Real-Time Traffic Packet Scheduling Algorithm in HSDPA System Considering the Maximum Tolerable Delay and Channel Assignment

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Abstract. In this paper, we consider a new packet scheduling algorithm for real time traffic in the high speed downlink packet access system that has been introduced for WCDMA system to providing high transmission rates. The objective of the design is to meet the maximum tolerable delay and consider the channel assigning based on the received SIR for real-time traffic users. The proposed scheduling algorithm shows that the users are ranked by the ratios of the bits in the buffer to the residual time for transmission, the ranked users are assigned channels based on the SIR value table and get service one by one. The simulation results show that the proposed algorithm can provide the lower packet drop rate for real time quality of service (QoS) requirement.

1 Introduction

Wideband Code Division Multiple Access (WCDMA) is the most widely adopted air interface for Third Generation system. One of these technologies is the High Speed Downlink Packet Access (HSDPA), which permits to increase user peak data rates up to 10Mbps, reduce the service response time, and improve the spectral efficiency for downlink packet data service. Its concept consists of a fast scheduling that supports per 2-ms transmission time interval (TTI), adaptive modulation and coding scheme (AMC), fast cell selection (FCS) and multiple input multiple output (MIMO) antenna technology for higher performance.

In HSDPA, fast scheduling is the mechanism determining which user transmits in a given time interval. Maximum system throughput is obtained by assigning all available radio resources to the user with the currently best radio-channel conditions, while a practical scheduler should include some degree of fairness.

In this paper, we propose a QoS guarantee of packet scheduling algorithm considering requirement maximum tolerable delay and backlogged packet in the buffer to decrease the packet drop of real-time service users. In this scheme we get the priority users ranking by the ratio of maximum tolerable delay to backlogged packet in the buffer, after that we assign different number of HS-PDSCH to the service user by the received SIR that can provide the system throughput at the same time.

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2 Related Works

Many wireless packet scheduling algorithms have been designed to support data traffic in the Third Generation Partnership Project (3GPP). Three schemes of packet scheduling are introduced in the HSDPA specification such as Max CIR (maximum carrier to interference), Round robin and Proportional Fairness.

The Max CIR scheduler directs transmission to the user with the momentarily best channel conditions, allowing for the highest possible data rate at each instant and thus maximizing the overall throughput. This serving principle has obvious benefits in terms of cell throughput, although it is at the cost of lacking throughput fairness.

The round robin scheduler cycles through the list of active users and thus is fair in the average sense. As the round robin scheduler is not based on the varying channel quality, the throughput performance however suffers.

The proportional fairness scheduler schedules the user with the currently highest ratio between instantaneous C/I and average transmission rate. It serves in every TTI the user with largest priority:

$$P_i(t) = \arg\max(R_i(t) / r_i(t)), \tag{1}$$

where $P_i(t)$ denotes the user priority, $R_i(t)$ is the instantaneous supportable data rate experienced by user i, and $r_i(t)$ is the user throughput. In the current investigation, the user throughput $r_i(t)$ is simply computed as the number of bits correctly received by user i during the period (t_i, t) divided by such a period, where t_i denotes the instant when the user starts his downlink transmission. This scheme is introduced to compensate the disadvantage of Max CIR and Round robin.

Above schemes are very well suited for non-real time traffic, that only considering the throughput and fairness as QoS requirement, but the transport of real-time traffic over HSDPA is an important challenges in order to guarantee its quality of service (QoS) requirement. Providing QoS, in particular meeting the data rate and packet drop constraints of real-time traffic users, is one of the requirements in emerging high-speed data network.

3 Proposed Algorithm

In this section, we present some basic concepts and definitions, and then described in detail the steps to operate our proposed algorithm.

3.1 Minimum Requirement of Bit Rate for Maximum Tolerable Delay

For each user i with the total length of $L_i(t)$ bits for the backlogged packet in the buffer at time slot t, $T_{\max}$ is the maximum tolerable delay for real-time traffic waiting in the buffer, $W_i(t)$ is the waiting time for a head of line (HOL) packet for user i in each buffer. A minimum requirement bit rate at slot time t is defined as

$$P_i(t) = \frac{L_i(t)}{T_{\max} - W_i(t)} \tag{2}$$

From a conceptual perspective, the minimum requirement of bit rate can guarantee the transmission of the backlogged packets in buffer transmitted without packet drop. That is, if a user wants to transmit packets without packets drop, the user must conduct in accordance with the minimum requirement of bit rate of transmission in next several time slots.

3.2 Channel Assignment in HSDPA

Scheduler in HSDPA system uses HS-DSCH for high transmission rate in downlink case. HS-DSCH is consisting of 15 the real channels (HS-PDSCH) that can be assigned to the service users in one time slot.

In previous scheduling algorithm, 15 channels could be assigned to a service user selected by the scheduler in each time slot. Here we assign different number of channels to multi-users depending on the SIR table. Scheduler collects all SIR values of each user in CPICH to select service users for the next time slot. Table 1, is the assigned channel number based on received SIR. In this case, even highest-priority user in a poor channel state can still guarantee the transmission of information. When the channel state changes better, the user will get more channels to complete the information transmission.

Table 1. SIR table for number of assigning channe

SIR level	Assigned channel numbers
SIR ≥27dB	15
27dB> SIR≥22dB	9
22dB> SIR≥16dB	6
16dB> SIR	4

3.3 Proposed Scheme Procedure

In this paper, we consider the real-time traffic in HSDPA system. Our design is to decrease the packet drop of real-time service users ranking by the minimum requirement of bit rate transmission. For providing the system throughput at the same time, we assign different numbers of HS-PDSCH to the service users by the received SIR value. Figure 1 shows the proposed scheme process.

Step1: We need to calculate the minimum requirement on bit rate as a priority ranking standards. In this process, if greater the amount of data storage in the buffer, the

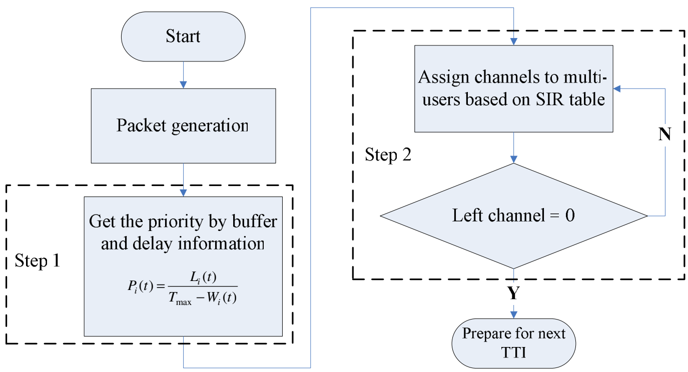

Fig. 1. Proposed scheme flow chart

remaining service time is shorter; the user will be in a higher priority in the ranking. Otherwise, the user would get service later.

Step2: Scheduler defines the SIR level using the table for the assigning channel number of the highest priority user. When the highest priority user gets service with the defined number of channels, the residual channels will be assigned to the secondary priority user based on the table. Step2 will be repeated until the residual amount of channel is 0.

This scheme guarantees the higher priority and SIR value user transmitted with more number of channels, the difference with previous schemes is the user with lower SIR value but higher priority also could get channels for service, thus avoiding packet drop by the long time delay and buffer overflow.

4 Performance Analysis and Evaluation

4.1 Simulation Environment

In this paper, the simulation system is based on the 3GPP standards. Table 2 gives the major simulation parameters.

This subsection describes the configuration of the system level simulation. We employed a 3-sectored 19-cell model. In each sector, the distribution of topography and constriction is basically the same. The correlation coefficient between cells sites and that between sectors were 0.5 and 1.0. The location of each user was randomly assigned with a uniform distribution within each cell. Once the simulation begins, the location of all users is fixed. The propagation model between the base station and mobile station is $128.1 + 37.6\log(R)$, here $R(Km)$ is the distance between base station and mobile station, lognormal shadowing with a standard deviation of 8dB and instantaneous 12 multi-paths fading.

Table 2. Simulation Parameters

Parameter	Value
Cell layout	19 cells, 3sector/cell
User distribution	Uniform
Cell radius	1Km
BS total Tx power	17W
Standard deviation of shadowing	8dB
Correlation between sectors	1.0
Correlation between cells	0.5
Number of paths	12 paths
Hybrid ARQ scheme	Chase combing
Carrier frequency	2000MHz
The number of users	Fixed (100 to 500 real-time traffic user)

Table 3. Modulation and Coding Scheme

MCS level	Coding rate	Modulation	Data rate
1	1/4	QPSK	1.2Mbps
2	1/2	QPSK	2.4Mbps
3	3/4	QPSK	3.6Mbps
4	3/4	8PSK	5.4Mbps
5	1/2	16QAM	4.8Mbps
6	3/4	16QAM	7.2Mbps
7	3/4	64QAM	10.8Mbps

Meanwhile, we applied AMC in the radio link level simulation, which controls the MCS according to the average received SIR over one TTI. In the 7 MCS levels, in table 3, the MCS used in this paper is mcs2, mcs5, mcs6 and mcs7. For FCS case, the user will chose 3 cells with the most SIR values as the active set; the most one of the 3 cells will get service. In each cell, the number of the service provider is same.

To implement the HSDPA feature, the HS-DSCH (High Speed Downlink Shard Channel) is introduced in the physical layer specification. HS-DSCH consists of 15 HS-PDSCH (High Speed Physical Downlink Shard Channel) which are the real channels, and can be assigned to the service users in one time slot. The Transmission Time Interval (TTI) or interleaving period has been defined to be 2ms for the operation between the base station and mobile station.

4.2 Traffic Model

In the paper it is assumed that the modified streaming traffic model is real-time traffic model. Traffic model parameters of an RT streaming traffic are show in table 4.

The size of each packet call is distributed based on the Pareto distribution with the maximum size of m. This probability density function $f_\rho(x)$ is expressed by using the minimum value of the distribution k,

$$f_\rho(x) = \begin{cases} \dfrac{\alpha \times k}{x^{\alpha+1}}, k \le x < m \\ \beta, x = m \end{cases}, \beta = \left(\frac{k}{m}\right)^\alpha, \quad (3)$$

where α = 1.1, k = 4.5 Kbytes and m = 2 Mbytes. Based on these parameters, the average value of the packet call size becomes 25 Kbytes. The reading time is approximated as a geometrical distribution with the average value of 5 sec. The maximum tolerable delay for each packet is fixed as 72ms; the maximum buffer capacity is 450000 bits.

Table 4. Major Traffic Model Parameters

	Distribution	Parameters
Packet calls size	Pareto with cutoff	α =1.1, k =4.5Kbytes m =2Mbytes Average packet call size 25Kbytes
Reading time	Geometric	Average 5 sec
Packet size		12Kbit
Packet inter-arrival time	Geometric	Average 6 ms
Maximum tolerable delay ($T_{\max}$)	Fixed	72ms

4.3 Definition of Performance Indicators

We introduce the concept of the service throughput and packet loss rate for evaluation of system performance.

Service throughput is a ratio between the transmission good bits and the total number of the cell:

$$Service\_throughput = \frac{1}{N_{cell}} \sum_{k=1}^{N_{cell}} Service(k), \quad (3)$$

in the above function, $Service(k)$ are the successful transmission bits per TTI in cell k. It is calculated as follows:

$$Service(k) = \frac{1}{N_{second}} \sum_{i=1}^{N_{second}} N_{good\_bits}(i), \quad (4)$$

here $N_{good\_bits}(i)$ is a successful transmission bit at time slot i and N_{second} is total simulation time.

The real-time traffic packet drop rate is measured as the number of drop packets divided to the total transmission packets

$$packet\_drop\_rate = \frac{drop\_packet}{total\_trans\_packet}, \tag{5}$$

where $drop\_packet$ consists of two parts. One is the packet loss caused by exceeding the maximum tolerable delay and the other is total packet of the user exceeding its buffer capacity.

4.4 Numerical Result and Evaluation

We compare the service throughput and packet drop rate among the proposed algorithm and previous two schemes. An obviously improvement in packet drop rate as well as high throughput can be obtained

Figure 2 shows the HSDPA service throughput versus the number of users. The service throughput of the proposed scheme is increasing with the number of users in the system. The proposed scheme is not much different from the Max C/I and PF schemes.

Figure 3 shows the relationship between number of users and the packet loss rate. The packet loss rate means the ratio of packet exceeding the maximum tolerance delay to the total transmission packets. Considering the 72ms requirement delay, the proposed scheme packet loss rate is lower than the MCI and PF schemes. More number of users is, the higher the performances are.

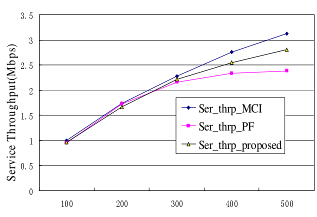

Fig. 2. Throughput performances

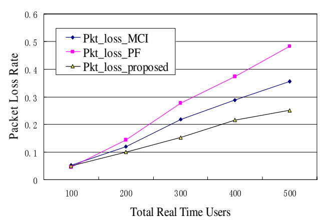

Fig. 3. Packet loss performances

Figure 4 shows the buffer overflow performances. When the buffers of users exceed the maximum buffer capacity, the packets will be lost. In this simulation, the buffer capacity is set 450000bits. In the packet drop statistic, the buffer overflow rate is less influenced than the packet loss rate caused by exceeding the requirement delay, but still need to be considered for real time traffic. The proposed scheme decreases by 0.5% as compared with the MCI scheme and 2% as compared with PF scheme.

Figure 5 shows the packet drop rate as the number of real-time traffic users increases. It is obvious that the proposed scheme outperforms in packet drop rate performance over the other two schemes, especially when the number of users increases.

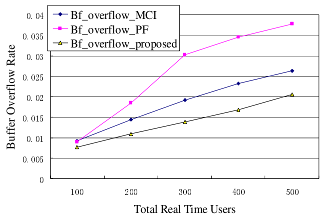

Fig. 4. Buffer overflow performances

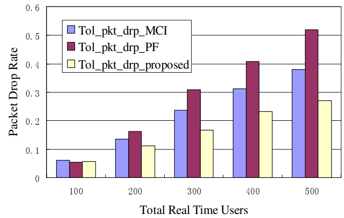

Fig. 5. Packet drop performances

For the 72ms maximum tolerable delay, 450000 bits buffer capacity and 500 users, the packet drop decrease of the proposed scheme is 10% as compared with Max C/I. This improvement is more obvious with the PF schemes.

5 Conclusions

In this paper, we propose a scheduling algorithm for the real-time traffic in HSDPA system for WCDMA downlink case. The proposed scheme can satisfy the QoS guarantee of the real-time traffic for both throughput and packet drop. The simulation results elucidate that although the proposed scheme throughput is between the Max C/I and PF method, it is advantageous in reduction of user packet loss rate and buffer overflow. The last simulation in Fig. 5 is shown the packet drop rate of proposed scheme is reduced by approximately 10% compared to Max C/I method and 15% to PF method.

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