An Access Control Scheme for bursty traffic in CDMA System

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***Abstract*- In this paper, we examine the traditional power control and associated access control schemes in cellular CDMA systems and analyze their effectiveness for bursty data traffic. We propose a novel access/admission control scheme based on a probabilistic channel usage framework. Simulation results show that the proposed scheme could improves system capacity considerably without QoS degradation. Compared with existing access control schemes, the algorithm proposed in this paper improves the system capacity by 50% for high traffic loads and by 100% for low to medium traffic loads.**

Keywords-access control; bursty traffic; wireless network; CDMA

I. INTRODUCTION

The last two decades have witnessed a tremendous growth in the use and deployment of wireless devices and networks. Initially 1G and 2G wireless systems were deployed primarily aimed at providing audio services. Now 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]. Due to the bursty nature of the data traffic, novel Medium Access Control protocols are desirable to improve the radio resource utilization in 3G systems. The access or admission control in existing cellular systems is usually studied as part of the power control scheme. Both centralized [2] and distributed power control schemes [3] [4] have been suggested. In most of the existing power control schemes access/admission control is based on the signal to interference ratio *(SIR)* or E_b/N_0. One or more users are removed from the system, or denied access to the radio resource, if a given *SIR* cannot be guaranteed for all existing users. These schemes work well for continuous or long-lasting data streams, such as voice data. However, their efficiency for bursty data traffic needs to be investigated. In this paper, we study the influence of bursty data on the performance of cellular CDMA systems. We propose a novel access control scheme to handle bursty traffic as well as constant rate traffic. We show that by applying the proposed access scheme we can improve the utilization of the radio resource. We also show how the proposed scheme could be combined with the existing power control mechanisms in cellular CDMA systems. The main idea of our proposed scheme is to make the channel access decision taking into account both the system status as well as the probabilistic behavior of users to transmit data. Compared with existing access control schemes [5], the algorithm proposed in this paper improves the system capacity by 50% for high traffic loads and by 100% for low to medium traffic loads.

The organization of the rest of the paper is as follows. In the next section, we analyze the existing power control schemes for bursty subscribers. In Section III we present a novel access/admission control algorithm. The proposed algorithm runs in polynomial time based on the number of users in the system. The simulation results of the proposed algorithm are presented in Section IV and Section V concludes this paper.

II. THE ACCESS CONTROL PROBLEM IN CELLULAR CDMA SYSTEM

In a CDMA system, all the users share the same radio spectrum and use orthogonal codes for identification. Due to the imperfectness of orthogonality, users in a single sector transmitting data simultaneously cause cochannel interference. This interference also enhances the near-far problem [6] in the uplink channel and degrades quality of service (QoS).

The traditional way to calculate the system capacity for data traffic and thus the number of users admitted is based on the following equation [6]:

$$K = \left[\frac{G_p}{E_b / N_0} \right] \times \frac{1}{1+f} \times \eta_c \times \frac{1}{v_f} \times \beta$$

Where G_p is the processing gain by spectrum spreading, f is the interference from other cells, η_c is the power control factor, β is gain due to sector antenna and v_f is the voice/data activity factor. For bursty traffic users, v_f is the average probability of a user to transmit signal during any given time interval. The equation assumes all users having same data/voice activity factor, which is not sufficient for systems supporting both data and voice traffic.

Generally, in the physical layer, a QoS parameter α is defined based on the probability of signal-to-interference-and-noise ratio (*SINR*) greater than or equal to a certain threshold. i.e.

$$\alpha = prob(SINR \geq Th)$$

Let's consider the uplink *SINR* of a single sector CDMA system at the receiver (i.e. base station) side. In one sector, suppose there are altogether *L* users who intend to use the system and only *K* users are permitted to access the radio resource without sacrificing the QoS, where $K \le L$. We assume a synchronized CDMA system with fixed length time intervals. So the *SINR* at a certain time interval *n* for an admitted user *j* is:

$$SINR_j(n) = \frac{P_j(n)G_{jj}(n)}{\sum_{\substack{i=1 \\ i \neq j}}^{K} P_i(n)G_{ij}(n) + N + I} \qquad (1)$$

Where $P_j(n)$ is the transmitting power of user *j*, $G_{jj}(n)$ is the path gain between the user *j* and the base-station receiver, $P_i(n)G_{ij}(n)$ is the interference to user *j* caused by user *i*, *N* is the thermal noise and *I* is the interference from users outside the cell. We assume *N* and *I* are constant during the analysis. In the CDMA system, it is reasonable to assume that the inter-user interference is much larger than the thermal noise, so in the following analysis, we simply ignore this term and use the acronym *SIR* instead of *SINR* to refer to the inter user interference. We also ignore *I* so that we can concentrate on the intra-cell interference. Let s_j denote the signature code of user *j*, then equation (1) could be rewritten as:

$$SIR_j(n) = \frac{P_j(n)G_{jj}(n)}{\sum_{\substack{i=1 \\ i \neq j}}^{K} P_i(n)G_{ii}(n) < s_i, s_j >^2}$$

where $<s_i, s_j>$ is the normalized inner product between two signatures. To simplify the analysis, we assume that the inner product has a constant value, that is:

$$< s_i, s_j >= \delta, i \neq j$$

Thus the *SIR* for user *j* could be formulated as:

$$SIR_j \approx \frac{P_j(n)G_{jj}(n)}{\delta^2 \sum_{\substack{i=1 \\ i \neq j}}^{K} P_i(n)G_{ii}(n)} = \frac{P_{rj}(n)}{\delta^2(P_r(n) - P_{rj}(n))}$$

where $P_{rj}(n) = P_j(n)G_{jj}(n)$ is the receiving power of user *j*, and

$$P_r(n) = \sum_{i=1}^{L} P_i(n)G_{ii}(n)$$

is the total receiving power from all users. For continuous data stream, it is proved in [2] that, with probability one, there exists a unique maximum achievable *SIR* level for any user. Let this optimal SIR level be γ, we have:

$$SIR_{jo} = \frac{P_{rj}(n)}{\delta^2(P_r(n) - P_{rj}(n))} = \gamma$$

So

$$P_{rj}(n) = \frac{\gamma\delta^2}{1+\gamma\delta^2} P_r(n) \qquad (2)$$

In the optimal case, since all the users have the identical *SIR* level, the receiving power for each user must also be identical. So we have

$$P_{rj}(n) = \frac{1}{K} P_r(n) \qquad (3)$$

By substituting (3) into (2) we can find the unique achievable *SIR* level represented in terms of number of users permitted in the sector, and the cross-correlation between two code words, δ. That is

$$SIR_{jo} = SIR_o = \frac{1}{\delta^2(K-1)}$$

So given a certain *SIR* requirement, let's say, *Th*, we can find the maximum number of users that we could admit to the system. Let L_{th} represent this number, then we have

$$L_{th} = \frac{1}{\delta^2 \times Th} + 1$$

Thus the QoS parameter α can be translated as follows:

$$prob(SIR > Th) = prob(K \le L_{Th}) \qquad (4)$$

The threshold *Th* is a function of the processing gain, gain due to sector antenna, imperfectness of power control and interference from other cells. Since we assume *I* is constant for all users, if the level of inter-cell interference *I* is high, we can include it in the calculation of the threshold and thus the above analysis is still valid.

Now let's study the data receiving process for users. To simplify the analysis, we assume a threshold model for the receiving process. That is, a data frame could be successfully received if and only if the *SIR* is greater than or equals to the *Th*. According to the equation 4, a data frame from user *j* can only be received successfully if the user is permitted to access the radio resource and the total number of users accepted is less than or equals to L_{th}. We use $r_j(n)$ to represent the receiving process of user *j*, we have $r_j(n) = 1$ if *j* is admitted and $K <= L_{Th}$, $r_j(n) = 0$ otherwise. Then we measure the system capacity *C* as the average number of data packets we can successfully receive from all the users.

$$C(n) = \sum_{i=1}^{L} E(r_j(n))$$

Our goal is to find *K* that maximizes *C(n)* for every time interval *n*.

If all the users transmit data continuously and the power control is perfect, the maximum capacity for the system is L_{Th} for all *n* and $L = K = L_{Th}$ all the time. However, if data flows from users are bursty, it will be a different story. A bursty user is the one that sends large chunks of data intermittently. This is a very common situation in data communication, which is being envisaged as major traffic in the coming CDMA systems. Moreover, even voice users will generate bursty data streams since the transmission will be suppressed during the silence duration.

If the data source is bursty, then in a certain time interval n all the admitted users might not really have data packets to send. However, the existing access control schemes will still reject the new users although we still have radio resource for them. This results in under utilization of the system capacity. Let L denote the total number of users intending to enter in the system, K denote the number of users in the system at time interval n and M denote the number of users that actually send data packets in that time interval, where $L \geq K \geq M$. It is reasonable to assume that $K > L_{Th}$. However, if K is too large, the probability of $M > L_{Th}$ will also be high thus leading to denial of service. So an access control scheme is needed to find the optimal K that maximizes the system capacity.

Let's consider increasing K by 1 and analyze the change in system capacity. There are 3 cases regarding the status of the users already in the system:

1) $M > L_{Th}$. If there're already more than L_{Th} users to transmit data in the time interval, no matter whether the new admitted user transmits data or not, no data could be received from either this user or the previously admitted users since the *SIR* for every users will be lower than the threshold anyway. Therefore capacity has no change in this case.
2) $M = L_{Th}$. In this case, if the newly admitted user does transmits data in the following time interval, then the *SIR* for all users will fall below the threshold and all $L_{Th}+1$ packets will be lost. So the capacity will decrease to $p(n)L_{Th}$, where $p(n)$ is the transmitting probability for the newly admitted user. Note that if the user is forbidden to transmit, the packet from it will still be lost, so the change is $p(n)L_{Th}$ instead of $p(n)(L_{Th}+1)$
3) $M < L_{Th}$. if the newly admitted user does transmits data in the following time interval then data packets from the newly admitted user as well as from the existing users will be successfully transmitted. The capacity of the system will thus be improved by $p(n)$.

So the total increase in the capacity by increasing K by 1 is:

$$\Delta C(n) = p(n) \times [p(M < L_{Th}) - L_{Th} \times p(M = L_{Th})]$$

So if $p(M<L_{Th}) > L_{Th}p(M=L_{Th})$, adding one more user could improve the total system capacity.

It is convenient to find out the probability $p(M<L_{Th})$ and $p(M=L_{Th})$, by using a recursive algorithm, presented in the next section.

III. The Proposed Access Control Scheme

Based on the discussion in the previous section, we propose the following access control scheme. The purpose of the algorithm is to make the access decision for each user of every time interval so that the system capacity could be optimized. The input parameters of the algorithm are:

a) The number of users, L, attached to the station in the current time interval.
b) The transmitting probability estimation of each of the L users in the next time interval, $p_1(n), p_2(n), \ldots p_L(n)$

Let $p_{I,J}$ denote the probability of J users transmitting data simultaneously given I users are admitted. Then the algorithm could be described as following:

1) Sort the transmitting probability of all the users in descending order, so that let
$$p_1(n) \geq p_2(n) \geq \ldots \geq p_L(n)$$
2) Initialize the algorithm by setting
$$p_{1,0}(n) = 1 - p_1(n)$$
$$p_{1,1}(n) = p_1(n)$$
$$P_{1,J}(n) = 0, J = 2,3,4,\ldots, L_{Th}$$
3) Let $I = 2$;
4) Calculate $p_{I,0}$ to $p_{I,LTh}$ by using the following formulas:
$$p_{I,0}(n) = p_{I-1,0}(n) \times (1 - p_I(n))$$
$$p_{I,J}(n) = p_{I-1,J-1}(n) \times p_I(n) + p_{I-1,J}(n) \times (1 - p_I(n)), J = 1,2,\ldots,L_{Th}$$
This calculation is based on the idea that the event of J users transmitting given I users admitted could be divided into two situations:
    a. J-1 users transmit when I-1 users admitted and the newly admitted user also transmits.
    b. J users transmit when I-1 users admitted and the newly admitted user does not transmit.

    Then the total probability equation used is:
$p=p(a)+p(b)$. We get the formulas for this step since all users transmit independently.
5) Calculate
$$\frac{\Delta C(n)}{p(n)} = \sum_{J=0}^{L_{Th}-1} p_{I,J}(n) - L_{Th} p_{I,L_{Th}}(n),$$
If $\triangle C(n)/p(n) > 0$, then $I = I + 1$, and goto step 4.
6) $K = I$, the number of users allowed to transmit the data in the nth time interval.

This scheme finds the first local maximal value of the channel capacity

$$\arg\max_K C_K(n) = \sum_{i=1}^{K} E(r_j(n))$$

We can easily verify that this algorithm has a polynomial time complexity. In the worst case, it needs to calculate $p_{I,J}$ for I from 1 to L and J from 0 to L_{Th}. In each step there're most 2 addition and 2 multiplications, so altogether we need to perform $2L(L_{Th}+1)$ additions and multiplications to complete the algorithm. So the algorithm has a time complexity no worse than $O(L^2)$ where L is the total number of users in the system.

One issue that still needs to be explained is how to estimate the transmitting probability. This depends on how the data generation process is modeled. If it is modeled as a Poisson process (a common model used in data communication [7]) then during time T (the duration of one time interval), the probability that k packets are generated is:

$$p(k) = \frac{(\lambda T)^k e^{-\lambda T}}{k!}$$

Where λ is a parameter. The expectation value of packets generated in time T is λT. The power control scheme needs to know the parameter λT from higher layer monitors to make an accurate decision. The calculation of transmitting probability also depends on how users handle packets when they are not admitted to transmit. Let's consider the following 2 cases:

1) Users do not buffer the data. That is if one packet has been generated in slot n and this user is not allowed to transmit in the following time interval, then user will discard this packet immediately. Under this assumption the transmitting probability remain the same for a user during the communication.

2) Users have one-packet length buffer. In this case, if a packet is generated and the user is not allowed to transmit it, the user stores the packet and waits for the next time interval. If a new packet is generated before the old packet could be successfully transmitted, then the new packet replaces the old one. So the transmitting probability depends on the access control decision. If a user is not admitted to transmit, then the transmitting probability needs to be increased by λT since data is accumulating at the transmission side.

The second situation is more likely to happen in practice since physical layer need to buffer data anyway before transmitting. However, the first situation is simple for analysis and simulation since it leads to a constant traffic parameter λT.

In general, we can use the transmitting history information to estimate the transmitting probability for each user. The relationship between the length of the history information used to estimate the probability and the accuracy of the probability will be studied in the future work.

IV. THE SIMULATION RESULT

We compare the performance of four access control schemes. First is unbuffered access control scheme of the proposed algorithm. That is there're no buffers at the transmitting side, so if the transmitters are not allowed to send the packet, they just drop it. The second scheme is buffered scheme of the proposed algorithm. We also provide two traditional fixed access control schemes for comparison.

To compare the proposed scheme and traditional scheme, we choose Th so that:

$$L_{th} = \left[\frac{G_p}{E_b / N_0} \right] \times \frac{1}{1+f} \times \eta_c \times \alpha$$

In the first traditional scheme, we set $v_f = 1$, which is equivalent to have a dedicated radio resource for each user so that there is not data loss due to contention. We refer to this scheme as "constant user" scheme, since the number of users is set to a constant ($K=L_{Th}$). We also look into a variation of this standard schemes where we set $v_f = \lambda T$, which represents the probability of a user to transmit in a time interval. This can also be related to the voice activity factor. We refer to this scheme as the "constant-traffic" scheme, since in this case the average load generated by the users admitted to the system is constant ($Kv_f=L_{Th}$).

In the simulations, we assume $L_{Th}=20$ and there're L = 40 active users trying to transmit data. The traffic parameter λT varies from 0.1 to 0.9 with a step of 0.1. The average values used in the graphs are based on 5,000 simulation runs.

From Figure 1 we find out that all the three schemes, the constant-traffic, buffered and unbuffered schemes have the same performance under low traffic rates, however, the constant-traffic will cause higher capacity loss under high

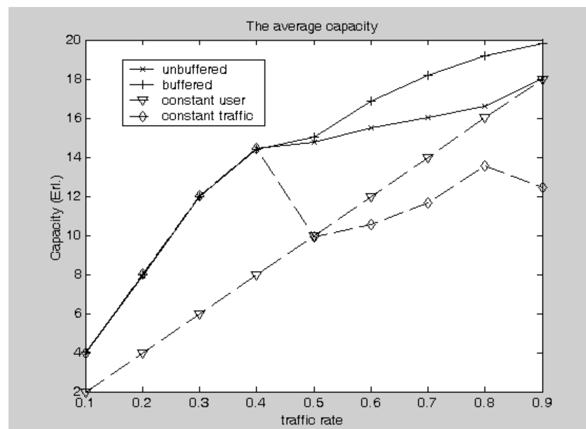

Figure 1. The average capacity of the system

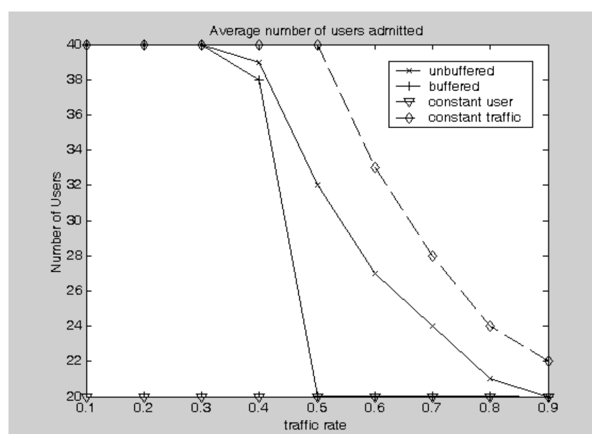

Figure 2. The average number of users admitted to transmit data

traffic rates. Overall, the buffered scheme has the best capacity performance in all cases and its performance almost reaches the upper bond of the system capacity under high traffic rate. (The maximum capacity the system could achieve is $L_{Th}=20\ Erl$.) Figures 1 and 2 show the average capacity and the average number of users admitted, respectively.

For low average traffic rate, the two proposed schemes have a capacity improvement of 100% (0.1) to 80% (0.4) compared to the constant-user scheme. And in high average traffic rate cases (0.5-0.9), two proposed schemes have much better performances. The buffered scheme has an improvement of 50% (0.5) to 10% (0.9) over the "constant-user" scheme.

Figure 2 shows that the constant-traffic scheme always admits the largest number of users. This leads to a high probability of "collision". Note that the average number of users allowed transmitting data for the buffered scheme is lower than that of unbuffered scheme, although it provide a higher capacity. This is because the transmitting probability is calculated dynamically, so this scheme is also a fair scheme for users.

One more issue we may have interest in is the probability of *SIR* below the threshold. i.e. $p(SIR<Th)$. This is shown in Fig. 3. It is important since the probability is defined as a QoS measurement. Wireless data service providers are not likely to apply a scheme with low QoS even though the scheme could increase the system capacity. In Fig. 3, we can find both buffered and unbuffered schemes have a high QoS performance (The outages due to QoS degrading is less than 10%), The "constant traffic" scheme will lead to bad QoS when traffic is high. The constant-user scheme has the best QoS performance, however, the cost is lower system capacity. The buffered scheme has higher QoS and provides higher system capacity as well.

Note though we assume all users have the same traffic rate, it is not a requirement of the algorithm. The algorithm can handle users with different traffic rates.

V. Conclusion And Discussion

We have shown that the traditional access control schemes are not efficient to handle the bursty data traffic in uplink wireless channels of cellular CDMA systems. We introduced a novel access control scheme to incorporate the bursty usage of the wireless channel. Simulation results have shown that the proposed scheme yields better utilization of the systems capacity without significant QoS degradation. Following are some issues open for future research:

A. Buffered or Unbuffered scheme

It is obvious that the buffered scheme has better performance than the unbuffered scheme. However, the buffered scheme involves dynamic calculation of transmission probability. On the other hand, though the performance of unbuffered scheme is a little bit lower than the buffered scheme, the decision only depends on the average traffic rate and once the decision is made, it will not change rapidly. So this scheme could be applied when a user is trying to access the network. This will simplify the implementation.

B. How to get the traffic rate

Another issue is how to calculate the traffic rate. It could either be provided by users along with the transmission request, or the receiver could monitor the data rate and calculate it based on the observation. Both the solutions require a parameter transfer scheme between physical layer and higher layer protocols.

Future work includes evaluating the influence of imperfect path gain prediction due to fast channel fading on the performance of the proposed scheme, and developing a distributed control scheme to handle different QoS requirement.

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