THROUGHPUT ENHANCEMENT USING ADAPTIVE DELAY BARRIER FUNCTION OVER HSDPA SYSTEM IN MIXED TRAFFIC SCENARIOS

Yong-Seok Kim
LTE Advanced Tech. Lab., Telecommunication R&D center, Telecommunication Network Business, Samsung Electronics, Korea

Keywords: Adaptive delay barrier function, a mixed traffic, VoIP, HSDPA.

Abstract: In this paper, we consider a method to enhance the throughput of HSDPA system in the mixed traffic scenario. Adaptive delay barrier (DB) function and adaptation method are proposed to maximize throughput of best-effort (BE) traffic with satisfying the delay latency of voice over internet protocol (VoIP) service. Moreover, in order to guarantee quality-of-service (QoS) for the mixed traffics, a design of modified scheduling scheme is provided in this paper. Simulation results show that the throughput of BE traffic service with a channel-adaptive DB function increases up to 30%, compared with the conventional scheme, without degrading the capacity of VoIP service over HSDPA system.

1 INTRODUCTION

High-speed downlink packet access (HSDPA), which supports the peak rate of 14.4 Mbps outperforms the third generation (3G) WCDMA system specified in 3GPP Release'99, has been designed to increase the downlink packet data throughput by using the advanced techniques such as hybrid automatic repeat request (HARQ), adaptive modulation and coding (AMC), fast scheduler in Node B (Holma and Toskala, 2004). Currently, the transmission of the real-time (RT) traffic using packet data internet protocols (IPs) is arguably the hottest attention in telecommunication technology. This is because it has high visibility in the consumer space. In other word, it implies cost-efficiency for operator by reason that the operating, maintaining and updating of circuit-switch (CS) related part would not be needed anymore. The challenge for new services that combine voice with data applications such as video and file sharing is also popular in today's research issues (Janevski, 2003). Scheduling at medium access control (MAC) layer is a core function to support quality-of-service (QoS) in these mixed traffic scenario. The basic scheduling algorithm, known as the proportional fairness (PF) scheduler, designed to support data traffics in 3GPP2 wireless system (Jalali et al., 2000). PF scheduler provides the effectiveness from the viewpoint of the multiuser diversity and fairness. However, it does not able to support RT traffic. In (Ramanan et al., 2001), modified-largest weighted delay first (M-LWDF) was proposed to support the stability and throughput while satisfying QoS requirements of RT traffic user. Although M-LWDF algorithm considers QoS traffic, it is applied for RT traffic and non-RT traffic separately. That is, it does not consider various QoS parameter for multiple traffic scenarios. On the other hand, urgency and efficiency based packet scheduling (UEPS) was introduced in (Ryu et al., 2005), which designed to support RT and non-RT traffic user simultaneously by taking the urgency of scheduling and the efficiency of radio resource usage into account for. UEPS serves non-RT traffics until RT traffics approach their deadlines, then RT traffics are scheduled with higher priority during their marginal scheduling time interval, which is defined as the time-utility function. So it can maximize the throughput of non-RT traffic with satisfying the QoS parameter of RT traffic. However, the requirement of adaptive adjustment for UEPS scheme according to the various traffic situations such as different traffic load levels and scheduling urgent factor was left to the further study issues. Hence, in this paper, we propose a modified QoS scheduler based on PF scheme to guarantee QoS in both voice and

Kim Y. (2007).
THROUGHPUT ENHANCEMENT USING ADAPTIVE DELAY BARRIER FUNCTION OVER HSDPA SYSTEM IN MIXED TRAFFIC SCENARIOS.
In *Proceedings of the Second International Conference on Wireless Information Networks and Systems*, pages 127-133
DOI: 10.5220/0002145001270133

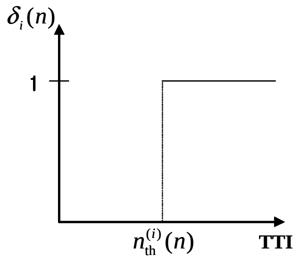


Figure 1: The concept of delay barrier function.

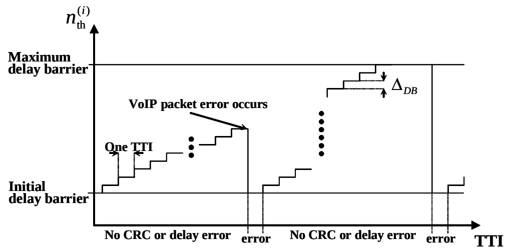


Figure 2: The outline of a proposed jump algorithm.

BE traffic service. Moreover, a channel-adaptive delay barrier (DB) function is proposed to increase the throughput of BE traffic service with satisfying the delay latency of VoIP service. The performance of UEPS will be compared with that of our proposed scheduling scheme. This paper is organized as follows. In Section 2, we present the scheduling method in mixed traffic environment. The simulation configuration and a criterion to measure the performance of VoIP service is described in section 3. Finally, simulation results and conclusion are presented in Section 4 and Section 5 respectively.

2 DOWNLINK SCHEDULING SCHEME

2.1 RF Packet Scheduler for BE Traffic Service

The use of effective scheduling algorithm is necessary for improving the throughput of system and keeping the fairness among users, since the HSDPA system shares resources with multi-users at the same transmission time interval (TTI). Generally, PF scheduling algorithm, described in (Jalali et al., 2000), is selected on HSDPA in order to achieve not only the maximum system throughput but also the fairness among users. Using the ratio of the current channel rate to the average allocated rate, the scheduler chooses a user who has the maximum output of scheduling metric as follow:

$$P_{\text{PF}}^{(i)}(n) = R_i(n)/r_i(n) \qquad (1)$$

where $R_i(n)$ is user i's estimated supportable bitrate in the n-th TTI and $r_i(n)$ is the filtered user's average throughput, which is updated by a low pass filtering parameter.

2.2 The Modified Scheduler for VoIP Traffic Service

Design objective of the PF algorithm is to maximize long-term throughput for only data service in CDMA-1x-EVDO system (Jalali et al., 2000). So, it can not support specific QoS parameter like maximum allowable delay for VoIP service. Therefore, for VoIP service, time-delay factor should be included in scheduling algorithm. Also, in order to measure delay, the scheduler puts time-stamps in each packet when it arrives at the priority queue that is operated with 8 queue buffers to support different QoS service. The priority would be more weighted as the delay increase. The modified scheduling algorithm operates by using the output metric as follow;

$$P_{\text{VoIP}}^{(i)}(n) = P_{\text{PF}}^{(i)}(n) \cdot f_i(\text{Que\_s}_i, \text{R\_y}_i) \qquad (2)$$

where $P_{\text{PF}}^{(i)}(n)$ is the value of scheduling output metric of user i calculated by using (1). The delay function $f_i(\cdot)$ can be designed by $f_i(\cdot) = r_i(n) \cdot \text{Que\_s}_i^{\beta}/\text{R\_y}_i^{\gamma}$, where Que_s_i is the accumulated buffer's volume of i-th user's VoIP packets that must be scheduled at n-th TTI, R_y_i is the remaining delay latency from the current n-th TTI to the delay bound at the MAC-hs scheduler. For BE service, $f_i(\cdot) = 1$. The parameters, β and γ are appropriate weight factor for each one.

2.3 A Proposed Scheduling Scheme in Mixed Traffic Scenarios

The QoS provisioning at the radio link means that the multimedia traffic should get predictable service form in the wireless system. However, when traffics demanding different QoS are mixed, it is difficult to achieve the task. The use of separate priority queues makes it possible to optimize HSDPA scheduling for each different QoS service. This paper optimizes the scheduling algorithm for traffics demanding different QoS by using priority handling. First of all, VoIP

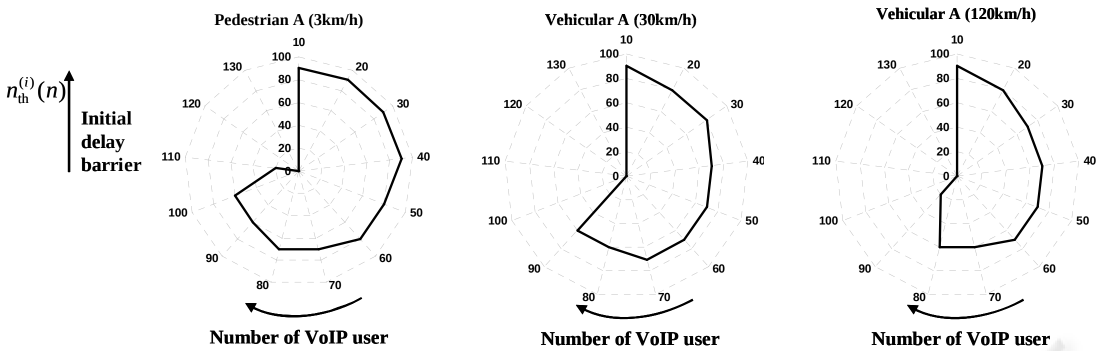


Figure 3: The value of initial delay barrier.

flows must be handled by higher priority than interactive traffic such as BE service. Generally, VoIP regarding conversational service has the highest priority of traffic classes since it is very sensitive to the transmission delay. Node B receives the value of priority from the radio network controller (RNC). Hence, we can differentiate between services through the priority values. Therefore, a modified proportional fair (PF) scheduling algorithm is adopted as follows:

$$P_{\text{Mix}}^{(i)}(n) = P_i^{\alpha} \cdot P_{\text{VoIP}}^{(i)}(n) \cdot \delta_i(n) \tag{3}$$

where P_i is the level of priority for i-th user which provides the high value to VoIP users and a DB function, $\delta_i(n)$, is represented as shown in Fig. 1, where $n_{\text{th}}^{(i)}(n)$ can be controlled according to the channel condition of i-th user's radio link. More detail description is in next subsection. For BE traffic, $\delta_i(\cdot) = 1$ and α is appropriate weight factor.

2.4 Throughput Enhancement by using a Channel-adaptive DB Function

The VoIP traffic has always the highest value of output scheduling metric since the parameter P_i, indicates the level of priority of i-th user's traffic class, may be decided the highest value among the types of service. Hence, as long as there is a VoIP class packet in the system, we have a low chance of selecting the BE traffic during the scheduling interval, and resulting in lower sector throughput for BE traffic user. Therefore, relaxing the level of traffic class priority should be need if VoIP traffic has some extra in delay latency, thereby giving more improved system throughput. In that point of view, the employment of adaptive DB function, $\delta_i(n)$, can increase the throughput of BE traffic service in mixed traffic scenario. That is, it gives the selecting opportunity to BE traffic packet in a proposed scheduler of (3) by applying the DB into VoIP traffic only if the VoIP packet has some marginal time of maximum allowable delay latency. For example, if the delay time of serving VoIP traffic is less than the threshold value of delay barrier, i.e. $n_{\text{th}}^{(i)}(n)$, BE traffic could be served by a proposed scheduler. On the contrary, if the delay time of serving VoIP traffic is larger than $n_{\text{th}}^{(i)}(n)$, VoIP traffic would be selected to go service by a proposed scheduler's selection rule. In this channel-dependent adaptive scheme, $n_{\text{th}}^{(i)}(n)$ is adjusted adaptively in accordance with the condition of i-th VoIP service user's radio link, which is similar to the processing of the channel-adaptive outer loop power control (OLPC) in (Holma and Toskala, 2004), (Koo et al., 2003). A proposed jump algorithm, which is illustrated in Fig. 2, is operated to maximize the throughput of BE traffic with maintaining the required QoS of VoIP user. That is, if the error detection device does not detect a cyclic-redundancy-check (CRC) error of VoIP packet or a VoIP packet dropped error in a TTI, the value of delay barrier is increased as follow:

$$n_{\text{th}}^{(i)}(n) = n_{\text{th}}^{(i)}(n-1) + \Delta_{DB} \tag{4}$$

where $\Delta_{DB} = \text{Step\_size} * \text{TargetBLER}$. Step_size and TargetBLER are any predetermined value. In this paper, TargetBLER is typically 2% for VoIP service and Step_size is set by 1. Moreover, the maximum delay barrier should be predefined, so it makes a value of delay barrier to do not increase above the maximum delay barrier. In like manner, if there is a CRC error or a VoIP packet dropped error due to a result of the exceeding required delay latency in scheuler, $n_{\text{th}}^{(i)}(n)$ is updated to the initial value of delay barrier, which can be predetermined according to the traffic load of VoIP users in a sector. It can be determined without considering of the load of BE traffic since the value of delay barrier has only to do with the load of VoIP traffic. Fig. 3 illustrates the initial value of delay barrier to make the percentage of all VoIP users satisfying the outage condition above 95%. In where, the outage is defined that VoIP user's packet error rate (PER) has been above 2% due to packet loss and packet delay exceeding the target budget. However, when this channel-adaptive DB function is applied, it makes the scheduler more flexible in choosing a BE traffic in mixed traffic scenarios.

3 VOIP SERVICE OVER HSDPA SYSTEM

3.1 Characteristic of VoIP Service

1) *Traffic model and protocol*: In this work, two traffic models are considered for the cases of BE service and VoIP service. In the context of BE traffic, we applied the full queue traffic model assuming that data can be always sent when a queue of certain user is chosen. On the other hand, conversation traffic can be approximated to the two state Markov traffic model with a suitable voice activity factor (VAF) (3GPP TR 25.896). The adaptive multirate (AMR) voice codec is mandatory for voice service in HSDPA systems. During bursts of conversation, with the AMR mode of 12.2kbps, the VoIP application generates 32-bytes voice payload at 20ms intervals (3GPP TS26.236). During silent periods, a 7-bytes payload carries a silence descriptor (SID) frame at 160ms intervals. A typical VoIP protocol stack, which employs the real-time transport protocol (RTP), is encapsulated to the user datagram protocol (UDP). This, in turn, is carried by IP. The combined these protocols demand a 40-bytes IPv4 header or a 60-bytes IPv6 header. Obviously, the overhead caused from the header to support VoIP service seriously degrades the spectral efficiency. Therefore, efficient and robust header compression (ROHC) technique can be used to reduce the effect of relatively large headers in the IP/UDP/RTP layers. This technique can reduce the size of the IP/UDP/RTP headers as little as 2 or 4 bytes. Maximum compression 1 byte can be achieved by imposing limitations (IETF RFC 3059, 2001).

2) *Definition of VoIP capacity*: The VoIP capacity is in the sense that there exists the maximum number of VoIP users that can be supported per sector without exceeding a given outage threshold. In packet-switch (PS) network, packets will be dropped under network traffic loads congestion due to packet loss and packet delay exceeding the target budget. Although some packet loss occurs, the voice quality is not affected if the amount of packet loss is less than outage threshold. To proceed with this work, we assume that the PER is kept within 2% and at least 95% of VoIP users should meet the above criterion (3GPP2 TSG-C.R1002-0).

Table 1: Summary of end-to-end delay component for VoIP.

Delay component	Delay assumption
Voice encoder	20ms(12.2Kbps)
NodeB scheduling+HARQ	Max. 110ms
NodeB site ROHC, RLC+MAC process Downlink propagation	30ms
UE scheduling+HARQ	40ms
UE site processing, buffering, etc Uplink propagation	30ms
Backhaul delay	30ms
IP network delay	About 42ms

3) *End-to-end delay latency for QoS support*: To ensure end-to-end QoS, the low delay is one of the most important criteria for maintaining high-quality VoIP service. But, to attain high VoIP capacity, the scheduler must have sufficient time to manage voice packets. Of the assumed 285ms end-to-end delay budget for qualified voice service, about 110ms is available for scheduling in the downlink (ITU-T G.114). The delay in IP and backhaul network is in general bounded to 72ms (3GPP TR 25.853), which is fixed value allowing us to focus on the delay budget within radio access network as shown in Fig. 4. The end-to-end delay budget in the case of mobile-to-mobile conversation can be assumed as table 1. Although VoIP performance depends on both downlink and uplink performance, we would like to set aside the consideration of both directions as comprehensive study for future research.

3.2 System-level Simulation Setup

To investigate the performance evaluation with a mixed voice and BE traffic, a system-level Monte-Carlo computer simulation is accomplished in this paper. The simulations are carried out with a regular hexagonal 19 cellular model, where the distance between Node B is 1km. Mobile terminals should be uniformly distributed on the 19-cell layout for each simulation run and assigned different radio link models according to the assignment probability specified in (3GPP TS25.101). Note that a realistic model of the wave propagation plays an important role for the significance of the simulation results. Shadowing is modeled by a log-normal fading of the total received power and a basic attenuation is determined by the Hata model (3GPP2 TSG-C.R1002-0). Moreover, we reserve the resources for control and common chan-

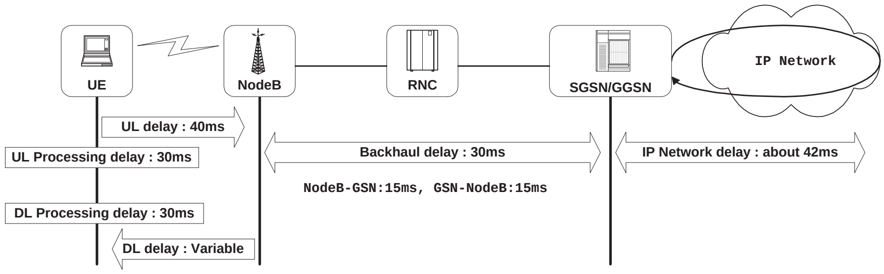


Figure 4: End-to-end network delay components for VoIP service.

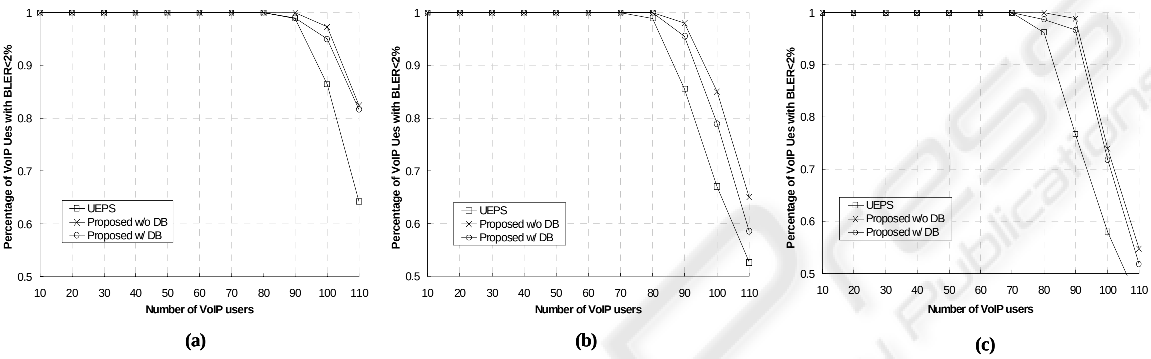


Figure 5: Outage probability versus number of VoIP user. (a) Pedestrian A 3km/h; (b) Vehicular A 30km/h; (c) Vehicular A 120km/h.

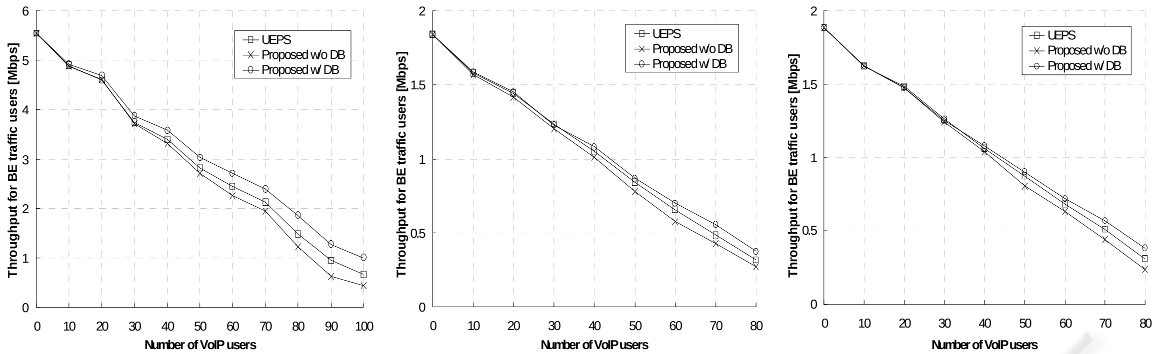
nel such as OVSF codes and HSDPA power to obtain the provided simulation results. As mentioned above, we applied the RTP/UDP/IP packet header compression using IETF RFC 3059 where the total size of all compressed header is with 3 bytes (1byte ROHC base header + 2bytes UDP checksum). Further, the fast link adaptation and hybrid ARQ (HARQ) modules are employed in this work. Finally, it has to be mentioned that we simulate 100,000 TTI snapshots in average for investigating the performance of the system. The main simulation parameters are summarized in Tables 2.

4 SIMULATION RESULTS

In this section we evaluate the throughput of BE traffic and the capacity of VoIP with employing a channel-adaptive delay barrier function in the various fading channel environments (3GPP TS25.101). Also, a proposed scheme is also compared with the conventional UEPS method. In all simulation, we assume the number of BE traffic users is 40. The percentage of users satisfying outage limitation is presented in Fig. 5 as a function of the number of VoIP users for different scheduling schemes when the delay latency available to scheduler is fixed as 110ms. From the figure, we observe that a percentage of VoIP

Table 2: System simulation parameters.

Parameter	Assumption
Source traffic	AMR 12.2kbps, VAF=0.32, 2-state Markov, ROHC 3 bytes [IETF RFC 3059]
Cellular layout	Hexagonal grid, 19 sites, 3 sectors (NB-to-NB 1km) Carrier frequency 1.9GHz
Propagation loss	Path loss=-128.1-37.6*log(R)
Shadowing model	Log Normal Std. dev. 8dB, [Hata model]
UE speed	3km/h,30km/h,120km/h
Antenna gain	Node B 14dB/UE 0dB, Other loss -10dB
Fading Model	Pad.A, Veh.A, Evaluated 3GPP(TS25.101)
UE Rx diversity	2 Rx antennas(UE catagory 10)
Retransmission	No RLC retransmission, Async. HARQ (max. retrial = 6)
CQI delay / error	5TTI (10ms) / 1%
Scheduling	Proposed scheme
Reserved	1.Common channel power:20% 2.Associated DCH power per UE: 0.3% of total 3.HS-SCCH power: 9dB offset over associated DPCH 4.common channel code:10 5.DPCH code per UE:1 6.HS-SCCH codes are considered



(a) (b) (c)

Figure 6: Throughput of BE traffic versus the number of VoIP user for different scheme. (a) Pedestrian A 3km/h; (b) Vehicular A 30km/h; (c) Vehicular A 120km/h.

Table 3: Summary of the system-level simulation results in mixed traffic scenario.

		Ped.A (3)	Veh.A (30)	Veh.A (120)
Capacity for VoIP traffic	Proposed w/o DB	100	90	90
	UEPS	90	80	80
	Proposed w/ DB	100	90	90
Throughput for BE traffic (Mbps)	Proposed w/o DB	0.63	0.27	0.24
	UEPS	0.95	0.32	0.31
	Proposed w/ DB	1.27	0.37	0.38

users with satisfying PER over outage threshold, i.e. 2%, decrease according to the increase of the number of VoIP users. This is because the probability of packet loss due to the exceeding target delay latency becomes more increase with a large number of VoIP users. However, according to the result, the more VoIP capacity can be achieved by employing a proposed scheduling scheme if compared with UEPS method. For example, when aiming for an identical percentage of 0.95 in pedestrian radio link environment, while the UEPS scheme can support 80 VoIP service users, a proposed scheduling scheme can support 90 VoIP users. In Fig. 6, we evaluate the achievable BE traffic throughput as a function of the number of VoIP users when the fixed BE traffic users of 40. As you can see the figure, the increase of the number of VoIP users exhibits the decrease of BE traffic's throughput remarkably. This is because the VoIP traffic has the most scheduling opportunity against BE traffic when the load of VoIP traffic increases. However, note that the throughput with a proposed scheduling scheme can be more achieved if it is compared with a proposed scheduler without $\delta_i(n)$ or UEPS. For instance, at the aim of 90 VoIP service user, the employing channel-adaptive DB enhances the BE traffic's throughput as 100% over against a proposed scheme without $\delta_i(n)$ in (3) and 30% against UEPS in pedestrian channel condition. We can also see from the figure, a proposed scheduling scheme with adaptive DB function still offers better throughput in vehicular radio link. Table 3 summarizes the capacity of VoIP and throughput performance in various propagation conditions. The results confirm that a proposed scheduler employing a channel-adaptive DB provides significant improved throughput without degrading the capacity of VoIP service if compared to the conventional UEPS.

5 CONCLUSION

In this paper, we propose a modified scheduler employing a channel-adaptive DB function to enhance the BE traffic's throughput in mixed traffic scenarios without degrading the QoS requirement of VoIP traffic over HSDPA. Our simulation results show that the capacity of VoIP traffic does not reduced by employing a proposed scheme as opposed to UEPS. In addition, the throughput of BE traffic service can be improved significantly compared to the conventional scheme. The consideration of the combination of other traffic types such as web and streaming in the system may be an interesting issue for future study.

REFERENCES

H. Holma and A. Toskala, 2004. *WCDMA for UMTS*, John Wiley and Sons, Third Edition.

T. Janevski, 2003. *Traffic Analysis and Design of Wireless IP Networks*, Norwood, MA, Artech House.

A. Jalali, R. Padovani and R. Pankaj, 2000. Data throughput of CDMA-HDR a high efficiency-high data rate personal communication wireless system, *IEEE Proc. VTC2000*, 1854-1858.

K. Ramanan, et al., 2001. Providing quality of service over a sharded wireless link, *IEEE Commun. Magazine*, 39, 150-154.

S. Ryu, B. Ryu, H. Seo and M. Shin, 2005. Urgency and efficiency based packet scheduling algorithm for OFDMA wireless system, *IEEE Proc. ICC2005*, 2779-2785.

C. -S Koo, et al., 2003. Outer Loop Power Control Using Channel-Adaptive Processing for 3G WCDMA, *IEEE Proc. VTC2003-Spring*, 490-494.

3GPP TR 25.896, *Feasibility Study for Enhanced Uplink for UTRA FDD*.

3GPP TS26.236, *Packet switched conversational multimedia application; Transport protocols*.

IETF RFC 3059, 2001. *Attribute List Extension for the Service Location Protocol*.

3GPP2 TSG-C.R1002-0, 2004. *CDMA2000 Evaluation Methodology*, version 1.0.

ITU-T G.114, *Transmission systems and media - General characteristics of international telephone connections and international telephone circuits*.

3GPP TR 25.853, *Delay Budget within the Access Stratum*.

3GPP TS25.101, *User Equipment radio transmission and reception (FDD)*.