A Probabilistic Transmission Slot Selection Scheme for MC-CDMA Systems using QoS History and Delay Bound

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Abstract. In this paper, we propose an efficient transmission slot selection scheme for Band Division Multi-Carrier-CDMA (BD-MC-CDMA) systems under the constraints of packet loss and delay bound for each individual session. By utilizing channel dynamics together with the delay deadline and loss history, one can determine whether to transmit or not on each time slot, based on the prediction of future channel variations. In this way, one can enhance the chance for transmitting packets with the best channel quality. To validate the efficiency of the proposed algorithm, we model each sub-band as a discrete time Markov Chain using a finite state Markov channel (FSMC) and derive the criteria required for transmission decision. Simulation results show that our proposed scheme can satisfy quality of service (QoS) requirements for real-time traffic with a minimum use of power, while increasing throughput of non-real-time traffic with the power saved from real-time traffic.

1 Introduction

Recently, Band Division Multi-Carrier-CDMA (BD-MC-CDMA) [1], which is one of the variations of MC-CDMA, has been proposed for high-speed wireless communications. In BD-MC-CDMA systems, the transmitter selects the frequency bands which are relatively under good condition according to feedback information from the receivers, thereby decreases required transmission power for each receiver according to its QoS requirements.

Several scheduling algorithms for Orthogonal Frequency Division Multiplexing (OFDM) systems have been proposed in recent years [2], [3]. A practical dynamic resource allocation scheme for OFDM systems has been investigated in [2]. However, it simply extended proportional fairness (PF) scheduling algorithm for CDMA/HDR (High Data Rate) to multi-carrier systems. A cross-layer adaptive resource allocation algorithm for packet-based OFDM systems has been studied in [3]. However, it concentrated on non-real-time traffic without considering QoS of real-time traffic.

Resource allocation problems in BD-MC-CDMA systems are investigated in Mori and Kobayashi's paper [4]. Mori's scheme selects a frequency band according to the SIR estimation with the pilot signal at mobile station and selects the most efficient transmission time slot using the SIR threshold. Although Mori's scheme provides some interesting and innovative concepts, it incurs the following issues to handle. First, it does not consider the heterogeneous QoS requirements of diverse traffic types in terms of packet loss ratio and delay bound. Second, system resources can be wasted because it transmits the packet regardless of the current channel condition when confronting the risk of a time-out. In other aspects, it is difficult to decide an appropriate threshold value, which is critical for the overall performance of proposed algorithm presented in [4].

In this paper, we propose a novel transmission slot selection algorithm for BD-MC-CDMA systems, which accommodates diverse QoS requirements while minimizing total power consumption. We utilize the opportunity for channel improvement while considering the margin for an individual packet's delay deadline and the individual session's packet loss ratio target.

The remainder of this paper is organized as follows. In Section 2, the system model considered in this paper is given and the architecture of the proposed algorithm is described. In Section 3, we present a description of the novel packet scheduler developed for BD-MC-CDMA systems. The performance of the proposed scheme is investigated in Section 4. Finally, Section 5 gives the conclusions.

2 System Model

2.1 BD-MC-CDMA Systems

A BD-MC-CDMA transmitter spreads the original data stream over multiple sub-carriers using a given spreading code in the frequency domain. Each subband consists of the same number of sub-carriers and the number of sub-carriers corresponds to the length of the spreading code. The base station selects the sub-band with the best channel quality for each user, adopts the appropriate ranking mechanism and transmits data to the mobile station via the selected sub-bands.

2.2 Scheduler Architecture

Figure 1 shows the architecture of the proposed algorithm. The proposed scheduler is composed of three sections. In priority determination section, the priority of the packet is determined. The optimal packet transmission time is determined during the probabilistic transmission slot selection section. In resource allocation section, we allocate the packet to the sub-band with the minimum power requirements. In our scheme, scheduling is performed on a frame by frame basis.

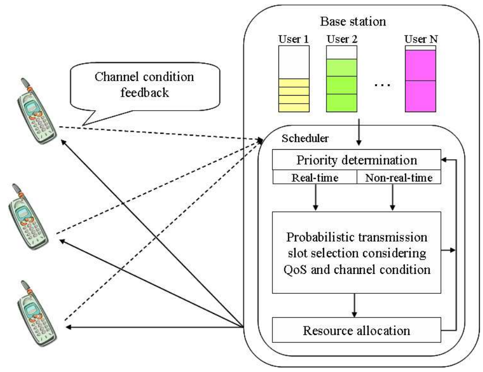

Fig. 1. Architecture of the proposed scheduler

3 Proposed Algorithm

3.1 Priority Determination

In order to provide guaranteed QoS for real-time traffic, a higher priority is always given to real-time traffic by decoupling real-time traffic and non-real-time traffic. The Delay-earliest-due-date (Delay-EDD) [5] and virtual clock [6] are used to determine the packet transmission order for real-time traffic and non-real-time traffic, respectively. It is well known that Delay-EDD scheme can decouple the bandwidth and delay requirements [7]. We choose virtual clock policy for simplicity while providing fairness of non-real-time traffic.

3.2 Probabilistic Transmission Slot Selection

3.2.1 FSMC Model

We assume a slow varying flat Rayleigh fading channel in each sub-band. A slowly varying flat Rayleigh fading channel can be represented as a FSMC model [8]. We assume that the Rayleigh fading channel is slow enough that the channel gain remains unchanged for the duration of a frame. Furthermore, we also assume that each sub-band suffers from independent fading and the fading fluctuation follows an exponential distribution with an average of 1.0.

3.2.2 Evaluation of Channel Quality Improvement

To best utilize channel fluctuation, it is necessary for each user to wait for better channel quality, as long as delay deadline is not expired. Then, the remaining question is how to evaluate the probability that a given channel will ever improve before the delay deadline occurs. Suppose that the current sub-band, which is selected for a certain user to transmit, is in state m as shown in Fig. 2. Then, one can derive the probability that this sub-band will be in a better channel condition more than once before a time-out occurs for the specific packet using the concept of *taboo probability.* A taboo probability, defined on Markov Chains, is the probability of going from one state to another without visiting a particular set of states, known as *taboo states.*

Let $mP_{m,n}^{l}$ be the l-th step transition probability from state state m to n, conditioned on never visiting state m during transitions. In this paper, l-th step can be defined as l-th frame occurring after current frame. Then, one can derive the probability that this sub-band will never be in a better channel condition within the l-th frame, $(m+1)P_{m,n}^{l}$, by denoting states $m+1$,..., $M-1$ as taboo states (see Fig. 2).

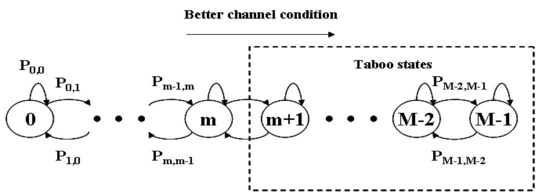

Fig. 2. Illustration of the probability $P_{m,better}^{l}$

Then, $(m+1)P_{m,n}^{l}$, $n = 0$,..., m, is given as:

$$(m+1)\,P_{m,n}^{l} = \begin{cases} (m+1)\,P_{m,0}^{l-1}P_{0,0} + (m+1)\,P_{m,1}^{l-1}P_{1,0}, & n = 0 \\ (m+1)\,P_{m,n}^{l-1}P_{n,n} + (m+1)\,P_{m,n-1}^{l-1}P_{n-1,n} & \\ \quad + (m+1)\,P_{m,n+1}^{l-1}P_{n+1,n}, & 1 \le n \le m-1 \\ (m+1)\,P_{m,m}^{l-1}P_{m,m} + (m+1)\,P_{m,m-1}^{l-1}P_{m-1,m}, & n = m \end{cases}$$

By subtracting $(m+1)P_{m,n}^{l}$ from 1, the probability that a specific sub-band in state m will be in a better channel condition more than once within the l-th frame, $P_{m,better}^{l}$, can be represented as:

$$P_{m,better}^{l} = 1 - \sum_{n=0}^{m} (m+1)\,P_{m,n}^{l}$$

3.2.3 Probabilistic Transmission Slot Selection considering both QoS and Channel Condition

After the initial priority determination, we decide whether to transmit the packet or not during the next frame by using a probabilistic transmission slot selection scheme. In this application, the time-out value represents the number of remaining frames from the current frame before a packet time-out occurs. A time-out value of '1' represents the fact that the next frame is the last chance for transmission. We determine optimal packet transmission time considering three factors: delay bound, packet loss ratio history of an individual session and the probability that the next selected sub-band will be in a better channel condition more than one time before a time-out occurs.

Denote $P^l_{m,better,k}$ as the probability that the k-th sub-band in state m will be in a better channel condition more than once within the l-th frame, P_i as the probability that the next selected sub-band will be in a better channel condition more than once before a time-out occurs, PLR_i as the average packet loss ratio until the previous frame for user i. The expression $TPLR_i$ indicates that the target packet loss ratio of user i, $(Average\ SIR)_i$ represents an average value of SIR in all sub-bands for user i and $SIR_{i,k}$ denotes the SIR value of the k-th sub-band for user i.

$P^l_{m,better,k}$ can be derived in the same way as $P^l_{m,better}$ presented in Section 3.2.2. Then, if the time-out value is l, P_i can be calculated as follows:

$$P_i = \frac{1}{K}\sum_{k=1}^{K} P^l_{m,better,k}$$

where K is total number of sub-bands. Through the priority determination method presented in Section 3.1, suppose that the j^*-th packet of user i^* has the highest priority. If it has a time-out value larger than one, we decide whether to transmit the packet or not during next frame by using the following criteria:

$$MAX\{1 - P_{i^*}, PLR_{i^*}/(TPLR)_{i^*}\} \geq Uniform\ random\ variable$$

$1 - P_{i^*}$ represents the probability that user i^* will never experience a better channel condition before time-out. If $1 - P_{i^*}$ is adequately large, it means that the user i^* will experience little chance for better channel quality than with the current situation existing before delay deadline. If PLR_{i^*} gets closer to $TPLR_{i^*}$, the transmission probability should be increased in order to control the packet loss ratio experienced under target value. To adopt the dominant factor between $1 - P_{i^*}$ and $PLR_{i^*}/TPLR_{i^*}$, we use the max operator between them.

On the other hand, if the j^*-th packet of user i^* with the highest priority has a time-out value equal to one, (i.e., last chance to transmit before deadline), we use a single threshold value. If PLR_{i^*} is greater than or equal to $TPLR_{i^*}$:

$$PLR_{i^*} \geq TPLR_{i^*}$$

Then, the packet should be given a chance to transmit in the next frame in order to meet the anticipated target packet loss ratio. However, if PLR_{i^*} is smaller than $TPLR_{i^*}$, we compare the SIR value of the sub-band k^* of the packet with $(Average\ SIR)_{i^*}$. If SIR_{i^*,k^*} is better than the $(Average\ SIR)_{i^*}$, then

$$SIR_{i^*,k^*} \geq (Average\ SIR)_{i^*}$$

and the average packet loss ratio until previous frame of user i^* satisfies the following equation, then

$$PLR_{i^*}/TPLR_{i^*} \geq Uniform\ random\ variable$$

packet j^* is allocated to the sub-band k^* in the next frame. Otherwise, the scheduler should drop the packet because the packet loss ratio of user i^* has some margin available to the target packet loss ratio and it would be beneficial not to transmit when the channel quality is bad.

3.3 Resource Allocation

If transmission for next frame is decided by the probabilistic transmission slot selection portion, we allocate each packet to the sub-band with the best channel quality, as long as the sub-band is not depleted of available code resources. Consequently, the base station can decrease the transmit power in such selected sub-band. Once the optimal sub-channel is determined, the minimum power level that satisfies the BER requirement of the traffic is assigned. In this way, we iterate on number of packets for resource allocation.

3.4 Aggregated Flow of the Proposed Algorithm

A flow chart providing the detailed process of the proposed algorithm is illustrated in Fig. 3. Assume that the j^*-th packet of user i^* has the highest priority and the k^*-th sub-band in C^l_{remain} is the best channel quality for that packet. Define $Q^l = S^l \cup R^l$, where S^l is a set of the packets waiting to be scheduled in the queue in l-th frame and R^l is a set of already scheduled packets in l-th frame. We define C^l_{remain} to be a set of the sub-bands that has remaining codes in l-th frame.

4 Performance Evaluation

4.1 Simulation Model and Parameters

Three different traffic types are considered. Table 1 summarizes the numerical values used for the simulation model.

- *Voice Traffic*: The voice traffic is modeled as a two state Markov Chain with talkspurts and silence gaps. The average durations of talkspurts and silence gaps are 1.00 and 1.35 s, respectively. The voice packet size is set to 384 bits.
- *CBR Video Traffic*: This model is a constant bit stream with the rate equal to 160 kb/s. The CBR video packet size is set to 1600 bits.
- *Data Traffic*: This model is used to simulate a non-real-time traffic. The bit rate is equal to 250 kb/s and the holding time of each data session is assumed to be exponentially distributed with a mean equal to 20 s. The data packet size is set to 2500 bits.

We consider a downlink BD-MC-CDMA system in single cell environments, thereby, no inter-cell interference is assumed. A slowly varying flat Rayleigh fading channel in each sub-band is assumed. We use an eight-state FSMC model

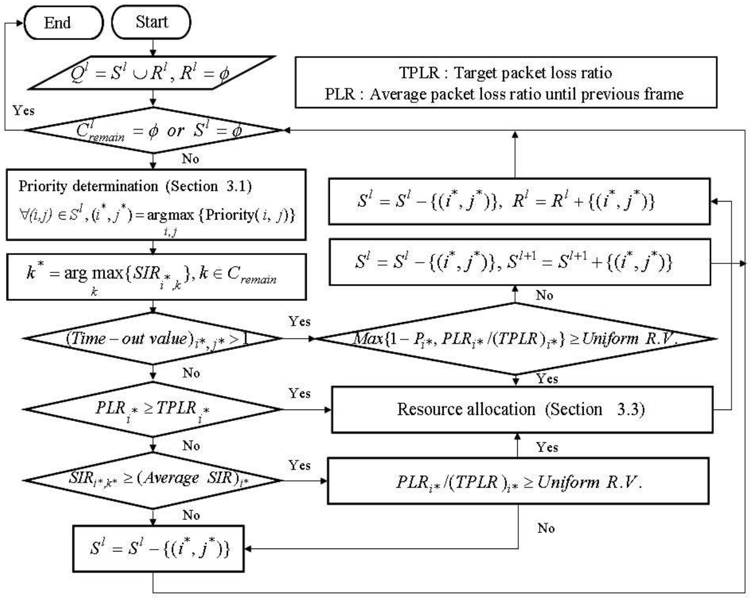

Fig. 3. Flow chart of the proposed algorithm

with the maximum Doppler frequency of 5 Hz and choose an equal probability method for dividing channel gains as described in [8]. Total transmission power of a base station is assumed to be limited. We assume no error in the channel information feedback. The system parameters of BD-MC-CDMA systems, considered in this paper, include frame length, system bandwidth, number of sub-carriers, number of sub-bands and spreading factor as shown in Table 2.

Table 1. Heterogeneous traffic types used in the simulation

Traffic types	Data rate	BER requirement	Delay bound	Packet loss ratio bound
Real-time traffic				
Voice	16.5 kbps	10^{-3}	30 ms	0.01
CBR video	160 kbps	10^{-5}	60 ms	0.01
Non-real-time traffic				
Data	250 kbps	10^{-9}	2 s	0.001

Table 2. System parameters used for BD-MC-CDMA systems

Frame length	System bandwidth	Number of sub-carriers	Number of sub-bands	Spreading factor
10 ms	2.5 Mhz	64	8	8

4.2 Comparison with Existing Algorithms

To evaluate the performance of our proposed algorithm, we compare some very recent algorithms in similar context. The details of each algorithm other than the proposed algorithm are given as follows:

- $Mori's\ Scheme$: This scheme is explained in [4]. In Mori's scheme, no priority determination method is offered. Therefore, we employ the basic FCFS service discipline without decoupling real-time traffic and non-real-time traffic.
- $Wang's\ Priority$: For this scheme, we adopt the priority function proposed in [9]. Other than the priority section, we use our own probabilistic transmission slot selection method and resource allocation scheme. We compared this scheme, especially with respect to the priority determination section. In [9], higher priority is not always given to real-time traffic over non-real-time traffic.
- $No\ PLR\ History$: This scheme uses our priority determination and resource allocation methods, but does not use the probabilistic transmission slot selection scheme. Therefore, the comparison will highlight the effects of our proposed probabilistic transmission slot selection section.

4.3 Simulation Results and Discussion

The average packet loss ratio of voice traffic is shown in Fig. 4. These results show that our proposed scheme can guarantee the desired packet loss ratio even with a very high system load. Mori's scheme does not satisfy target packet loss ratio because it employs a basic FCFS service discipline and thereby fails to decouple real-time traffic and non-real-time traffic.

The advantages of the priority determination section are shown in Fig. 5. Compared with the proposed scheme, Wang's priority does not show good QoS provision when real-time traffic load becomes high, due to the insufficient decoupling of real-time and non-real-time traffic, resulting in a considerably high packet loss ratio for CBR video traffic.

In Fig. 6, we present a throughput comparison of data traffic. Wang's priority scheme has the best performance in throughput. However, Wang's priority scheme and Mori's scheme sacrifice the target packet loss ratio of CBR video traffic as shown in Fig. 5. This figure also demonstrates the performance improvements achieved by the proposed algorithm when compared to the No PLR history scheme in data throughput. When the system load is 0.7, a gain of more than 20% in throughput can be achieved using the proposed scheme compared to the No PLR history scheme.

In Fig. 7, the total power consumption of all traffic is represented. Our proposed scheme consumes the second smallest power levels, next to Mori's scheme. A unit of power represents the amount of transmission power needed to satisfy the BER requirement of one voice packet with a channel gain of 1.0. When the system load is 0.7, about 10% of the power can be saved, when compared to the No PLR history scheme. This is mainly due to an efficient transmission selection strategy adopted in the proposed algorithm.

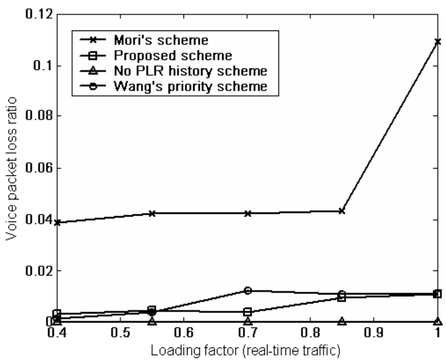

Fig. 4. The average packet loss ratio of voice traffic

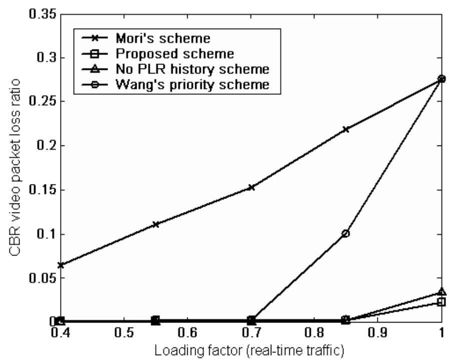

Fig. 5. The average packet loss ratio of CBR video traffic

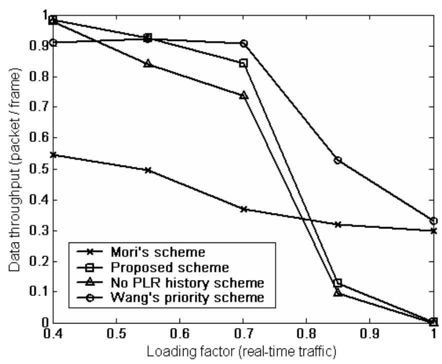

Fig. 6. Throughput of data traffic

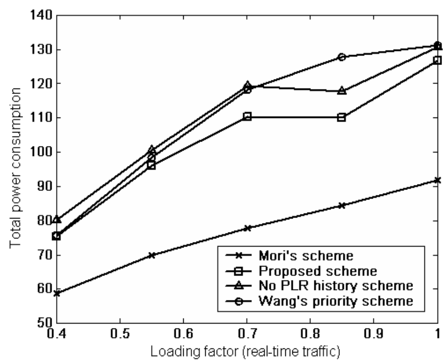

Fig. 7. Total power consumption

5 Conclusion

In this paper, we proposed a novel transmission slot selection algorithm for BD-MC-CDMA systems. By considering delay bound, packet loss ratio and channel diversity altogether, we show that one can find an efficient slot selection criteria, resulting in better QoS provisioning while minimizing total power consumption. Simulation results show that the delay guarantee for real-time traffic can be achieved by the decoupling of traffic types and packet loss ratio can be made under control by the utilizing packet loss ratio history of each session. On the other hand, power minimization can be acquired by efficiently selecting better channel within delay bound. Among all schemes compared in this paper, only the proposed method achieves all the required goals. Our proposed algorithm can be extended to other multi-carrier systems such as OFDM systems.

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References

1. D. Takeda and H. Atarashi, "Orthogonal multicode OFDM/CDMA system using partial bandwidth transmission," IEICE Trans. Commun., vol. E81-B, no.11, pp. 2183-2190, Nov. 1998.
2. L. Xiao, A. Wang, S. Zhou and Y. Yao, "A dynamic resource scheduling algorithm for OFDM system," IEEE Proc. APCC '03, vol. 2, pp.444-447, Sept. 2003.
3. Y. Zhang and K. Ben. Letaief, "Adaptive resource allocation and scheduling for multiuser packet-based OFDM networks," IEEE Proc. ICC '04, vol. 5, pp.2949-2953, Jun. 2004.
4. K. Mori and H. Kobayashi, "Frequency band and time slot selection scheme for downlink packet communication in cellular band division MC-CDM systems," IEICE Trans. Commun., vol. E87-B, no.5, pp. 1114-1122, May. 2004.
5. D. Ferrari and D. Verma, "A scheme for real-time channel establishment in wide-area networks," IEEE J. Select. Areas Commun., vol. SAC-8, PP. 368-379, Apr. 1990.
6. L. Zhang, "Virtual clock: A new traffic control algorithm for packet switching networks," Proc. ACM SIGCOMM'90, pp.19-29, Sep. 1990.
7. S. Keshav, An Engineering Approach to Computer Networking. Reading, MA: Addison-Wesley, 1997, ch. 9.
8. H. S. Wang and N. Moayeri, "Finite-state Markov channel - a useful model for radio communication channels," IEEE Trans. on Veh. Technol., vol.44, No. 1, pp. 163-171, Feb. 1995.
9. X. Wang, "Wide-band TD-CDMA with minimum-power allocation and rate and BER-scheduling for wireless multimedia networks," IEEE/ACM Trans. Networking, vol. 12, pp. 103-116, Feb. 2004.