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Downlink Packet Scheduling with Minimum-Bandwidth Provisions in LTE/LTE-A Networks

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With the growing number of human and machine devices in IoT (Internet of Things), ubiquitous network connection and traffic classification become more essential. The LTE/LTE-A system, proposed by 3GPP, is a solution. An evolved NodeB (eNB) in LTE/LTE-A system encounters the challenge of allocating the radio resource to meet different QoS (Quality of Service) requirements for multiple traffic classes. QCI (QoS Class Identifier) is defined to specify the QoS characteristics of user data in LTE/LTE-A networks. This paper proposes a packet scheduling algorithm with minimum-bandwidth provisions (PSMP) over LTE/LTE-A downlink. The PSMP scheme can calculate the minimum-bandwidth requirements for GBR (Guaranteed Bit Rate) and NGBR (non-GBR) traffic according to their associated packet delay constraint and packet loss ratio, respectively. Due to beginning with the minimum-bandwidth allocation by referring to QCIs, the proposed PSMP scheme not only can satisfy the delay constraint for real-time traffic, but also can alleviate non-real-time traffic starvation problem. The simulation results show our proposed PSMP scheme outperforms the previous works in the packet delay of GBR traffic and the throughput of NGBR traffic. As to VoIP service, the PSMP scheme can reduce L1/L2 control overhead to improve the network performance by alternatively adopting the semi-persistent scheduling (SPS) scheme.

Keywords: packet scheduling, minimum-bandwidth requirement, QCI, LTE, LTE-A

1. INTRODUCTION

Mobile wireless network development has changed with each passing day in the recent years. A large amount of smart handheld devices come out. An increasing number of human and machine devices in IoT (Internet of Things) not only bring people the convenience, but also enrich human life [1, 2]. However, diverse device types bring about different service requirements. Although the LTE/LTE-A system, proposed by 3GPP [3], depicts the packet scheduling framework with different QCI (QoS Class Identifier) bandwidth demands, how to allocate the scarce resource blocks to meet different QoS (Quality of Service) requirements remains an open research issue.

LTE/LTE-A (Long Term Evolution/Long Term Evolution-Advanced) improves bandwidth efficiency and offers rapidly steady network environment for 3GPP mobile communication networks. In order to speed up signaling procedure, LTE/LTE-A develops towards the structure of doing away with a hierarchical system. LTE/LTE-A simpli-

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fies RNC (Radio Network Controller) management and makes eNB (evolved NodeB) nodes directly connected to each other. To improve the spectral efficiency, LTE/LTE-A uses OFDMA (Orthogonal Frequency Division Multiple Access) technique in the downlink, while using SC-FDMA (Single-Carrier Frequency-Division Multiple Access) technique in the uplink. The OFDMA technique [11, 12] divides the radio resource in both time domain and frequency domain, in which each basic unit of bandwidth is called an RB (Resource Block). The MAC (Medium Access Control) packet scheduler at each eNB dynamically assigns the available RBs at every TTI (Transmission Time Interval) according to a certain packet scheduling algorithm [13], which is decided by the metrics on UEs (User Equipments). As depicted in Fig. 1, an LTE/LTE-A network mainly consists of two major components [4], EPC (Evolved Packet Core) Network and RAN (Radio Access Network). The RAN contains several E-UTRANs (Evolved Universal Terrestrial Radio Access Networks) with an eNB deployed in each E-UTRAN. Note that eNB nodes can directly communicate with each other in the signaling procedure. Therefore, wireless resource management is implemented at eNBs.

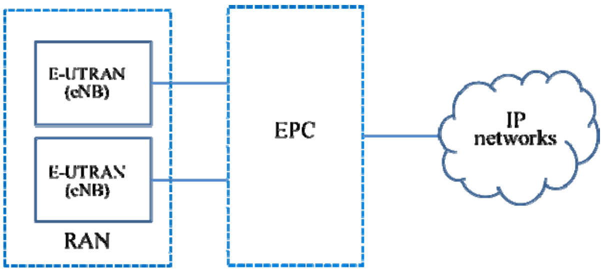


Fig. 1. The networking architecture in LTE/LTEA.

Traffic flows can be divided into two classes, RT (Real-time) and NRT (Non-real-time), according to their QoS characteristics. Corresponding to the QCIs of LTE/LTE-A [9, 10], RT/NRT traffic flows will carry on the screening through traffic flow template (TFT) of the PGW (Packet Data Network Gate Way) within the EPC. RT corresponds to GBR (Guaranteed Bit Rate) and NRT sorts out to NGBR in LTE/LTE-A systems. Similar to the previous-generation mobile wireless systems, two types of duplexing techniques, FDD (Frequency-Division Duplexing) and TDD (Time-Division Duplexing) are used in LTE/LTE-A. FDD is a technique with which the transmitter and receiver operate at different carrier frequencies. On the other hand, TDD separates uplink and downlink signals by matching full duplex communication over a half-duplex communication link.

This paper focuses on designing a packet scheduling algorithm to support different QoS (Quality of Service) demands at eNBs. The innovative aspect of our proposed scheme is that beginning with the minimum-bandwidth allocation by referring to QCIs not only can satisfy the delay constraint for real-time traffic, but also can alleviate non-real-time traffic starvation problem. For performance evaluation, we have modified LTE-SIM Simulator [16] by adding our proposed scheme into the scheduling component of MAC layer at an eNB. The simulation results have demonstrated that our proposed scheme not only can satisfy the delay constraint of real-time traffic, but also can alleviate non-real-time traffic starvation problem. Compared to the previously-proposed scheduling schemes, our proposed scheme outperforms them in the overall throughput of video flows in any traffic load condition due to providing only the minimum-bandwidth requirement for each UE at the beginning.

The remainder of this paper is organized as follows. Section 2 discusses the related works of our proposed packet scheduling algorithm with minimum-bandwidth provisions (PSMP) in the LTA/LTE-A system. The proposed PSMP scheme is elaborated in Section 3. The simulation results and discussions are presented in Section 4. Finally, concluding remarks are given in Section 5.

2. RELATED WORKS

Well-known packet scheduling algorithms, such as MT (Maximum Throughput), PF (Proportional Fairness) [5], and RR (Round Robin), do not consider users' service requirements. Taking users' service requirement and channel quality into account, M-LWDF (Modified Largest Weighted Delay First) [6, 7] and FLS (Frame Level Scheduler) [8], make service differentiation between real-time and non-real-time traffic in LTE/ LTE-A system.

As defined below, the M-LWDF (Modified Largest Weighted Delay First) scheme [6, 7] provides packet scheduling with QoS based on the metric for user *i* and RB *k*. According to Eq. (1), the *k*th RB is assigned to user *i* if user *i* has the largest metric on the *k*th RB. However, the M-LWDF has no QoS guarantee for RT traffic over NRT traffic since it takes different metrics between RT and NRT traffic.

$$m_{i,k}^{M-LWDF} = \begin{cases} \alpha_i \cdot D_{HoL,i} \cdot \dfrac{d_k^i(t)}{\overline{R^i}(t-1)}, & \text{for } RT \\ \dfrac{d_k^i(t)}{\overline{R^i}(t-1)}, & \text{for } NRT \end{cases} \tag{1}$$

where $\alpha_i = -\log\delta_i/\tau_i$ is a weight parameter, in which δ_i is the tolerant packet dropping probability and τ_i is the delay constraint of user *i*. In addition, D_{HoLi} is the head-of-line packet delay, $d_k^i(t)$ is the achievable data rate on RB *k*, and $\overline{R^i}(t-1)$ represents the past average throughput for user *i*.

A two-level downlink scheduling for real-time multimedia services, named Frame Level Scheduler (FLS), was presented by Piro and Boggia [8]. At the upper level, the FLS can calculate the expected throughput for real-time services in a frame by utilizing discrete-time linear control theory. Then, at the lower level, the RBs at each TTI are assigned to RT services as enough as possible based on the proportional fairness rule to satisfy the expected throughput in a frame. NRT services can obtain RB allocation only when there are RBs remaining un-masked after allocating RBs for RT ones. Even though the two-level scheduling design can improve the overall network throughput, NRT services are probably confronted by starvation.

In a word, the previously-proposed schemes with QoS consideration provide service to real-time traffic in a top priority, which results in the defect of NRT service starvation. To avoid NRT traffic from starvation, this paper proposes a PSMP scheme over LTE/ LTE-A downlink. The proposed PSMP scheme provides RT and NRT traffic with their individual minimum bandwidth requirements in the first stage according to the packet delay constraint and packet loss ratio, respectively. Due to beginning with the minimum-bandwidth allocation by referring to QCIs, the PSMP scheme not only can satisfy the delay constraint for real-time traffic, but also can alleviate non-real-time traffic starvation problem.

On the other hand, several solutions have been proposed to minimize signaling overhead for highly dense VoIP (Voice over IP) traffic [14, 15]. Aiming to increase the VoIP capacity in the network, the semi-persistent scheduling (SPS) schemes pre-allocate a certain subset of RBs to VoIP users. The subset of RBs are dedicated for VoIP flows in continuous TTIs and the VoIP users have only to listen to the specific subset of RBs. Although the SPS schemes are not conceived for improving spectral efficiency or reducing packet loss rate, they can probably improve the network performance as the number of VoIP flows drastically increases owing to much less control overhead. Therefore, our proposed PSMP scheme adopts the semi-persistent scheduling specifically for VoIP flows to improve the network performance by reducing L1/L2 control overhead.

3. THE PROPOSED PSMP SCHEME

3.1 Model Description

There are two resource types of traffic, GBR and NGBR, in LTE/LTE-A system. As illustrated in Fig. 2, the proposed PSMP scheme classifies traffic flows into GBR_VoIP, GBR_Video, and NGBR_BE. In order to satisfy the QoS demands for both GBR and NGBR traffic, the PSMP scheme firstly calculates the minimum-bandwidth requirements of all UEs by referring to their QCIs every TTI in a frame, and allocates RBs to GBR flows under the principle of providing just enough but not too much bandwidth. However, the masked RBs for VoIP flows are persistently reserved for them to reduce L1/L2 control overhead until the VoIP flows enters their individual silence period. Secondly, the residual RBs are allocated to NGBR flows on the same principle of minimum-bandwidth allocation. RB allocation is performed according to the modified proportional fairness metric for each UE in a descending order.

For the detail elaboration on the PSMP, all parameters and denotations used in the PSMP scheme are listed in Table 1. Let P denote the period of a TTI. B_m and Q_m respectively stand for the buffer size and queuing length of QCI index equal to m. Let r_i and τ_i represent the data generation rate and packet delay budget of user i, respectively. $T_i(t)$ is the average throughput of user i up to time t and $b_j^i(t)$ is the expected bit rate of user i on the jth RB at time t. $D_{HOL,i}$ denotes the head-of-line packet delay of user i. R_i is the minimum-bandwidth requirement of user i and is expressed in terms of Q_m, r_i, τ_i, and P for GBR traffic, as shown in Eq. (2). On the other hand, R_i for NGBR traffic is obtained by Eq. (3) by rewriting Eq. (4), which means the packet loss rate of NGBR flows must be equal to or lower than the tolerable packet loss rate, denoted as δ_i.

$$R_i = (Q_m + r_i \times P)/\tau_i, \text{ for } GBR \tag{2}$$

$$R_i = [Q_m + (r_i - \delta_i)P - B_m]/P, \text{ for } NGBR \tag{3}$$

$$(Q_m + r_i \times P - R_i \times P - B_m)/P \le \delta_i \tag{4}$$

$$m_j^i(t) = \begin{cases} D_{HOL,i} \times [b_j^i(t) / T_i(t)], & \text{for } GBR \\ b_j^i(t) / T_i(t), & \text{for } NGBR \end{cases} \tag{5}$$

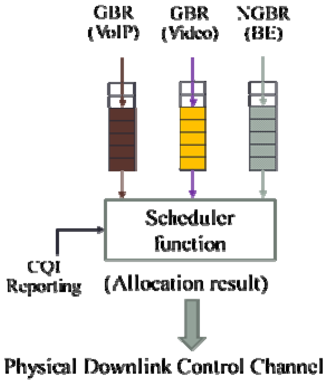


Fig. 2. The system model of the PSMP.

Traffic Flows are assigned RBs according to their individual metrics, denoted as $m_j^i(t)$, with GBR prior to NBGR. $m_j^i(t)$ represents the metric of user i on the jth RB at time t. As defined by Eq. (5), $m_j^i(t)$ basically takes into accounts the expected bit rate of user i on the jth RB, $b_j^i(t)$, and the average throughput that user i has experienced up to now, $T_i(t)$, in order to maximize bandwidth utilization and provide fairness among the users of the same resource type. Since GBR belongs to real-time traffic, which is time-limited, the metric in Eq. (5) for GBR includes one more factor, that's, the head-of-line packet delay of user i, $D_{HOL,i}$, to indicate the stringent degree in time for each GBR flow. An available RB j is firstly assigned to user i if it has the largest metric, $m_j^i(t)$, among the same resource type.

Table 1. Parameters used in the PSMP.

Parameters	Descriptions
P	The period of a TTI
Q_m	Queuing length of QCI index equal to m at eNB
r_i	Data generation rate of user i
$T_i(t)$	The average throughput of user i up to time t
τ_i	Packet delay budget of user i
δ_i	Tolerable packet loss rate of user i
B_m	Buffer size of QCI index equal to m
R_i	The minimum-bandwidth requirement of user i
$b_j^i(t)$	The expected bit rate of user i on the jth RB at time t
$D_{HOL,i}$	Head-of-line packet delay of user i
$m_j^i(t)$	The metric of user i on the jth RB at time t

3.2 The Pseudo Code

The pseudo code of the proposed PSMP scheme is shown in Table 2. At the beginning of each TTI, the minimum-bandwidth requirements for all UEs are obtained from

Table 2. The pseudo code of the PSMP.

```
If Ng is empty & Nn is empty    // Ng (Nn) is the set of GBR (NGBR) flows.
    Stop scheduling this TTI;
if |Ng| > 0
    Compute R_i for each UE_i in Ng;    // Eq. (2)
N = Ng;
if |Nn| > 0
    Compute R_i for each UE_i in Nn;    // Eq. (3)
N = N ∪ Nn;
if |Np| > 0    // Np is the set of VoIP flows; Np is a subset of Ng.
    Allocate RBs for VoIP flows in Np with semi-persistent scheduling;
    Update the available RBs this TTI;
    if VoIP's queue is empty for UE_i
        Np = Np – {UE_i};
if |Ng| > 0
    for each UE_i in Ng
        if (m_j^i(t) for GBR is largest && b_j^i(t) >= R_i)    //Eq. (5)
            Assign RB j to UE_i;
            Ng = Ng – {UE_i};
    Mark RBs that are assigned this TTI;
    if UE_i is VoIP
        Np = Np ∪ UE_i;
if there are available RBs & |Nn| > 0
    for each UE_i in Nn
        if (m_j^i(t) for NGBR is largest && b_j^i(t) >= R_i)    //Eq. (5)
            Assign RB j to UE_i;
            Nn = Nn – {UE_i};
    Mark RBs that are assigned this TTI;
if there are available RBs
    Do RB allocation in the descending order of b_j^i(t);
    //firstly for GBR, next for NGBR
Transmit RB allocation result over PDCCH;
```

Eqs. (2) and (3) by referring to their individual QCI parameters. When there are VoIP flows in GBR traffic, those masked RBs in the previous frames are still reserved for them until they enter their individual silence periods. Then, the residual RBs are allocated to GBR flows in the descending order of GBR metric, as defined by Eq. (5), to provide their individual minimum-bandwidth requirements. Following from RB allocation for GBR flows, the residual RBs are allocated to NGBR flows in the descending order of NGBR metric, as defined by Eq. (5), to provide their individual minimum-bandwidth requirements. If there are still available RBs after finishing the preceding stages, they can be assigned to GBR/NGBR flows in the descending order of the expected bit rate, denoted as $b_j^i(t)$, to increase the overall system throughput.

4. SIMULATION RESULTS

We evaluate the performance of our proposed PSMP scheme via simulation on LTE-SIM [16, 17]. Fig. 3 shows the simulation network topology. The parameters and their values used in the simulation are summarized in Table 3. The system bandwidth for an eNB in a cell is 5MHz, that's equal to 25 RBs for each TTI in LTE/LTE-A system. The coverage of an eNB is within a radius of 0.5 kilometer. *N* denotes the number of UEs and varies from 10 to 50. All of the UEs move in the random way-point mobility model at an average speed of 30 Km/hr. For the link between the eNB node and each UE, three types of flows (with traffic model), VoIP (8Kbps), video (H264-128Kbps), and BE (infinite buffer), are generated. The delay budget for GBR traffic (VoIP and video) is 100 milliseconds. The packet error loss rate for BE is 10^{-6}. Compared to the previous works, PF, M-LWDF and FLS, the performance measures, including the CDF (Cumulative Distribution Function) of one-way packet delay, the packet loss rate (PLR), and the throughput of GBR/NGBR, are plotted in Figs. 4-7. Unless explicitly specified, all simulation results are obtained by averaging 10 random samples. Each sample takes into accounts only the last 70 seconds in a running duration of 100 seconds. This section verifies the accuracy and reliability of the proposed scheme through simulation and comparison of the performance with several well-known schemes.

Table 3. The parameters and values used in the simulation.

Parameters	Values
System bandwidth	5 MHz (*i.e.*, 25 RBs)
Radio mode	FDD
Cell layout	Radius: 0.5 km
Number of UEs	10, 20, 30, 40
UE speed	30 km/hr
UE mobility model	Random way-point
Traffic model	Video: H264-128kbps, VoIP: 8kbps, BE: infinite buffer
Delay budget for GBR	Video: 100ms, VoIP: 100ms
Packet error loss rate for BE	10^{-6}

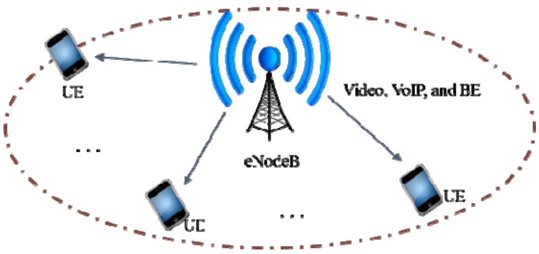
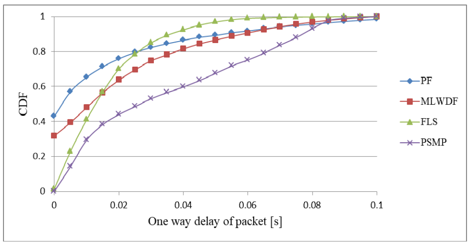
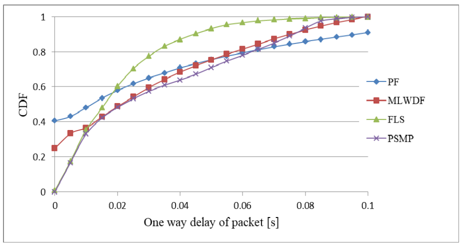


Fig. 3. The network topology used in the simulation.

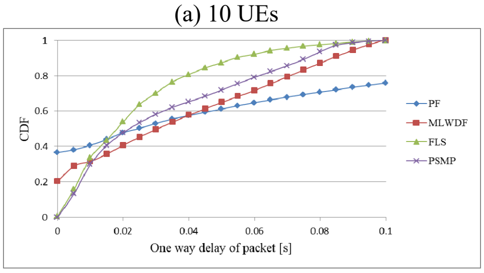
Figs. 4 (a)-(d) show the cumulative distribution function (CDF) of one-way packet delay of video for the number of UEs being 10, 20, 30 and 40, respectively. One can see that all of the schemes used in the simulation, PF, M-LWDF, FLS, and our proposed PSMP, have all video packets arrive in the packet delay budget (100 ms) because all of them are QoS-aware strategies. When *N* is 10 or 20, the average packet delay for video with the PSMP is larger than the previously-proposed works, PF, M-LWDF, and FLS. This is because the PSMP is intended to provide the minimum-bandwidth requirement of each UE at the beginning to avoid BE traffic starvation. Nevertheless, when the network load increases, *i.e.*, N is 30 or 40, our proposed PSMP outperforms the previously-proposed schemes except for FLS. Although FLS always has the smallest one-way packet delay for video among the schemes performed in the simulation, Fig. 5 shows that the packet loss rate for video with FLS becomes larger than that with our proposed PSMP when N is increased to 30. This is because FLS satisfies RT service in a top priority and provides a UE as much bandwidth as it can.



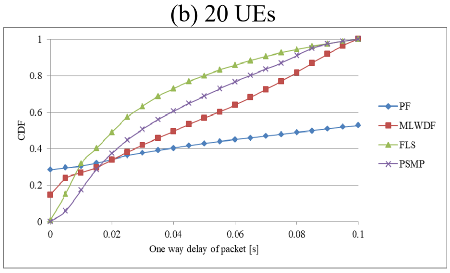
(a) 10 UEs



(b) 20 UEs



(c) 30 UEs



(d) 40 UEs

Fig. 4. One-way delay packet of video flows.

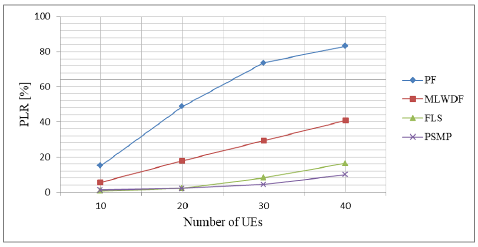


Fig. 5. Packet loss rate of video flows.

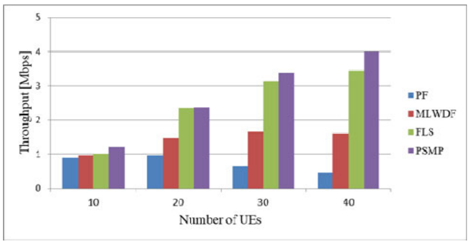


Fig. 6. Throughput of video flows.

In addition, it is noteworthy from Fig. 6 that the overall throughput for video with PSMP is always larger than the three previously-proposed schemes owing to the smaller packet loss rate. As observed in Fig. 7, PF and M-LWDF have larger overall throughput for BE traffic than FLS and PSMP because they do not guarantee that RT flows are always served in a higher priority than NRT flows. However, the proposed PSMP scheme outperforms FLS in the overall throughput for BE traffic. In summary, compared to the previously-proposed schemes, the proposed PSMP scheme always has the best throughput for video flows with no BE starvation under satisfying the delay budget of video.

Furthermore, it is shown in Fig. 8 that the proposed PSMP also has a lager overall system throughput than FLS with QoS guarantees for RT traffic.

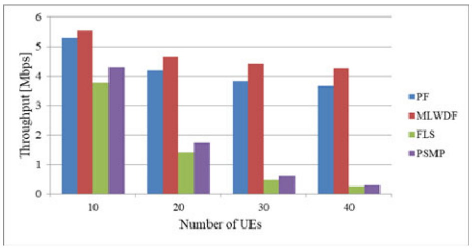


Fig. 7. Throughput of best effort flows.

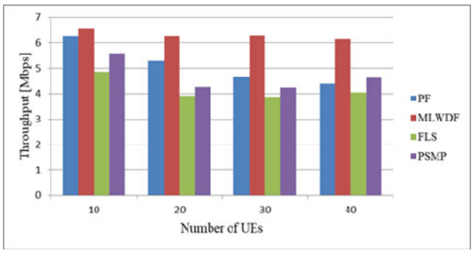


Fig. 8. System throughput of a cell.

Table 4. Packet loss rate for video with/without SPS for VoIP.

UEs	w/o SPS (1:1:1)	with SPS (1:1:1)	w/o SPS (1:1:3)	with SPS (1:1:3)
10	1.223	1.087	2.058	1.621
20	2.207	1.806	4.711	2.566
30	4.441	3.395	10.077	5.828

To further improve the cell throughput, the proposed PSMP scheme adopts the semi-persistent scheduling for VoIP flows to reduce L1/L2 control overhead, that's, the masked RBs in the previous frames are still reserved for them until they enter their individual silence periods. Table 4 shows the packet loss rate for video in two different proportions of traffic (BE: video: VoIP), 1:1:1, and 1:1:3. First of all, it is very straightforward that the packet loss rate for video increases when either the proportion of VoIP traffic or the number of UEs increases. Secondly, it is demonstrated that the packet loss rate for video gets less influenced as VoIP traffic grows more by adopting SPS for VoIP flows.

5. CONCLUSIONS

Aiming to alleviate NGBR service starvation when providing QoS guarantee for GBR service in LTE/LTE-A system, we have presented an algorithm of PSMP. With the PSMP scheme, radio resource block allocation for NGBR traffic is performed after GBR one since it is QoS-aware. From the simulation results, it is demonstrated that the proposed PSMP scheme not only can satisfy the delay constraint of real-time traffic, but also can alleviate non-real-time traffic starvation problem. Compared to the previously-proposed scheduling schemes, including PF, M-LWDF, and FLS, PSMP outperforms them in the overall throughput of video flows in any traffic load condition due to providing only the minimum-bandwidth requirement for each UE at the beginning. Furthermore, PSMP also has better throughput for BE traffic than FLS when the network load becomes getting heavy. Furthermore, the proposed PSMP scheme has alternatively adopted the SPS scheme for VoIP flows. The GBR traffic includes video and VoIP traffic in LTE/LTE-A system. By adopting the SPS for VoIP, the simulation results have demonstrated that the packet loss rate for video gets less influenced as VoIP traffic grows more. This is because the SPS can reduce L1/L2 control overhead and hence save the scarce system bandwidth through making use of the periodical behavior of VoIP.

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