논문 2013-08-38

Packet Scheduling Algorithm Considering Maximum Delay Tolerance for HSDPA System

Soojung Hur, Narzullaev Jakhongil, Yong-Wan Park*

Abstract : In this paper, we consider a new packet scheduling algorithm for real-time traffic in the HSDPA system that has been introduced for the WCDMA system, in order to provide high transmission rates. The objective of the design is to meet the maximum tolerable delay and consider channel assignment 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 as priority order; then the ranked users are assigned certain number of channels based on the SIR value table. The simulation results show that the proposed algorithm can provide a lower packet drop rate, and satisfy real time quality of service (QoS) requirements.

Keywords : packet scheduling, packet drop, maximum tolerance delay, HSDPA.

Ⅰ. Introduction

Wideband Code Division Multiple Access (WCDMA) is the most widely adopted air interface for Third Generation systems[1-3]. Multimedia streaming services are becoming increasingly popular in third-generation (3G) and fourth-generation (4G) mobile networks. One important characteristic of multimedia streaming services is the load asymmetry between downlink and uplink, and the services are mostly used in the downlink. In order to enhance downlink performance, some systems have been proposed. It includes the high data rate system, high-speed downlink packet access (HSDPA), and [3G partnership project technical report (3GPP TR) 25.848; 3GPP TR 25.308] system. These systems introduce many schemes such as adaptive modulation and coding technique (AMC), hybrid automatic rep eat request (HARQ) scheme, the high-speed channel shared by multiple users, quality of service (QoS) control for packet services. One of these technologies is High-speed Downlink Packet Access (HSDPA), which permits the increase of user peak data rates up to 10Mbps, reduces the service response time, and improves the spectral efficiency for downlink packet data service[4, 5]. It consists of fast scheduling that supports the 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, the mechanism for determining which user transmits over a given time interval is fast scheduling. Maximum system throughput is obtained by assigning all available radio resources to the user with the current best radio-channel conditions, while a practical

* Corresponding Author (ywpark@yu.ac.kr)
Received: 21 Mar. 2013, Revised: 16 Apr. 2013, Accepted: 3 Sep. 2013.
S. Hur, N. Jakhongil, Y.W. Park: Yeungnam University
※ This work was supported by the Industrial Strategic technology development program, 10041057 funded By the Ministry of Trade, industry & Energy(MI, Korea)
※ This work was supported by the 2009 Yeungnam University Research Grant

ISSN : 1975-5066
http://dx.doi.org/10.14372/IEMEK.2013.8.6.311

scheduler should include some degree of fairness.

In this paper, we propose QoS guarantee for a packet scheduling algorithm considering the requirement for 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 ranked by the ratio of maximum tolerable delay to backlogged packet in the buffer, after that we assign a different number of HS-PDSCH (High-speed Physical Downlink Shared Channel) to the service user by the received SIR, which can provide the system throughput at the same time.

This paper is organized as follows. In the next section, we briefly survey existing wireless packet scheduling algorithms. The detail of the proposed algorithm is mentioned in section III. In section IV, we provide a traffic simulation model and present the results of performance analysis. The conclusion is given in section V.

II. Relate work

In wireline networks, many packet fair queueing have been proposed which can provide fairness and delay bound. Because there are some characteristics of the wireless network such as location condition dependent error, channel contention, and burst errors, packet fair queueing in the wireline network cannot be applied directly in the wireless network. Several wireless fair queueing [1, 3] have been developed, and they can provide short-term fairness for an error-free system, long-term fairness, graceful degradation in service for flows perceiving clean channel, and delay bound.

In the Third Generation Partnership Project (3GPP) [6], several publications list different HSDPA packet scheduler options, including [7, 8]. There are three schemes of packet scheduling introduced in the HSDPA specification [9], which are the maximum carrier-to-interference power ratio (Max CIR) [5], the Proportional Fairness (PF) method [9] and Round Robin (RR) method [10].

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 operates at the cost of lacking throughput fairness because users under poor average radio conditions are served less frequently.

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 suffers. Hence, a scheduling algorithm that takes both takes channel quality and degree of fairness into account may be desirable.

The proportional fair scheduling idea is to identify the best user for scheduling; a relative channel quality indicator is calculated for each user as the ratio:

$$\text{Scheduling metric} = \frac{\text{User's instantaneously supported data rate}}{\text{User's average served throughput}} \qquad (1)$$

Thus, a user is prioritized either if he/she has good instantaneous conditions compared to the average level or the user has been served with little throughput in the past. So it offers an attractive trade-off between user fairness and system throughput.

The above schemes are very well suited for non-real time traffic, that only consider the throughput and fairness as QoS requirements, but the transport of real-time traffic over HSDPA is an important challenge in guaranteeing quality of service (QoS). 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 networks.

III. Proposed algorithm

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

3.1 Minimum bit rate requirement for maximum tolerable delay

For each user i with 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 ahead 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 bit rate requirement can guarantee the transmission of the backlogged packets in the buffer transmitted without packet drop. That is, if a user wants to transmit packets timely without packet drop, the user must get ordering in accordance with the minimum bit rate transmission requirement as a priority factor that calculated by (2) in the next several time slots.

3.2 Channel assignment in HSDPA

The scheduler in the HSDPA system uses HS-DSCH (High-speed Downlink Shared Channel) to achieve the high transmission rate in the downlink. HS-DSCH consists of 15 real HS-PDSCHs (High-speed Physical Downlink Shared Channel) that can be assigned to the service users in one time slot.

In the previous scheduling algorithm, 15 channels could be assigned to a service user selected by the scheduler in each time slot. Here we assign a different number of channels to multiple-users, depending on the SIR (signal-to-interference ratio) table. The scheduler collects all SIR values of each user

Table 1. SIR table for number of assigning channel

SIR level	Calculated level	Assigned channel numbers
$SIR_i \geq SIR^1_{level}$	SIR≥35dB	15
$SIR^1_{level} > SIR_i \geq SIR^2_{level}$	35dB> SIR≥20dB	9
$SIR^2_{level} > SIR_i \geq SIR^3_{level}$	20dB> SIR≥4dB	6
$SIR^3_{level} > SIR_i$	4dB>SIR	4

in CPICH (Common Pilot Channel) to select service users for the next time slot. Table 1, is the number of assigned channels based on received SIR. The principle of make sure the SIR^N_{level} for different users would meet the given constraints:

$$P(SIR^N_{level} > SIR_i > SIR^{N+1}_{level}) \leq \frac{1}{N}(\%) \tag{3}$$

This means the probability of user numbers in each SIR level area should be almost equity. Here, N is the number of service level. For this paper model case, we calculate the SIR level for number of assigning channel as follow:

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 by ranking users following the minimum bit rate requirement order. 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. Fig. 1 shows the proposed scheme process.

Step1: We need to calculate the minimum bit rate requirement for priority ranking standards. In this process, the greater the amount of data storage in the buffer, the shorter the remaining service time; the user will be of higher priority in the ranking. Otherwise, the user will be provided with service at a later time.

Step2: The scheduler defines the SIR level

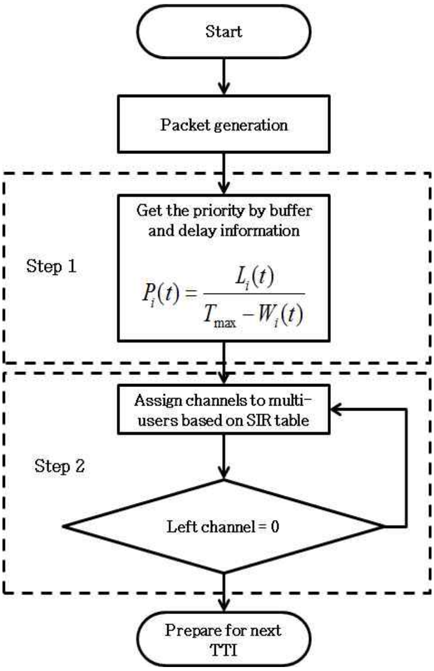

Fig. 1 Proposed scheme flow chart

using the table for assigning the channel number of the highest priority user. When the highest priority user obtains service with the defined number of channels, the residual channels will be assigned to the user with lower priority, based on the table. Step2 will be repeated until the residual amount of the channel are 0.

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

Ⅳ. Performance analysis and evaluation

4.1 Simulation environment

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

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

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 are basically the same. The correlation coefficient between cell 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+37.6log(R), here R(Km) is the distance between the base station and mobile station. The model operates log normal shadowing with a standard deviation of 8dB and instantaneous 12 multi-paths fading.

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

Table 4. Major Traffic Model Paramet

	Distribution	Parameters
Packet calls size	Pareto with cutoff	α=1.1, k=3.6Kbytes, m=2Mbytes Average 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

MCS used in this paper is mcs2, mcs5, mcs6 and mcs7. In the FCS case, the user will chose 3 cells with the highest SIR values as the active set; the cell with the highest SIR value will be provided service. In each cell, the number of service providers is the same.

To implement the HSDPA feature, the HS-DSCH is introduced in the physical layer specification. The HS-DSCH consists of 15 HS-PDSCHs 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 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 a real-time traffic model. The traffic model parameters of RT streaming traffic are shown in Table 4.

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

$$f_\rho(x) = \begin{cases} \dfrac{\alpha \times k}{x^{\alpha+1}}, & k \le x < m \\ \beta = \left(\dfrac{k}{m}\right)^{\alpha,} & x = m \end{cases}, \qquad (4)$$

where α = 1.1, k = 3.6 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 an average value of 5 sec. The maximum tolerable delay for each packet is fixed at 72ms; the maximum buffer capacity is 450000 bits.

4.3 Definition of performance indicators

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

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

$$Service\_throughtput = \frac{1}{N_{cell}} \sum_{k=1}^{N_{cell}} Service(k). \qquad (5)$$

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

$$Service(k) = \frac{1}{N_{\mathrm{second}}} \sum_{i=1}^{N_{second}} N_{good\_bits}(i), \qquad (6)$$

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 by the total transmission packets

$$packet\_drop\_rate = \frac{drop\_packet}{total\_trans\_packet}, \qquad (7)$$

where $drop\_packet$ consists of two parts. One is the packet loss caused by exceeding the maximum tolerable delay and the other is the 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 obvious improvement in packet drop rate as well as high throughput can be obtained

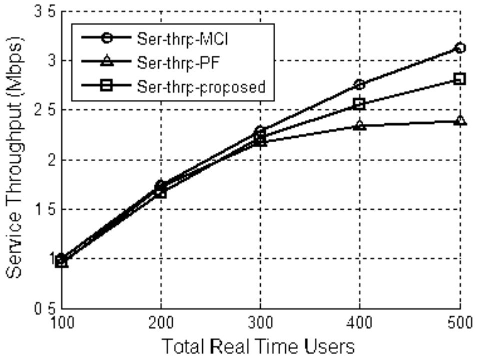

Fig. 2 Throughput performance

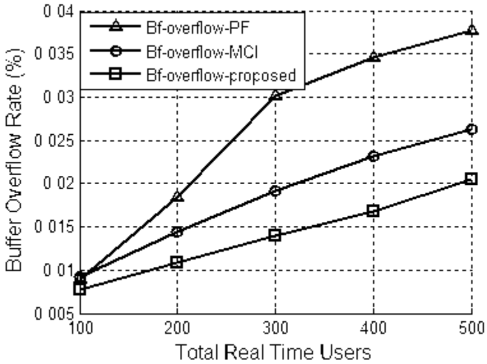

Fig. 4 Buffer overflow performances

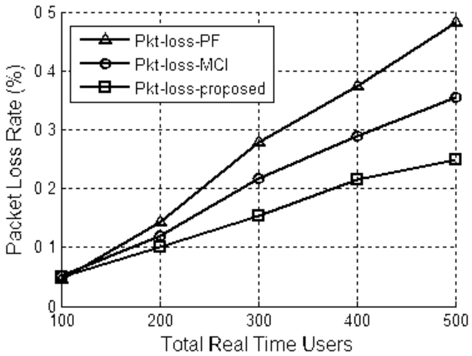

Fig. 3 Packet loss performances

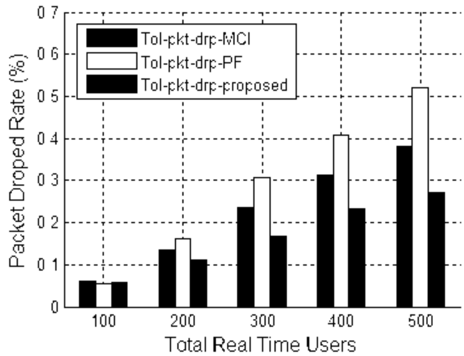

Fig. 5 Packet drop performance

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

Fig. 3 shows the relationship between the 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. It is shown that, the greater the number of users, the higher the performance.

Fig. 4 shows the buffer overflow performance. When the buffers of users exceed the maximum buffer capacity, the packets will be lost. In this simulation, the buffer capacity is set to 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 needs 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 the PF scheme.

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

V. Conclusions

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

References

[1] R. Razavi, S. Kucera, C. Androne, and H. Claussen, "Characterisation of uplink other-cell interference in co-channel WCDMA small cell networks," in Proceedings of IEEE VTC-Spring, pp.1-5, 2012.

[2] Z. Hao, H. Zhao, T. Mo, Y. Tang, H. Yu, "Uplink Interference Rejection Combining for WCDMA Using Multiple Antennas," Springer LNCS, Vol. 98, pp.91-96, 2012.

[3] D. Komnakos, D.N. Skoutak, D. Vouyioukas, A. Rouskas, "Delay Optimal Scheduling Policy for Real Time Services in WCDMA Systems," Springer LNCS, Vol. 64, No. 2 pp.435-445, 2011.

[4] N. Arulselvan, V. Ramachandran, S. Kalyanasundaram, "Distributed power control mechanism for HSDPA femtocells," in Proceedings of IEEE VTC, 2009.

[5] 3GPP Technical Report 25.848, "Physical layer aspects of UTRA High Speed Downlink Packet Access," version 4.0.0, 2001.

[6] 3GPP Technical Specification 25.211, "Physical Channels and Mapping of Transport Channels onto Physical Channels (FDD)," version 5.0.0, 2002.

[7] R.C. Elliot, W.A. Krzymien, "Scheduling Algorithms for the cdma2000 Packet Data Evolution," in Proceedings of IEEE VTC, Vol. 1, pp.304 - 310, 2002.

[8] P. Ameigeiras, "Packet Scheduling and Quality of Service in HSDPA," Ph.D, thesis, Department of Communication Technology, Aalborg University, 2003.

[9] A. Jalali, R. Padovani, R. Pankaj, "Data throughput of CDMA-HDR a high efficiency-high data rate personal communication wireless system," in Proceedings of IEEE VTC, Vol. 3, pp.1854-1858, 2000.

[10] L. Kleinrock, "Queueing System Volume I: Theory," Wiley-Interscience Publication, 1975.

Authors

Soojung Hur

2001: B.E. degree at Daegue Univ. Korea.
2004: M.S. finished course at SanDiego State Univ. USA.
2007~2012: M.S. and Ph.D. degree in Information and Communication Engineering from Yeungnam Univ. Korea.
2012~current: Yeungnam University Institute of Information & Communication as a researcher.
Research Interests: mobile telecommunication, loccation-based service.
Email: sjheo@ynu.ac.kr

Narzullaev Jakhongil

2009: B.S. in radio engineering from Tashkent University.

2012: M.S. degree in information and communication engineering from Yeungnam Univ. Korea.

2012~current: Alp Jahon Himoya as a Director.

Research Interests: wireless sensor network, Zigbee sensor network, mobile telecommunication, radio communication.

Email: luckerone@ynu.ac.kr

Yongwan Park

1982~1984: Received the B.E. and M.E. degree in Electrical Engineering from Kyungpook University.

1989~1992: M.S. and Ph.D. degree in Electrical Engineering from State Univ. of New York(Buffalo). USA.

1992~1993: California Institute of Technology as a research fellow.

1994~1996: Chief researcher for developing IMT-2000 system at SK Telecom.

1996~current: Professor of information and communication engineering in Yeungnam University.

Research Interests: 3G/4G systems, location-based services in wireless communication, etc.

Email: ywpark@yu.ac.kr