A Two level Control Scheme for Data Traffic in CDMA System

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ABSTRACT

Call admission control schemes play an important role in optimizing channel capacity in CDMA systems and guaranteeing quality of service (QoS) for different users. In this paper, we study the limitations of existing call admission control (CAC) algorithms for voice and data traffic having different delay tolerances. We propose an efficient hierarchical access control scheme to administer call admission as well as channel access at the slot level by exploiting disparity present in delay tolerances of voice and data users. Compared to the existing CAC algorithms, the proposed scheme improves the channel utilization by 5%-20%, depending on the packet arriving rates, for traffic with single packet delay tolerance.

I. INTRODUCTION

The third generation (3G) wireless systems, based on CDMA technology, are arriving in the commercial market and have been designed to support data as well as voice traffic [1][2]. Due to bursty nature of data traffic, novel Medium Access Control (MAC) protocols are desirable to improve radio resource utilization in these systems. Several call admission control (CAC) schemes for wireless systems have been proposed in recent years [4][14-18]. Most of these schemes concentrate on power management and interference control to guarantee strict QoS requirement and ignore burstiness that may be present in data traffic. On the other hand, some studies suggest using slot level scheduling to improve data traffic throughput[5][6]. When these independently designed CAC and slot level scheduling schemes are put together, they do not yield optimal performance. This paper develops a hybrid scheme consisting of efficient CAC and slot level scheduling algorithms in a unified framework. An early work in [3] follows this approach, but it assumes zero delay tolerance for voice traffic. Our work studies how to maximize the channel capacity for traffic with different data transmission rates and delay tolerances. We propose computationally efficient call admission control algorithms that maximize the channel capacities for voice and data traffic and then design a slot level control algorithm using the characteristics of the CAC scheme. Our simulation results show that with this unified framework, channel utilization can be improved by 5%-20% for single packet delay tolerant traffic, depending on the packet arrival rate.

The organization of this paper is as follows. In the next section, we introduce channel capacity and traffic models assumed in our work. Section III discusses the relationship between the transmission probabilities and optimized channel capacity and proposes a computationally efficient algorithm for call admission control. In Section IV, the new call admission control and bursty control algorithms for traffic with delay constraints are discussed and novel control schemes are proposed. Simulation results of the proposed algorithm are presented in Section V, and Section V concludes this paper.

II. THE CDMA SYSTEM AND TRAFFIC MODELS

System Model:

We consider the slotted CDMA (MC-CDMA) model for a single sector consisting of *N* users [2]. All the users in the sector share same radio spectrum and use orthogonal codes for identification. So the channel is logically divided into sub-channels by these codes. We assume the system is perfectly power controlled and use a simple threshold model [14] for data link analysis. That is, packets can only be received if the number of active users is less than a receiving threshold. Otherwise all packets are lost. The formula of the capacity is

$$C(N)=\begin{cases} N, & N \le L_{Th} \\ 0, & N > L_{Th} \end{cases}$$

The threshold L_{Th} is a function of the desired SIR requirement E_b/N_0 for successful data packet decode, the processing gain, gain due to sector antenna, and interference from other cells.

Traffic Model:

The arrival process for data packets is modeled based on Bernoulli process. The arrival rate is denoted as v_f packets per slot. The range of v_f is [0,1]. For different users we may have different value of v_f. The probability of *k* packets arriving in T_s slots is given as follows:

$$p(k)=\begin{cases} \binom{T_s}{k} v_f^k (1-v_f)^{(T_s-k)}, & 0 \le k \le T_s \\ 0, & otherwise \end{cases}$$

We discuss two different types of users based on their delay tolerance characteristics. The first type of users (referred to as Type I) has zero delay tolerance and for these users a packet is discarded if it cannot be sent in the immediate slot. The second type of users (referred to as Type II) has fixed delay tolerance, i.e., data can be delayed for up to T_D slots. This stands for data traffic with some delay tolerance, e.g. audio/ video or images with jitter.

For Type I users, the transmission probability, p_0, (here the subscript 0 stands for zero delay tolerance) in a slot is equal to the packet arriving rate, i.e., $p_0 = v_f$

For Type II users with delay tolerance T_D, all the packets that arrive in previous T_D slots are stored in a buffer for transmission. So the maximum buffer length should be T_D. Consider a user being rejected for d slots. The probability for this user transmitting in the next slot is the probability of at least one packet arrives in the rejected slots and is given as follows:

$$p_{T_D}(d) = \begin{cases} 1-(1-v_f)^{d+1}, d \le T_D \\ 1-(1-v_f)^{T_D+1}, d > T_D \end{cases}$$

Here the subscript T_D means the maximal delay tolerance and d in the parentheses stands for the number of consecutive slots a user is blocked. Here we assume the transmission buffer is initially empty.

III. PROPOSED ACCESS CONTROL SCHEME FOR TYPE I USERS

In this section, we study the channel capacity maximization problem for users with different transmission probabilities but zero delay tolerance. Traditionally, the network engineering assumes 100% channel utilization. The number of users to be admitted is computed by [11]:

$$N = L_{Th} / v_f \tag{1}$$

Where v_f is the average arriving rate of all users and L_{Th} is the receiving threshold we mentioned in system model. The admission control schemes based on equation (1) do not optimize the usage of channel capacity since they ignore the burstiness in the data traffic. Let's consider all the users have same transmission probability v_f in a time slot and analyze the Erlang channel capacity C, which can be formulated as:

$$C = N \times v_f \times P_S(N, v_f) \tag{2}$$

where P_S is the probability of successful transmission. i.e. the probability of no more than L_{Th} users transmitting simultaneously.

$P_S(N, v_f)$ is a function of N and v_f.

$$P_S(N,v_f) = \sum_{i=0}^{L_{Th}-1} \binom{N-1}{i} \times {v_f}^i \times (1-v_f)^{N-1-i} \tag{3}$$

By substituting (1) and (3) into (2), we have

$$C = L_{Th} \times \sum_{i=0}^{L_{Th}-1} \binom{L_{Th}/v_f - 1}{i} \times v_f^i \times (1-v_f)^{L_{Th}/v_f - 1 - i} \tag{4}$$

The perfect channel utilization cannot be achieved since $P_S(N, v_f)$ cannot reach 100% if v_f <1

In order to maximize the channel utilization, we need to maximize $P_S(N, v_f)$. For this we proceed by first computing the 1st derivative of $P_S(N, v_f)$

$$P_S(N+1, v_f) = \sum_{i=0}^{L_{Th}-1} \binom{N}{i} \times {v_f}^i \times (1-v_f)^{N-i}$$

$$= \sum_{i=0}^{L_{Th}-1} \frac{N(1-v_f)}{N-i} \binom{N-1}{i} \times {v_f}^i \times (1-v_f)^{N-1-i}$$

$$\le \frac{N(1-v_f)}{N-L_{Tn}} P_S(N, v_f)$$

Then

$$P_S(N+1, v_f) - P_S(N, v_f) \le \left(\frac{N(1-v_f)}{N-L_{Th}} - 1 \right) P_S(N, v_f)$$

and

$$\frac{\partial}{\partial N} C \le v_f \times P_S(N, P) \times \left[1 - N \left(1 - \frac{N(1-v_f)}{N-L_{Th}} \right) \right]$$

$$= v_f \times P_S \times \frac{L_{Th} + N^2 - L_{Th} N v_f - N}{L_{Th} - N}$$

for $N > \frac{L_{Th} + 1}{v_f}$, we have $\frac{\partial}{\partial N} C$ <0

For a fixed value of v_f, the value of N to maximize channel utilization is in the range:

$$L_{Th} \le N < \frac{L_{Th} + 1}{v_f}. \tag{5}$$

The optimal value of N can be searched by comparing all the values in the range specified in equation (5), which amounts to a linear search.

However, this approach cannot be applied to users having different value of v_f. Under this scenario, in order to find out the optimal number of users to be admitted, let us group the users by their v_f values. Let's assume there are K such groups in the system. The number of users in these groups can be described by a vector $\vec{N} = (\mathrm{N}_1, \mathrm{N}_2, \ldots, \mathrm{N}_\mathrm{K})$ and transmission probability $\vec{p} = (\mathrm{p}_1, \mathrm{p}_2, \ldots, \mathrm{p}_\mathrm{K})$, then P_S should be computed for each group and the total channel capacity is

$$C = \sum_{i=1}^{K} N_i \times p_i \times P_{S_i}(\vec{N}, \vec{p}) \tag{6}$$

and $P_{S_i}(\vec{N}, \vec{p})$ is the probability of transmission success for group *i*. Note that in equation (6) $P_{S_i}(\vec{N}, \vec{p})$ is a function of vectors, $\vec{N}, \vec{p}$, and cannot be directly computed using equation (3). We formulate it as follows:

$$P_{S_i}(\vec{N}, \vec{p}) = \sum_{j=1}^{L_{Th}-1} p_{i,j}$$

Where $p_{i,j}$ is the probability of an event viewed by user *i* that altogether *j* other users from all the groups are simultaneously transmitting in the same slot. Numerically, $p_{i,j}$ can be stated as follows:

$$p_{i,j} = \sum_{l_1+l_2+\ldots l_K=j} \binom{N_i-1}{l_i} p_i^{l_i}(1-p_i)^{N_i-1-l_i} \times \prod_{m=1, m\neq i}^{K} \binom{N_m}{l_m} p_i^{l_m}(1-p_i)^{N_m-l_m} \qquad (7)$$

Where l_i is the number of users from group *i* transmitting in a slot. Each time a new user from group *i* with transmission probability of p_i is admitted, the value of $p_{i,j}$ needs to be updated. A brute force method to compute all of $p_{i,j}$ s is given with time complexity of $O(N^3)$ [4]. However, we notice that the new value of $p_{i,j}$ can be computed using the following recurrence relation:

$$p_{i,j(new)} = p_i \times p_{i,j-1(old)} + (1-\ p_i) \times p_{i,j(old)}$$

That is the probability of *j* users transmitting when user *i* has been added can be divided into two parts:

1. probability of *j-1* previously admitted users transmit and user *i* also transmits, or
2. probability of *j* previously admitted users transmit and user *i* does not transmit.

Using an incremental update framework, we can efficiently compute the above recurrence relationship in $O(N^2)$ time, where *N* is the total number of users contesting for call admission. For additional detail we refer to the full length paper [20].

Based on the above discussions, the outline of the admission control algorithm for Type I user having different arrival rates is given below:

Algorithm I:

1. Initialize all $p_{i,j}$ to zero
2. Admit one new user and update $p_{i,j}$ for all *i* and *j*
3. Compute P_{S_i} and *C*
4. If all P_{S_i} satisfy the QoS requirements and C increases, admit this new user and goto step 2, else reject the new user.
5. End.

IV. PROPOSED CONTROL SCHEME FOR TYPE II USERS

For Type II users even when we have determined the number of users to the admitted at the call admission level, their transmission probabilities varies from slot to slot and depend on their respective channel access history and delay tolerance characteristics. This is because if a user is denied channel access in a given slot, its probability of transmission increases in the next slot since data packets starts piling up in the user's transmission buffer. Let's consider Type II users with delay tolerance of one packet. The status of a user can be modeled using a finite state machine shown in Figure 1.

A user in State I correspond to the situation where it has a packet and it will transmit with probability 1. State II stands for a user that was not admitted in the previous slot, so packets are piling up in the buffer. The transmission probability for users in state II is $p_1(1) = 1-(1-p)^2$, as described in section II. Stat III stands for a user that transmits a packet previously and now its buffer is empty. For such a user, the

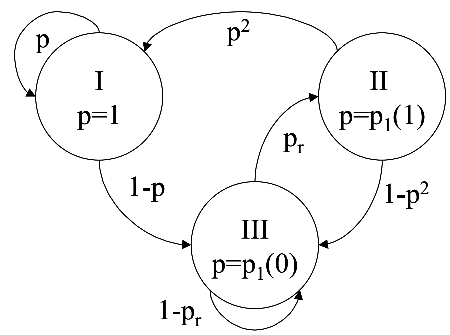

Figure 1: Finite State Machine For Type II user with delay tolerance of one slot

transmission probability for current slot is $p_1(0) = p$. Table 1 summarizes all possible scenarios of state transitions.

Old State	*Buffer Empty?*	*New Packet arrived?*	*Previous Slot Access*	*New State*
I, II	*No*	*Yes*	*Yes*	*I*
I, II	*No*	*No*	*Yes*	*III*
II	*Yes*	*No*	*Yes*	*III*
III	*Yes*	*Yes/No*	*Yes*	*III*
III	*Yes*	*Yes/No*	*No*	*II*

Table I: State Transfer Table of Type II users with delay tolerance T_D = 1 slot.

If a user is blocked, then its state is changed to II. We assume that all the users in states II and I are given access to the channel. Only users in state III can transfer to state II. If a user is admitted and claims that its buffer is empty, then its state is changed to III. Note that failure of a user to transmit a packet implies empty buffer. If a user transmits a packet and its buffer is still not empty, then it goes to state I. In Figure 1,

p_r is the ratio of number of users in state III that are blocked to the number users in State III.

Given the number of users admitted at the call admission level, an intuitive scheme for bursty control is that we maximize the channel capacity slot by slot (say, by using algorithm I again) according to the immediate transmission probability of users for that given slot. However, a local maximum does not always imply global maximal. An alternate scheme is to block certain number of users to access the channel in each slot. Suppose we admit N users and out of which K users are rejected at each slot. We can analyze the performance by studying the state distribution of system in steady state and find the value of K that maximizes the channel capacity. Suppose there are K_1 users in state I, K_2 users in state II, and K_3 users in state III. Then we have

$$K_1 = K_1 \times p + K_2 \times p^2$$

$$K_2 = K_3 \times p_r$$

$$K_3 = K_1 \times (1-p) + K_2 \times (1-p^2) + K_3 \times (1-p_r)$$

$$and \;\; p_r = \frac{K}{K_3}$$

By solving the above equations we have

$$K = K_2 = \frac{1-p}{p^2} K_1 \qquad (8)$$

Assuming the data arriving rate is $p = v_f$, the average channel capacity can be computed as

$$C = K_1 \times p_{g1} + K_2 \times p_1(1) \times p_{g2} + K_3 \times p_1(0) \times p_{g3}$$

p_{g1}, p_{g2} and p_{g3} are the successful transmission probabilities for state I, II and III, respectively. The condition for successful transmission is that the number of all the user transmitting in a slot is less than or equal to the threshold, L_{Th}.

$$P_{g1} = \sum_{i+j \le L_{Th}-K_1} \binom{K_2}{i}\binom{K_3}{j} p_1(1)^i p_1(0)^j$$
$$(1-p_1(1))^{K_2}(1-p_1(0))^{K_3}$$

$$P_{g2} = \sum_{i+j \le L_{Th}-1-K_1} \binom{K_2-1}{i}\binom{K_3}{j} p_1(1)^i p_1(0)^j$$
$$(1-p_1(1))^{K_2-1}(1-p_1(0))^{K_3}$$

$$P_{g3} = \sum_{i+j \le L_{Th}-1-K_1} \binom{K_2}{i}\binom{K_3-1}{j} p_1(1)^i p_1(0)^j$$
$$(1-p_1(1))^{K_2-1}(1-p_1(0))^{K_3-1}$$

Since the total traffic generated by all these users is

$$C_g = N \times v_f \qquad (13)$$

The packet error rate can be estimated as

$$PER = C_g / C \qquad (14)$$

In the following we describe Algorithm II for searching the optimal number of users to be admitted at the call admission level and the appropriate number of users to be blocked per slot to optimize the channel utilization. The algorithm is described as follows:

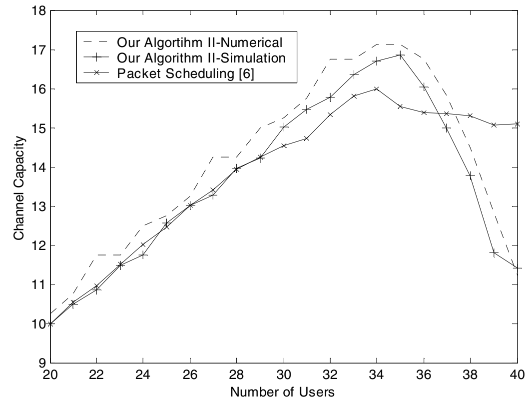

Figure 2: Channel capacity, v_f= 0.5 for Type II users with delay tolerance of one slot, L_{Th}=20

1. Compute probability $p_1(0)$ and $p_1(1)$
2. Choose K_{1max} such that
$$K_{1\max}\left(1 + p_1(1) \times \frac{1-p}{p^2}\right) = L_{Th} / v_f$$
3. For $K=1: K_{1max}$, compute K_1 and K_2 according to equation (8). Search for the optimal number K_3 using Algorithm I. Compute the capacity and PER.
4. Search for (K, K_1, K_2, K_3) which satisfy the PER requirement and have the maximal channel capacity utilization.

For slot level control, in each slot, only N-K_2 out of N users are admitted. We can generalize the above algorithm for Type II users with delay tolerance = M. Due to the page limitation, we leave this derivation for a full-length paper [20].

V. Numerical and Simulation Results

In this section we compare numerically computed performance of the proposed algorithm with simulation results. In all simulations, we model traffic behavior as Bernoulli process. Each simulation data point is averaged over 5000 slots. The proposed algorithms are programmed and simulated using MATLAB and its communication toolbox.

We compare the performance of proposed Algorithm II with existing slot level packet scheduling algorithms that maximize channel capacity slot by slot [6]. Figures 2 and 3 show the numerical and simulation results for the existing and proposed schemes. In our numerical calculation, we fix the packet arrival rate to be 0.5, which is close to the voice activity factor in real world [3]. Figure 2 shows the relationship between the number of users admitted by each scheme and the resulting channel capacity utilization. Figure 3 compares the proposed algorithm with the existing scheme [6]in terms of PER. For the algorithms proposed in this paper , both numerical calculations and simulation results are presented, while for the scheme in [6] only simulation results are reported. As shown

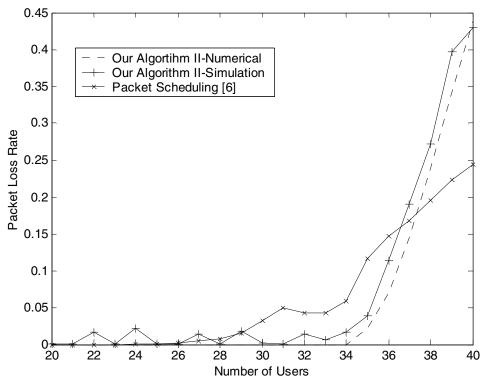

Figure 3: The PER of slot-level burst control, $v_f = 0.5$

in Figure 2, the proposed algorithm outperforms the scheme in [6] when the number of users is between 30 and 36. The proposed scheme achieves the maximal capacity for 35 users and the optimal channel capacity utilization for the proposed scheme is 84%, which is 6% higher than the scheduling scheme in [6]. With maximal channel utilization, the packet error rate of the proposed scheme is 3.8%, while the scheduling scheme in [6] have a PER of 5.8%, as shown in Figure 3.

For a given packet arrival rate, the number of users to be admitted is picked to maximize the channel utilization. Figure 4 shows the simulation results. We find the performance improvement of Algorithm II ranges from 5.2% (packet arriving rate=0.2) to 20.8% (packet arriving rate=0.7).

VI. Conclusion

In this paper we have analyzed the performance of existing call admission control algorithms. We have proposed an efficient hierarchical access control scheme to administer call admission as well as channel access at the slot level by exploiting disparity present in delay tolerances of voice and data users. Compared to the existing CAC algorithms, the proposed scheme improves the channel utilization by 5%-20%, depending on the packet arriving rates, for data traffic with single packet delay tolerance. We expect many superior improvements in channel utilization for data traffic that can tolerate larger delays.

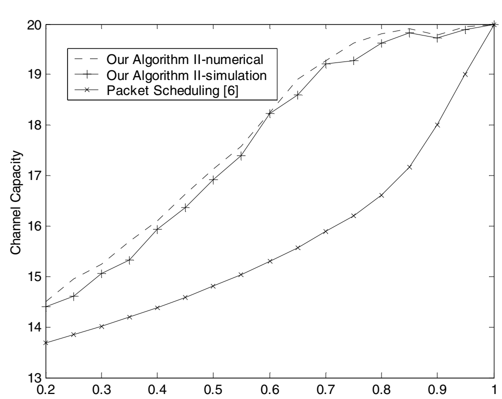

Figure 4: simulation result of Algorithm II with different packet arriving rate, L_{Th}=20

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