the requested bit error probability. For example, it was reported that the required E_b/I_0 to obtain a bit error probability of 10^{-3} (digital voice quality requirement) is 7 dB under some transmission technology [1]. In this paper, the required E_b/I_0 for any type i user is denoted by QR_i. Assume all users are in the ON state. As a result, (E_b/I_0) for any type i user is given by

$$\left(\frac{E_b}{I_0}\right)_i = \frac{\dfrac{P_i}{r_i}}{\dfrac{\left(\displaystyle\sum_{j=1}^{M} k_j P_j\right)(1+f) - P_i + \eta}{W}} = \frac{P_i F_i}{\displaystyle\sum_{j=1}^{M} k_j (1+f) P_j - P_i + \eta} \tag{1}$$

where η represents the thermal noise and f denotes the relative other-cell interference factor defined in Section II. Notice that in (1) the transmitted power of a user is not considered part of its total interference. This is true if there is no multipath fading or the energy of a signal in all multipath components can be 100% resolved. The system capacity derived based on this assumption is thus an optimistic upper bound.

For type i users to meet their QoS requirement, we must have

$$\frac{P_i F_i}{\displaystyle\sum_{j=1}^{M} k_j (1+f) P_j - P_i + \eta} \geq QR_i. \tag{2}$$

After neglecting the thermal noise and rearranging (2), we get

$$\sum_{j=1}^{M} k_j P_j \leq P_i \frac{1 + \dfrac{F_i}{QR_i}}{1+f}. \tag{3}$$

Consequently, all users meet their QoS requirements if one can find a power vector $\vec{P} = (P_1, P_2, \cdots, P_M)$ such that the inequality in (3) holds for all i, $1 \leq i \leq M$. Finding a power vector to meet the inequality in (3) is obviously not a trivial task. In the following theorem, we prove that some constraint can be put on the power vector to simplify the problem.

Theorem 1: Given $k_1, k_2, \cdots,$ and k_M. If there exists a power vector $\vec{P} = (P_1, P_2, \cdots, P_M)$ such that

$$\sum_{j=1}^{M} k_j P_j \leq P_i \frac{1 + \dfrac{F_i}{QR_i}}{1+f}, \qquad \text{for all } i, \qquad 1 \leq i \leq M$$

then there exists a power vector $\vec{P}^* = (P_1^*, P_2^*, \cdots, P_M^*)$ such that

$$P_i^* \frac{1 + \dfrac{F_i}{QR_i}}{1+f} = P_j^* \frac{1 + \dfrac{F_j}{QR_j}}{1+f}, \qquad \text{for all } i \text{ and } j$$

and

$$\sum_{j=1}^{M} k_j P_j^* \leq P_i^* \frac{1 + \dfrac{F_i}{QR_i}}{1+f} \qquad \text{for all, } i, \qquad i \leq i \leq M.$$

Proof: Assume there exists a power vector $\vec{P} = (P_1, P_2, \cdots, P_M)$ such that

$$\sum_{j=1}^{M} k_j P_j \le P_i \frac{1+\dfrac{F_i}{QR_i}}{1+f}, \qquad \text{for all } i, \qquad 1 \le i \le M.$$

Let k be an integer which satisfies

$$P_k \frac{1+\dfrac{F_k}{QR_k}}{1+f} \le P_J \frac{1+\dfrac{F_j}{QR_j}}{1+f}, \qquad \text{for all } j, \qquad 1 \le j \le M.$$

Let $P_k^* = P_k$ and

$$P_j^* = P_k^* \frac{1+\dfrac{F_k}{QR_k}}{1+\dfrac{F_j}{QR_j}}, \qquad \text{for all } j \ne k, \qquad 1 \le j \le M.$$

It is clear that $P_j^* \le P_j$ and, therefore

$$\begin{aligned}\sum_{j=1}^{M} k_j P_j^* \le \sum_{j=1}^{M} k_J P_j \le P_k \frac{1+\dfrac{F_k}{QR_k}}{1+f} \\ = P_k^* \frac{1+\dfrac{F_k}{QR_k}}{1+f} = P_1^* \frac{1+\dfrac{F_i}{QR_i}}{1+f}, \\ \text{for all } i, \qquad 1 \le i \le M.\end{aligned}$$

This completes the proof of Theorem 1.

From the above theorem, one can assign power levels with the constraint

$$\frac{P_i}{P_j} = \frac{1+\dfrac{F_j}{QR_j}}{1+\dfrac{F_i}{QR_i}}, \qquad \text{for all } i \text{ and } j. \tag{4}$$

In [4] and [5], the assigned power levels satisfy

$$\frac{P_i}{P_j} = \frac{\dfrac{F_j}{QR_j}}{\dfrac{F_i}{QR_i}}, \qquad \text{for all } i \text{ and } j \tag{5}$$

which is slightly different from (4). In [4] and [5], a user's transmitted power is considered part of the interference to its own signal. Consequently, their derivation gives a lower bound of system capacity. According to the numerical results presented in Section VI, the system capacity could be significantly larger than the lower bound if the effect of multipath fading can be removed.

IV. Admission Criterion

In this section, we first derive an admission criterion for all users to satisfy their QoS requirements at all times. The criterion is then generalized to a system which allows a small probability to violate QoS requirements.

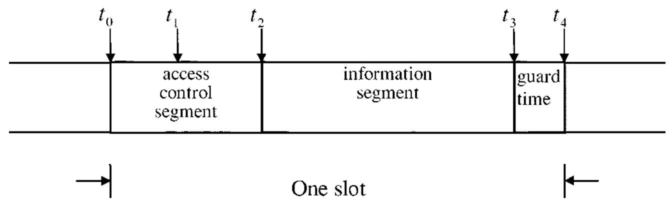

Fig. 2. Time slot structure.

TABLE I
Implementation Requirements of the Proposed Access Control Schemes

Scheme	Implementation requirements
SP/ND	• need a field in every packet to carry next deadline • need to categorize users • base station maintains deadlines of users
SP	• need to categorize users • base station maintains deadlines of users
DP/ND	• need a field in every packet to carry next deadline • base station maintains deadlines of users
DP	• base station maintains deadlines of users

Theorem 2: Given $k_1, k_2, \cdots$, and k_M. There exists a power vector $\vec{P} = (P_1, P_2, \cdots, P_M)$ such that

$$\sum_{j=1}^{M} k_j P_j \le P_i \frac{1+\dfrac{Fi}{QR_i}}{1+f}, \qquad \text{for all } i, \qquad 1 \le i \le M$$

if and only if

$$\sum_{j=1}^{M} \frac{k_j}{1+\dfrac{F_j}{QR_j}} \le \frac{1}{1+f}.$$

Proof: Assume there exists a power vector $\vec{P} = (P_1, P_2, \cdots, P_M)$ such that

$$\sum_{j=1}^{M} k_j P_j \le P_i \frac{1+\dfrac{F_i}{QR_i}}{1+f}, \qquad \text{for all } i, \qquad 1 \le i \le M.$$

According to Theorem 1, one can choose a power vector $\vec{P} = (P_1, P_2, \cdots, P_M)$ such that

$$\frac{P_i}{P_j} = \frac{1+\dfrac{F_j}{QR_j}}{1+\dfrac{F_i}{QR_i}}.$$

As a consequence

$$\sum_{j=1}^{M} k_j P_j \le P_i \frac{1+\dfrac{F_i}{QR_i}}{1+f}$$

TABLE II
PARAMETERS OF AN EXPERIMENTAL SYSTEM (AN INTEGRATED VOICE/DATA CELLULAR DS CDMA SYSTEM)

Chip rate	1.2288 Mbps
Relative other-cell interference factor	0 (single cell) / 0.55
Voice service parameters	
Data rate	9.6 Kbps
QoS requirement ($\frac{E_b}{I_0}$)	7 dB
Mean residence time in ON state	1.0 s
Mean residence time in OFF state	1.35 s
Data service parameters	
Data rate	38.4 Kbps
QoS requirement ($\frac{E_b}{I_0}$)	9 dB
Mean residence time in ON state	0.1 s
Mean residence time in OFF state	1.0 s
Access control schemes	
Cell size (radius)	10 Km
Slot time	20 ms
Access control segment	0.107 ms
Guard time	0.033 ms
Delay bound requirements	
voice service	40 ms
data service	200 ms

implies

$$\sum_{j=1}^{M} k_j \frac{P_j}{P_i\left(1+\dfrac{F_i}{QR_i}\right)} \leq \frac{1}{1+f}$$

which in turn implies

$$\sum_{j=1}^{M} \frac{k_j}{1+\dfrac{F_j}{QR_j}} \leq \frac{1}{1+f}.$$

Conversely, assume that

$$\sum_{j=1}^{M} \frac{k_j}{1+\dfrac{F_j}{QR_j}} \leq \frac{1}{1+f}.$$

Select arbitrarily a power level P_1 for type 1 users.

Let

$$P_i = P_1 \frac{1+\dfrac{F_1}{QR_1}}{1+\dfrac{F_i}{QR_i}}, \qquad \text{for all } i, \qquad 2 \leq i \leq M.$$

As a consequence, we have

$$\frac{P_i}{P_j} = \frac{1+\dfrac{F_j}{QR_j}}{1+\dfrac{F_i}{QR_i}}$$

and thus

$$\sum_{j=1}^{M} \frac{k_j}{1+\dfrac{F_j}{QR_j}} \leq \frac{1}{1+f}$$

implies

$$\sum_{j=1}^{M} \frac{k_j P_j}{P_i\left(1+\dfrac{F_i}{QR_i}\right)} \leq \frac{1}{1+f}$$

which in turn implies

$$\sum_{j=1}^{M} k_j P_j \leq P_i \frac{1+\dfrac{F_i}{QR_i}}{1+f}, \qquad \text{for all } i, \qquad 1 \leq i \leq M.$$

This completes the proof of Theorem 2.

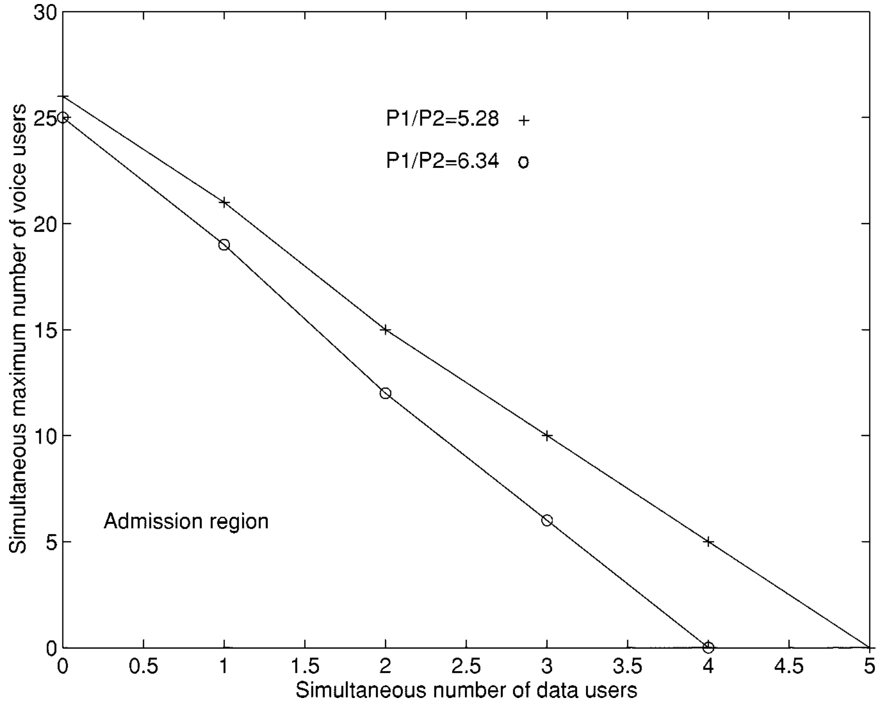

Fig. 3. Comparison of admission regions for $f = 0$.

As a result of Theorem 2, the admission criterion for all users to meet their QoS requirements at any time is

$$\sum_{j=1}^{M} \frac{k_j}{1+\dfrac{F_j}{QR_j}} \leq \frac{1}{1+f}. \tag{6}$$

The same criterion was also reported in [8] for two types of users in an MC-CDMA system. In [8], the criterion was obtained by setting $(E_b/I_0)_1 = QR_1$ and $(E_b/I_0)_2 = QR_2$. Theorem 2 formally generalizes the criterion to M types of users.

Obviously, more connections can be accommodated if the QoS requirements can be violated with a certain small probability. Let N_i denote the population size of type i users with activity factor α_i. Assuming all users are independent, the instantaneous number of type i users in the ON state is then a binomial distributed random variable $\mathbf{K}_i$ whose probability density function is given by

$$\text{prob}(\mathbf{K}_i = k_i) = \binom{N_i}{k_i} \alpha_i^{k_i}(1-\alpha_i)^{N_i-k_i}. \tag{7}$$

Let p_{failure} denote the probability that the requested QoS are not satisfied. That is,

$$p_{\text{failure}} = \text{prob}\left(\sum_{j=1}^{M} \frac{\mathbf{K}_j}{1+\dfrac{F_j}{QR_j}} > \frac{1}{1+f}\right). \tag{8}$$

Numerical results presented in Section VI show that significantly more connections can be accommodated if p_{failure} is allowed to be 0.01.

V. Access Control Schemes

In addition to a specified SIR, an application may request a maximum packet delay bound or an expected maximum packet delay. Obviously, if inequality (6) is used as the admission criterion, then all packets can be delivered to the base station in a single transmission. However, since the users are not always active, the bandwidth can be under utilized most of the time. In this section, we propose AC schemes to increase bandwidth utilization. Again, we assume there are M types of users. An integer D_k, called the delay bound requirement for convenience, is associated with type k users.

We classify applications into loss applications and lossless applications. If type k users belong to loss applications, then D_k represents a stringent maximum delay bound of all packets generated by any type k user. In other words, a packet generated by a type k user in slot m has to be delivered to the base station no later than slot $m + D_k$. Otherwise, the packet is useless and discarded by the user. Most voice users can be classified as loss applications. On the other hand, if type k users belong to lossless applications, then D_k represents the expected maximum packet delay and no packet will be discarded even if a packet is not delivered within D_k slots after its generation. If delay is not a concern of type k users, then one can set $D_k = \infty$. Clearly, most data users can be treated as lossless applications. Without loss of generality, we assume that $D_1 \leq D_2 \leq \cdots \leq D_M$. In this section, two users are in the same type if and only if they both belong to loss applications or lossless applications and have identical data rates, SIR requirements, and delay bound requirements.

We assume that every admitted user is allocated a unique PN code and there is a correlator at the base station for each user. As a consequence, the base station can tell which users sent packets even if inequality (6) is violated because the output power level of a correlator for a user in the ON state is much greater than

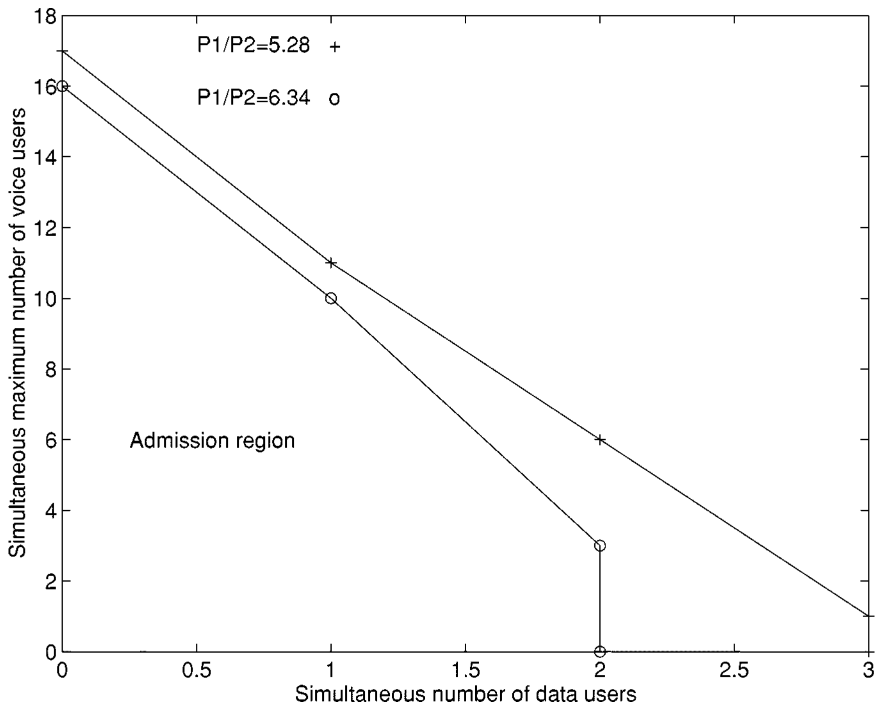

Fig. 4. Comparison of admission regions for $f = 0.55$.

that for the same user in the OFF state. We further assume that packets generated by the same user are transmitted to the base station in accordance with their order of generation. Suppose that the head-of-line (HOL) packet of a type k user was generated in slot m and the current time is slot n. The residual life of the HOL packet is defined to be $m + D_k - n$ and is called the deadline of the user for convenience. Moreover, the residual life of the packet right after the HOL packet is called the next deadline of the user. The next deadline is equal to D_k if the user has only one packet to be transmitted.

The slot format of our proposed AC schemes is illustrated in Fig. 2. At the beginning of a slot, every user who has packets waiting for transmission sends a short message (say, 50 chips) in time period t_0 to t_1. The base station detects those users who sent messages and selects from high priority to low priority as many users as possible without violating inequality (6). The proposed AC schemes differ in priority assignment and will be discussed later. After selection, the base station sends acknowledgments back to the users via the forward channel. This is performed in time period t_1 to t_2. Those users who receive a positive acknowledgment transmit a packet in time period t_2 to t_3. A guard time of duration $t_4 - t_3$ follows packet transmission. The process is repeated for every slot. It is clear that inequality (6) is always satisfied in time period t_2 to t_3 for the proposed AC schemes. In the following, we describe two priority assignment algorithms. For ease of description, we say a collision occurs if inequality (6) is not satisfied in time period t_0 to t_1. Similar AC schemes which totally eliminate packet loss due to collisions can be found in [17] and [18].

A. Static Priority Assignment

In the static priority (SP) assignment scheme, priorities are assigned to different types of users based on their delay bound requirements. That is, any type i user is assigned a higher priority than any type j user if $D_i < D_j$. Without loss of generality, we assume that a type i user has a higher priority than a type j user if $i < j$. For two users of the same type, priorities are assigned according to their deadlines such that the user with a smaller deadline is assigned a higher priority. In case there is a tie, the priorities are assigned randomly. We assume every HOL packet carries the next deadline and base station maintains the deadline of every user. Initially, the deadline for a type i user is set to D_i. If a user is involved in a collision, then its deadline is reduced by one. After a successful transmission, the deadline is replaced with the next deadline carried in the HOL packet.

It is possible that the next deadline cannot be carried in the HOL packet because of the limitation of packet format. In this case, the next deadline is estimated to be the delay bound requirement. For convenience, the SP scheme with and without the next deadline carried in the HOL packet will be referred to as SP/ND and simply SP, respectively.

B. Dynamic Priority Assignment

In the dynamic priority (DP) assignment scheme, all users are treated as if they all are of the same type but with different delay bound requirements. As a consequence, priorities are assigned on packet level. Again, ties can be broken randomly. As in the SP scheme, base station maintains the deadline of each user and every HOL packet carries the next deadline. The deadline of a user is reduced by one if it is involved in a collision and is replaced with the carried next deadline after a successful transmission.

The next deadline is again estimated to be the delay bound requirement if it cannot be carried in the HOL packet. The dynamic priority assignment scheme with and without next dead-

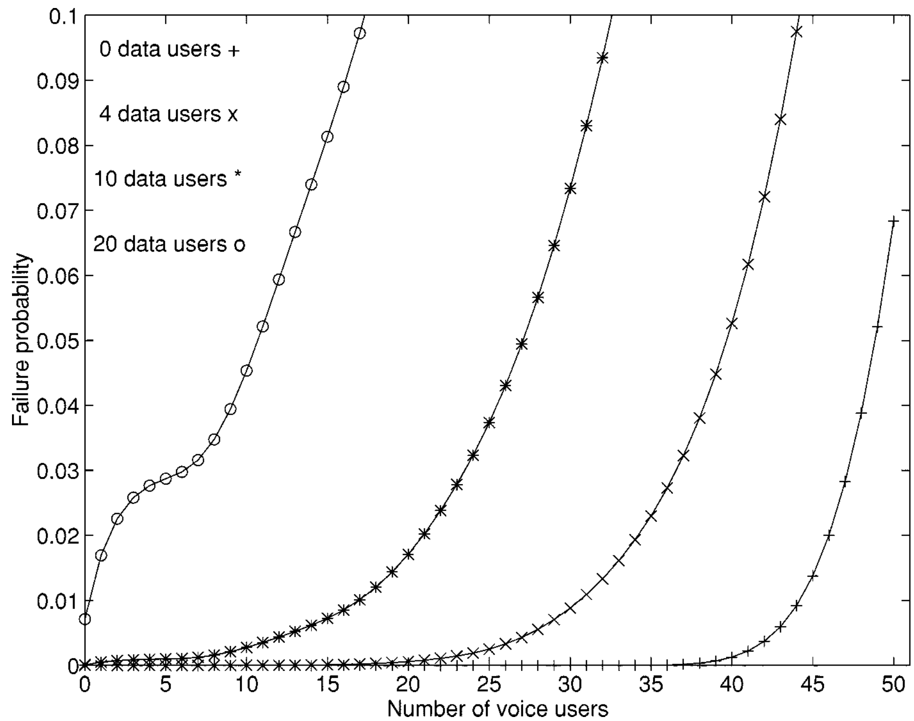

Fig. 5. Failure probability p_{failure} against number of voice users for $f = 0$.

line carried in the HOL packet will be referred to as DP/ND and DP, respectively.

It should be pointed out that, for all the four schemes, a type i user has a smaller packet loss probability than a type j user if they both belong to loss applications and $i < j$. Moreover, the packet loss probabilities of two loss application users of the same type are expected to be close to each other. Similarly, the average packet delay of a type i user is smaller than that of a type j user if both belong to lossless applications and $i < j$; and two lossless application users of the same type are expected to have roughly the same average packet delay. Implementation requirements of the above AC schemes are summarized in Table I.

VI. Numerical Examples

In this section, we study an integrated voice/data system. The parameters of the studied system is depicted in Table II. The chip rate is chosen to be 1.2288 Mbps, the same as that of current IS-95 cellular CDMA system. Relative other-cell interference factor f is chosen to be 0 (for single cell) or 0.55 (for multiple cells). The characteristics of voice and data services used in [5] and [8] are adopted here. If inequality (6) is to be satisfied at any time for $f = 0.55$, then this system can accommodate simultaneously at most 17 voice users in one sector without any data users or at most three data users without any voice users. Let data users and voice users be type 1 and type 2 users, respectively. In the following, the discussions are divided into admission criterion and access control schemes.

A. Admission Criterion

Since data users and voice users are considered as type 1 and type 2 users, respectively, P_1/P_2 is equal to 5.28 if (4) is satisfied or 6.34 if (5) is satisfied. In Figs. 3 and 4, we plot the admission regions for both power assignment algorithms for $f = 0$ and 0.55, respectively. As mentioned before, the curve for $P_1/P_2 = 5.28$ is an "upper bound" and that for $P_1/P_2 = 6.34$ is a "lower bound" of admission region. As an example, in Fig. 4, the system can accommodate six voice users and two data users in one sector if $P_1/P_2 = 5.28$ or three voice users and two data users if $P_1/P_2 = 6.34$.

In Figs. 5 and 6, we plot p_{failure} against the number of voice users for various number of data users for $f = 0$ and 0.55, respectively. In Fig. 6, for $p_{\text{failure}} = 0.01$, this system can support 27 voice users in one sector without any data user which is about 60% improvement over that for $p_{\text{failure}} = 0$. Similarly, if there are no voice users, this system can support ten data users in one sector, a big improvement compared with three for $p_{\text{failure}} = 0$.

B. Access Control Schemes

The parameters used in access control schemes are also shown in Table II. We assume that all voice users are loss applications and all data users belong to lossless applications. Time slot duration T is set to 20 ms. Assume the base station needs 50 chips to detect users in the ON state. As a result, $t_1 - t_0 = 0.041$ ms. The time for selecting users to send messages is expected to be small and thus $t_2 - t_1$ roughly equals maximum round-trip propagation time. The guard time $t_4 - t_3$ is set to the maximum one-trip propagation time to reduce interference from previous slot due to propagation delay. The radius of cells is 10 Km, and thus the maximum one-trip propagation time is 0.033 ms. Consequently, $t_2 - t_1 = 0.066$ ms and $t_4 - t_1 = 0.033$ ms. The total time spent in access control, $t_2 - t_0$, is equal to 0.107 ms, which occupies 0.535% of a slot time.

The locations of the users are uniformly distributed over the cell area and the propagation attenuation is proportional to the

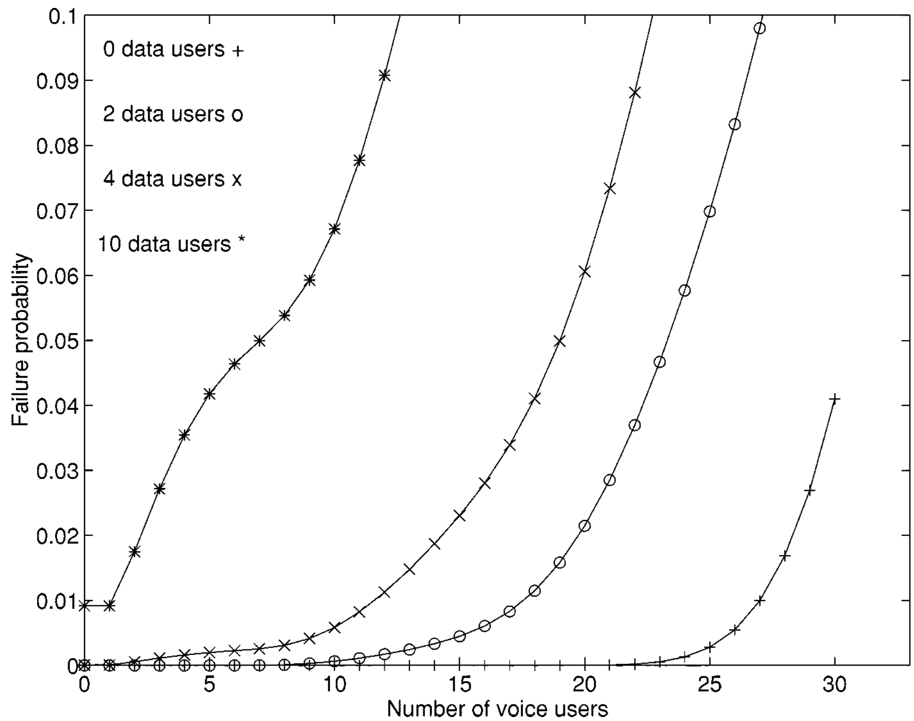

Fig. 6. Failure probability $p_{\rm failure}$ against number of voice users for $f = 0.55$.

TABLE III
PERFORMANCE COMPARISONS OF ACCESS CONTROL SCHEMES FOR $N_v = 25$ AND $N_d = 5$

	SP/ND	SP	DP/ND	DP
Maximum voice packet loss probability	0.000	0.000	0.031	0.030
Minimum voice packet loss probability	0.000	0.000	0.029	0.027
Overall average voice packet loss probability	0.000	0.000	0.030	0.028
Maximum average data packet delay (slots)	64.49	64.60	5.90	7.24
Minimum average data packet delay (slots)	63.25	63.14	5.82	7.14
Overall average data packet delay (slots)	63.68	63.68	5.87	7.20

TABLE IV
PERFORMANCE COMPARISONS OF ACCESS CONTROL SCHEMES FOR $N_v = 15$ AND $N_d = 15$

	SP/ND	SP	DP/ND	DP
Maximum voice packet loss probability	0.000	0.000	0.093	0.080
Minimum voice packet loss probability	0.000	0.000	0.088	0.077
Overall average packet loss probability	0.000	0.000	0.091	0.078
Maximum average data packet delay (slots)	117.26	119.97	6.72	8.18
Minimum average data packet delay (slots)	113.85	115.91	6.57	8.00
Overall average data packet delay (slots)	115.57	117.68	6.64	8.08

fourth power of the distance times a lognormally distributed component with 8-dB differential standard deviation. Here, relative other-cell interference factor f is chosen to be 0.55. The delay bound requirements of voice users and data users are 40 ms (two time slots) and 200 ms (ten time slots), respectively. Any voice packet held beyond two time slots is discarded by its terminal, and data packets will be held until they are successfully transmitted.

All simulations are performed for 10^6 slots with a warm-up period of 10^4 slots. Performance of voice users and data users are measured in terms of packet loss probability and average packet delay, respectively. The 95% confidence intervals are all very small and thus not plotted.

In Tables III and IV, we list some simulation results for $N_v = 25, N_d = 5$ and $N_v = 15$, $N_d = 15$, respectively, where N_v denotes the number of voice users and N_d represents the number of data users. In these tables, maximum (or minimum) packet loss probability is the largest (respectively, smallest) of the loss probabilities encountered by all voice users. Maximum and minimum average packet delay have similar meanings. It can be seen that the difference between maximum and minimum is small. Furthermore, the difference between SP and SP/ND or between DP and DP/ND is also small. However, the average

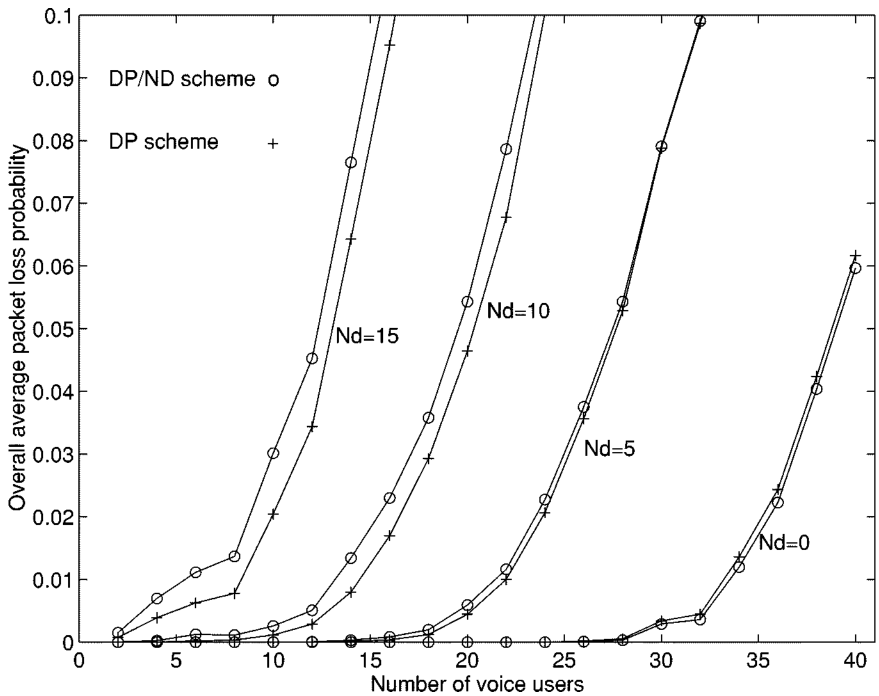

Fig. 7. Overall average packet loss probability of voice users against the number of voice users for DP and DP/ND schemes.

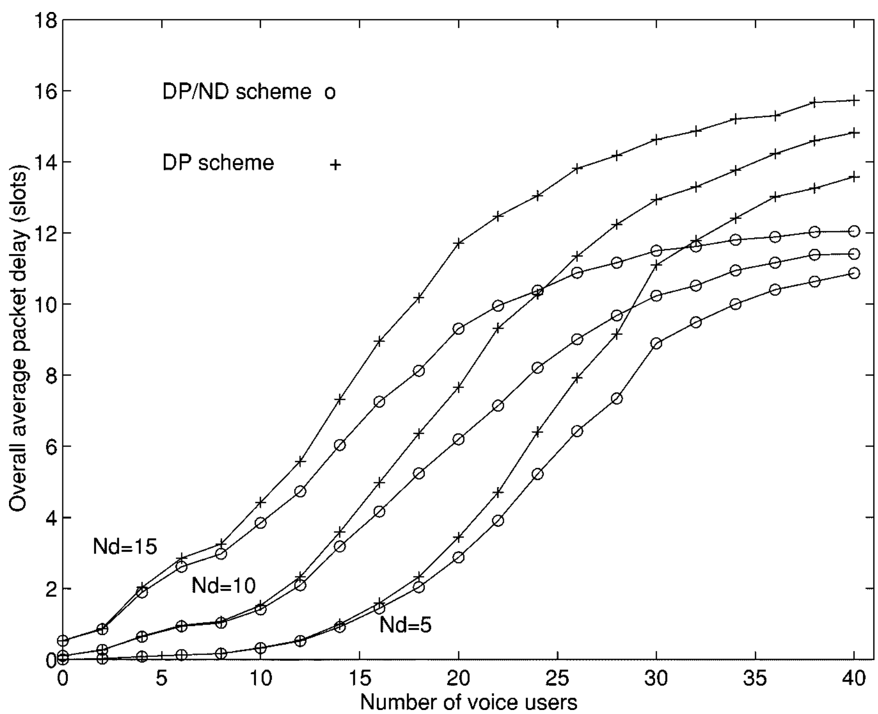

Fig. 8. Overall average packet delay of data users against the number of voice users for DP and DP/ND schemes.

packet delay of data users for DP and DP/ND is much smaller than that of SP and SP/ND. The price to pay is larger packet loss probability for voice users.

In Fig. 7, we plot the overall average packet loss probability of voice users against the number of voice users for various number of data users for DP and DP/ND. It is interesting to note that DP results in a smaller loss probability than DP/ND if there are some data users. This is because voice users are favored in DP. Of course, if there is no data user, then DP/ND results in a smaller loss probability than DP. The packet loss probabilities for SP and SP/ND with or without data users are very close to those for DP and DP/ND with $N_d = 0$ and thus are not presented. The reason is that, for SP and SP/ND, voice users are always assigned higher priority than data users and thus loss probability for voice packets is independent of the number of data users.

In Fig. 8, we plot the overall average packet delay for DP and DP/ND. As expected, DP/ND results in a smaller average delay

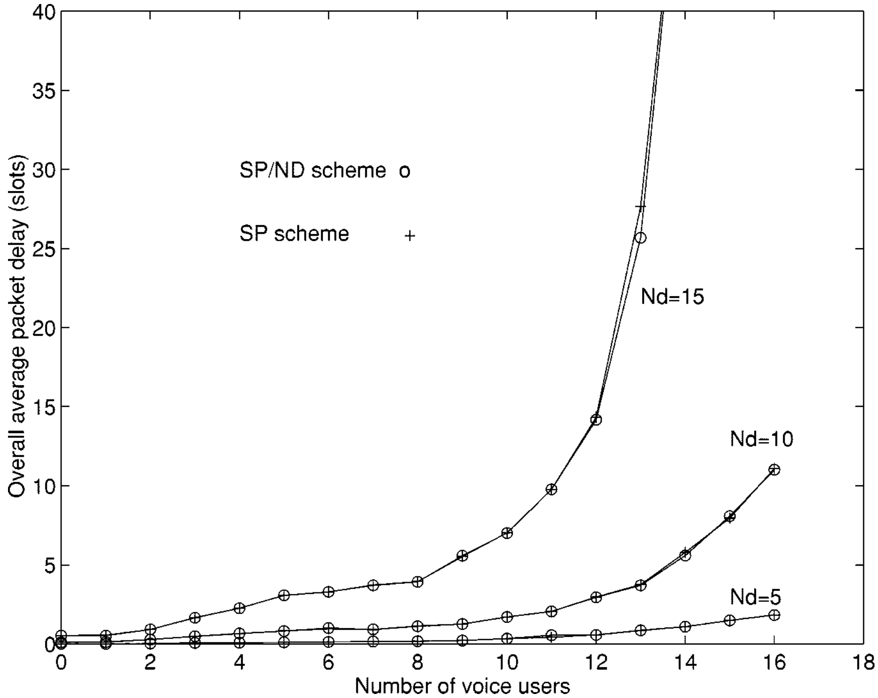

Fig. 9. Overall average packet delay of data users against the number of voice users for SP and SP/ND schemes.

than DP. The maximum difference is about 25%. Notice that the average delay saturates because we specify a finite delay bound requirement for data users and assign priorities on packet level. Fig. 9 shows similar results for SP and SP/ND. The average packet delay may grow unbounded.

For all schemes, simulation results show that the system can accommodate 33 voice users while keeping loss probability smaller than 0.01 when there is no data user present. This is about a 22.2% increase compared with 27 for $p_{\text{failure}} = 0.01$ discussed in part A. Similarly, 27 data users can be supported with average packet delay no greater than ten slots when there is no voice user present.

VII. Conclusion

We have derived in this paper the optimum power vector assignment for a VSG-CDMA system supporting integrated services under the assumption that the effect of multipath fading can be removed. A simple admission criterion is also derived for all users to achieve their QoS requirements simultaneously. Numerical results show that the admission region could be significantly larger than the lower bound presented in [4] and [5]. Besides, more connections can be accommodated if QoS requirements can be violated with a small probability.

To guarantee delivery of packets with time constraints, we propose SP, SP/ND, DP, and DP/ND access control schemes. These access control schemes totally eliminate the possibility of packet loss caused by collision. According to the results obtained by computer simulations, DP seems to be a good choice for an integrated voice/data system. The assumption of $f = 0.55$ in computer simulations may not be valid for mixed traffic because users with higher QoS requirements could lead to a higher f factor to other cells. However, we expect similar results for the proposed access control schemes. Detailed analysis and computer simulations which take this effect into consideration can be further studied.

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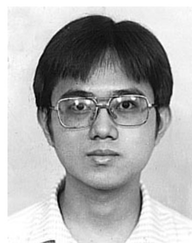

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