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Research Article

A Hybrid Network User Satisfaction-Based Downlink Scheduling in LTE-A Network

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To provide high-quality wireless network multimedia services, the 4G/5G network media access control (MAC) layer has adopted the concept of the quality of service (QoS) with different traffic classification service requirements. To efficiently allocate the limited radio resources for achieving a good balance in QoS, throughput and fairness are necessary for the wireless network MAC scheduling issues. While many conventional scheduling schemes only considered one factor: rate or delay, network users might actually have different rate and delay requirements in concert. Herein, a hybrid network user satisfaction-based downlink scheduling approach, namely, max rate delay urgency first (MRDUF), is developed. The MRDUF approach simultaneously considers rate and delay requirements and adopts a hybrid strategy comprising time- and frequency-domain schedulers using two schemes, namely, the first come first serve (FCFS) and max throughput (MT), for the LTE-A downlink environment. Simulation study results have shown that our proposed approach outperforms the conventional scheduling schemes, including MT, proportional fair (PF), blind equal throughput (BET), and earliest deadline first (EDF). The benefit of MRDUF in terms of fairness of satisfaction of rate and delay is demonstrated by the simulation study.

1. Introduction

The *long-term evolution (LTE)* [1] technology is designed to work with different bandwidth requirements and to provide a peak data rate of 100 Mbps in the downlink and 50 Mbps in the uplink. It is also the *fourth generation (4G)* of wireless broadband communications standardized in recent years. In wireless broadband network, LTE and *long-term evolution-advanced (LTE-A)* [2–7] is commercially deployed in many countries. The LTE-A offers traditional voice telephone services and provides a cost-effective broadband communication service. The *Third Generation Partnership Project (3GPP)* formally recognizes the LTE-A platform as the technology standard for wireless communications. Since the LTE-A standard is defined by telecom vendors and is backward compatible with the GSM or UMTS cellular systems, its deployment is much easier than that of the traditional IEEE wireless network technology. Moreover, the latest *fifth-generation (5G)* combined with *device-to-device (D2D)* communication technology is used to improve transmission quality for users, and it achieves a higher data rate even in high-speed movement. Therefore, the 5G *New Radio (NR)* international standard 3GPP has already regarded D2D as an extremely important application scenario in the communication technologies of R15 [8, 9] and R16 [10, 11]. At present, the 4G LTE-A networks and 5G NR cooperate with each other and provide mobile network services. *Media access control (MAC)* layer data scheduling in 4G LTE-A and 5G NR has many technologies in common. To support multimedia services and high-bandwidth data delivery, the LTE-A MAC layer supports *quality of service (QoS)* with different *QoS class indicator (QCI)* [6, 7] levels. Therefore, some researchers have tried to adopt *max throughput (MT)*, *earliest deadline first scheduling (EDF)*, *proportionally fair (PF)*, or *blind equal throughput (BET)* algorithm as LTE-A MAC scheduler in *evolved NodeB (eNB)* (in the center of Figure 1) to maximize throughput or allocate a fairness bandwidth. However, based on LTE-A current QCI priority and QoS requirement in user equipment (UE), there is no appropriate scheduling scheme to fit different traffic flow types in a single UE. The proposed approach may be suitable for different rate and delay requirements of UE and can achieve the fairness goal in real-time traffic flows of various UE.

The remainder of this study is organized as follows. An introduction network scheduling and brief survey of LTE-A scheduling are presented in Section 2. The proposed network user satisfaction-based scheduling approach in the LTE-A network is presented in Section 3. Performance evaluation with several scenarios is presented in Section 4. Finally, Section 5 concludes this study.

2. Related Work

2.1. LTE-A MAC Layer. In the LTE-A network, the basic time unit for packet scheduling and transmission period is called a *transmission time interval (TTI)* with a length of 1 ms and there is 10 ms as an LTE-A radio frame. Thus, TTI is the time unit for resource allocation in LTE-A. In each TTI, a scheduling decision is made, in which each scheduled UE is assigned a certain number of radio resources in the time and frequency domain. In the time domain, a TTI is split into two slots (one slot is 0.5 ms). Each slot comprises seven *orthogonal frequency-division multiplexing (OFDM)* symbols in the case of the normal cyclic prefix length. In the frequency domain, resources are grouped into units of 12 subcarriers, such that of one unit of 12 subcarriers for a duration of one slot is called an RB, which is the smallest element of resource allocation. The smallest unit of a resource is a *resource element (RE)* that comprises one subcarrier for a duration of one OFDM symbol. Therefore, an RB is comprised of 84 (7×12) REs in the case of the normal cyclic prefix length in Figure 2. The channel capacity was assumed to be static for traditional MAC scheduling, and it was revised for the LTE-A network environments. In LTE-A, an eNB typically selects the *modulation and coding scheme (MCS)* depending on a prediction of the *downlink (DL)* channel condition, which is according to the UE's CQI report transmitted (Figure 1). The 3GPP LTE-A has given a table of references for the efficiency of each CQI index (CQI ranges from 1 to 15 by the modulation type of 64QAM, 16QAM, and QPSK) as Table 1.

2.2. Related Research. In the 4G/5G wireless broadband network research, the research on the scheduling of various MAC layer networks has always been the focus of many researchers, such as the research on handover scheduling in homogeneous or heterogeneous mobile networks [12–14] and research on multimode QoS guarantee [15–18]. In LTE-A mobile network scheduling research fields, some research studies focus on the discussion of uplink scheduling [19–21], but most of the research studies focus on downlink scheduling. In addition, to support network managers to allocate resources within the limited wireless network

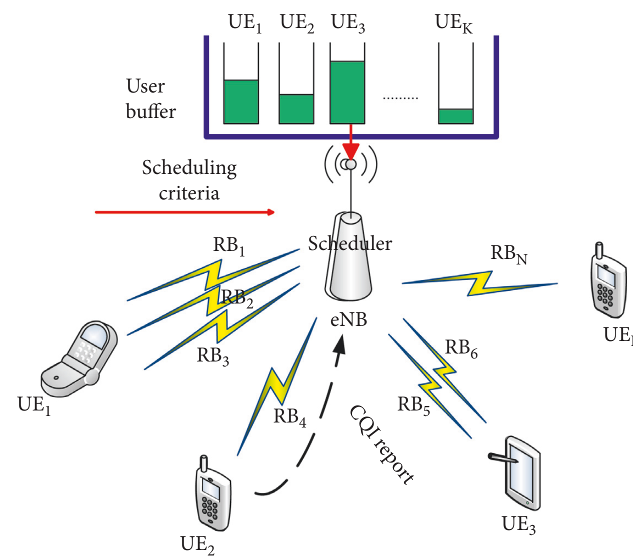

Figure 1: Diagram of LTE-A MAC scheduling.

bandwidth, it is necessary to incorporate QoS considerations in scheduling.

For LTE-A downlink scheduling, Biernacki et al. [22] proposed some fairness algorithms to find a balance between different QoS type traffic to avoid starvation at low QoS-level traffic flows. The article [23] had proposed a scheme that can support real-time VoIP traffic and dynamically adjust traffic rates to avoid buffer overflow. Considering the scheduling algorithm and QoS support at the same time, Aminu et al. [24] conducted a survey on many scheduling algorithm mechanisms for the LTE-A MAC layer scheduling mechanism and the characteristics of QoS considerations. Some comparative analyses have been performed on their research results to explore the various parameters of the various downlink scheduling algorithms for resource allocation. Nasralla et al. [25] discussed and analyzed many current QoS-aware downlink scheduling algorithm mechanisms in the LTE-A networks and divided these mechanisms into four main classes: delay aware, queue aware, target bit rate aware, and hybrid aware. They also proposed the use of the hybrid-aware category as a conceptual mechanism to propose a resource allocation scheduling algorithm that considers QoS while taking fairness into account. In doing so, there can be a certain degree of scheduling fairness in the face of real-time and non-real-time traffic.

Some MAC scheduling schemes comprise the time-domain and the frequency-domain scheduler in the LTE-A network. Wang et al. [26] proposed a novel packet scheduling algorithm based on *frequency domain (FD)* prediction and *time-domain (TD)* grouping technique for real-time applications in the downlink LTE-A system. The proposed algorithm, which is robust to simultaneous multiple channel defects, is proven to satisfy QoS requirements for real-time users and achieve the rate requirement. Grøndalen et al. [27] applied the standard TD scheduling mode, where all RBs in a subframe can be allocated to a single UE. They also considered their FD mode, where the RBs in each subframe can

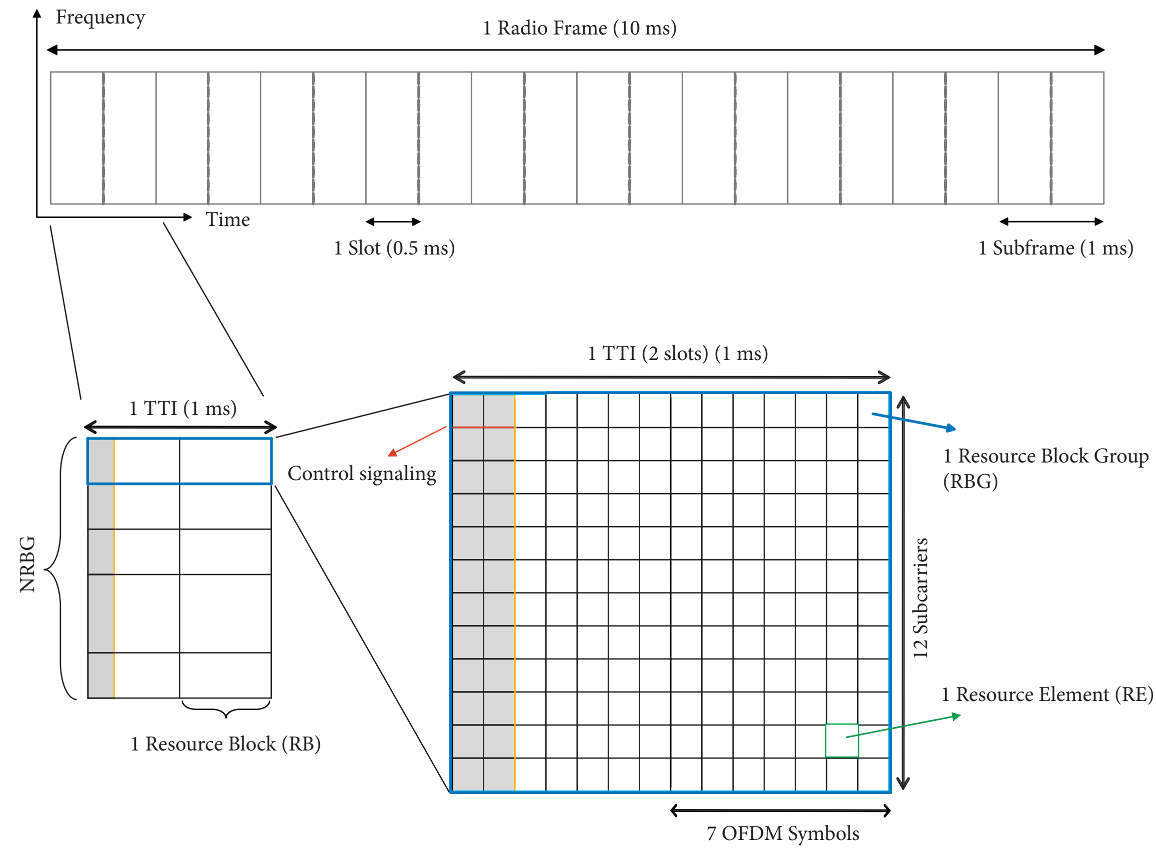

Figure 2: Basic resource structure of LTE-A frame.

Table 1: CQI table by 3GPP.

CQI index	Modulation	Approximate code rate	Efficiency (bits/RE)
0	No Tx	—	—
1	QPSK	0.076	0.1523
2	QPSK	0.12	0.2344
3	QPSK	0.19	0.3770
4	QPSK	0.3	0.6016
5	QPSK	0.44	0.8770
6	QPSK	0.59	1.1758
7	16QAM	0.37	1.4766
8	16QAM	0.48	1.9141
9	16QAM	0.6	2.4063
10	64QAM	0.45	2.7305
11	64QAM	0.55	3.3223
12	64QAM	0.65	3.9023
13	64QAM	0.75	4.5234
14	64QAM	0.85	5.1152
15	64QAM	0.93	5.5547

be allotted to multiple UEs. The scheduling indexes of the schedulers have been accordingly changed to account for the finer RB granularity in resource allocation and the richer CQI returned by each UE. Furthermore, both the TD and FD versions of the proposed scheduling have been implemented and adopted throughput guarantees. Tuan et al. [28] integrated both time and FD scheduling as the proposed algorithm, and it achieved the QoS requirement for real-time flows and provided fairness to non-real-time flows.

To develop a scheduler, which can consider different real-time traffic flow's requirements, might be necessary for much real-time traffic UE in the LTE-A downlink environment. However, current research scheduling schemes may only focus on the rate or delay of one aspect to improve the wireless network quality satisfaction purpose. Moreover, to apply both the time and FD concept schedulers and consider both rate and delay real-time user requirements in the LTE-A network is very important to design MAC scheduling.

The proposed hybrid network user satisfaction-based downlink scheduling approach is named *max rate delay urgency first (MRDUF)*. The main contributions of this study are summarized as follows:

(i) We propose the idea of simultaneously considering the UE's traffic flow's both rate and delay requirements.

(ii) We propose the idea to achieve the goal of fairness in all UE different real-time traffic flows.

(iii) By considering both the time and frequency domains as the hybrid and two-stage scheduling to improve the scheduling performance, our proposed idea can provide a flexible network quality scheduling approach with low computing.

(iv) To reduce huge computing overhead from different UE's traffic in a TTI period, our proposed idea has also designed a threshold of *virtual scheduling list (VSL)*, which can help the resource allocation mechanism to decide how many candidates downlink delivery packets to the next stage.

(v) By considering both the rate and delay requirements for each real-time traffic flow, our proposed MRDUF approach can have good performance for the fairness gain of satisfaction in rate and delay for the LTE-A networks.

3. Methods: Network User Satisfaction-Based Downlink Scheduling Approach

3.1. Introduction to Wireless Network Downlink Scheduling Algorithm. In LTE-A wireless network, a basic time unit for packet scheduling and transmission is called a TTI with a length of 1 ms. So, TTI is the time unit for LTE-A MAC layer resource allocation. In each TTI period, a downlink scheduling decision is made where each scheduled UE is assigned a certain amount of radio resources in the time and frequency domain. An eNB is the major entity in charge of performing the resource allocation. A scheduler functionality (in an eNB) can handle downlink packet scheduling tasks for connected UE in Figure 1. The radio channel capacity is assumed to be static for the traditional MAC scheduling scheme; it was revised for the LTE-A network environments. In the LTE-A network, an eNB typically selects the MCS depending on a prediction of the DL channel condition from UE (as UE_1-UE_K of Figure 1), which is according to each UE's CQI report transmitted. The 3GPP LTE-A standard has given a table of reference for the efficiency of each CQI index (CQI ranges from 1 to 15 by modulation type of 64QAM, 16QAM, and QPSK) in Table 1. Estimation of the channel capacity depends on the CQI reports from a UE, meaning that different UE have different views of the channel capacity. Since the current per RB capacity via each UE is dynamic, it is important to schedule these DL RBs to appropriately meet a UE's requirement. Moreover, an eNB should simultaneously consider many UE throughput and fairness requirements. Finding a balance between the two requirements is an important research topic for the wireless network area.

3.1.1. Max Throughput (MT). This study introduces the conventional MT algorithm [29], which can maximize throughput. MT algorithm will select users according to the channel conditions of UE and eNB upon selection, i.e., it will select the user with the optimum channel and allocate radio resources to this user, so the user with an optimum channel at that time always can be selected. For example, the equation (1) below is the equation of the MT algorithm:

$$m_{j,k}^{\mathrm{MT}} = \max_i \left\{ d_k^i(t) \right\}, \tag{1}$$

where both i and j indicate UE, k indicates RB, $m_{j,k}^{\mathrm{MT}}$ indicates that RB_k is allocated to UE_j according to the MT algorithm, and $d_k^i(t)$ indicates the expected amount of data available by UE_i at the k^{th} RB in the t^{th} TTI. Therefore, it can be seen from equation (1) that the MT algorithm only considers the allocation of RB to UE with optimum CQI value so that RB can provide the UE with the highest number of bits to achieve the goal of maximizing throughput but does not consider other factors.

Figure 3 is the schematic of the MT algorithm. The diagram has three users, namely, UE1, UE2, and UE3, whose channel quality changes with time. As the MT algorithm allocates resources to the user with optimum channel quality at that time, and UE1 has the optimum channel quality in the first time interval in the figure, the MT algorithm allocates the resources to UE1 as per the algorithm characteristics. While UE2 has the optimum channel quality at that time in the next time interval, the MT algorithm allocates the resources to UE2. Thus, as the channel quality changes over time, the user with optimum channel quality is UE1, UE1, UE1, UE3, UE2, UE2, and UE1 in sequence, and the MT algorithm allocates resources to the users in this order.

3.1.2. Blind Equal Throughput (BET) and Proportional Fair (PF). If throughput and fairness are simultaneously considered, BET [20] and PF [29] algorithms can be adopted. This is because all users want to achieve a certain balance when contending for radio resources so that all users can be provided with similar service levels. These two algorithms record the average throughput achieved in the past as shown in equation (2) and keep updating. The following are the equations of BET and PF algorithms:

$$\overline{R_i(t)} = \alpha * \overline{R_i(t-1)} + (1-\alpha) * r^i(t), \tag{2}$$

$$m_{j,k}^{\mathrm{BET}} = \max_i \left\{ 1/\overline{R_i(t-1)} \right\}, \tag{3}$$

$$m_{j,k}^{\mathrm{PF}} = m_{j,k}^{\mathrm{MT}} * m_{j,k}^{\mathrm{BET}}, \tag{4}$$

where both i and j indicate UE, k indicates RB, $m_{j,k}^{\mathrm{BET}}$ indicates that RB_k is allocated to UE_j according to the BET algorithm, $m_{j,k}^{\mathrm{PF}}$ indicates that RB_k is allocated to UE_j according to PF algorithm, $r^i(t)$ indicates the amount of data obtained by UE_i in the period t, $\overline{R_i(t)}$ indicates the past average throughput of UE_i up to the present time t expressed in exponential average, and α indicates the weight value from 0 to 1, as shown in equation (2). Therefore, it can be seen from equations (3) and (4) that both BET and PF algorithms constantly update the past average throughput of users to achieve the fairness of throughput. In contrast, the PF algorithm can consider the channel quality of users at that time via $m_{j,k}^{\mathrm{MT}}$ more than the BET algorithm.

3.1.3. Earliest Deadline First (EDF). The EDF algorithm [31] is a commonly used algorithm in real-time data traffic, and real-time traffic always has a delay budget to which the traffic belongs. When the *head-of-line (HOL)* delay of a packet exceeds its delay budget, the packet is discarded. Therefore, when selecting users, the EDF algorithm decides according

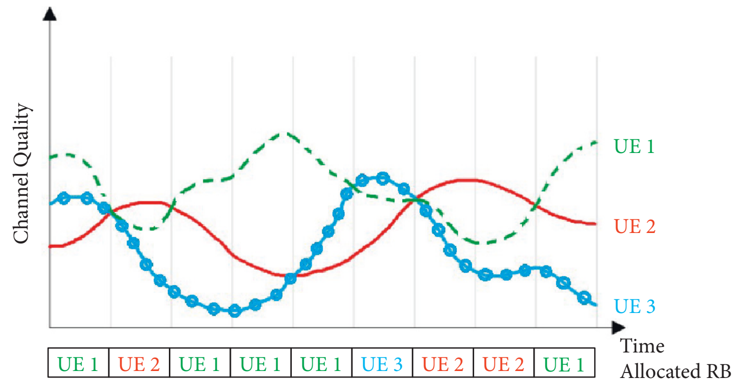

Figure 3: MT algorithm for RB allocation based on UE's channel quality.

to the current time span between HOL delay and the delay budget of the users at that time. The following is the equation of the EDF algorithm.

$$m_{j,k}^{\mathrm{EDF}} = \max_{i}\left\{1/\left(\tau_i - d_{\mathrm{HOL},i}\right)\right\}, \quad (5)$$

where both i and j indicate UE, k indicates RB, $m_{j,k}^{\mathrm{EDF}}$ indicates that RB_k is allocated to UE_j according to the EDF algorithm, τ_i indicates the delay budget of UE_i, and $d_{\mathrm{HOL},i}$ indicates HOL delay, which is the current delay of the first packet of UE_i in the buffer of eNB. In equation (5), the EDF algorithm allocates the resource based on the time span between the HOL delay and the delay budget. Thus, when the HOL delay of a UE is close to the delay budget to which it belongs, it indicates that this packet will be discarded soon, so it gets high priority.

3.2. Proposing a Hybrid Network User Satisfaction-Based Downlink Scheduling. Current scheduling algorithms, such as MT, BET, PF, and EDF, only consider all UE rate or delay requirements. These one-stage scheduling schemes cannot fit the individual UE's requirement with different traffic specifications. In the time unit of each TTI, all users simultaneously contend for all RBs, and for each RB, the user with the highest priority is selected and provided with this RB. For example, there are 5 UE to be scheduled in Figure 4. When each TTI is scheduled, 5 UE simultaneously contend for 10 RBs in each TTI, and for each RB, the UE with the highest priority is selected and provided with this RB, for example, RB1 is allocated to UE3.

To provide a flexible network quality scheduling approach with lower computing effort, we consider both time and FDs as the hybrid and two-stage scheduling to improve the scheduling performance. In Figure 5, during each TTI, several UE are selected from all users, of which the process is called *time-domain packet scheduler (TDPS)* processing mechanism, and then, RB resources are actually allocated to the UE selected as per TDPS, of which the process is called *frequency-domain packet scheduler (FDPS)*. There are 5 UE to be scheduled in Figure 5. In this TTI scheduling, UE1, UE3, and UE5 are first selected as per TDPS, and the results of TDPS are input to the frequency domain. These 3 UE then contend for all RBs in this TTI, and for each RB, UE with the highest priority is selected and provided with this RB, for example, RB2 is allocated to UE1. Thus, the proposed two-stage scheduling approach can effectively reduce the computational load in the case of many UE. Some MAC scheduling schemes did include both time and frequency domains [26–28]. Some studies had focused only on the rate of satisfaction [26, 27], while the other considered only the fairness goal for non-real-time traffic flows [28]. The proposed MRDUF approach is based on UE's requirements and can appropriately allocate radio resources for UE real-time traffic in the LTE-A network. In addition, considering the requirements of rate and delay, an algorithm for fairness between different satisfactions is proposed according to the different requirements of rate and delay among UE. Moreover, two-stage scheduling is adopted. First, in the first-stage TDPS, the rate or delay urgency of each UE is calculated according to the different requirements of rate and delay among UE, and the specific UE with high urgency is selected to FDPS to allocate RB, with the scheduling algorithm of TDPS named as *MRDUF*. In the second-stage FDPS, two existing algorithms' concepts are applied according to the different RB allocation methods, namely, *first come first serve (FCFS)* and *MT*.

3.2.1. TDPS: MRDUF. In our proposed approach, the first stage is called MRDUF. The MRDUF scheme calculates real-time UE rate distribution and urgency of the UE delay budget to allocate the priority of each UE traffic. The calculation formula equations (6)–(8) are as follows:

$$U_{\mathrm{rate},i} = 1 - \min\left(\frac{\overline{r_i}}{\mathrm{GBR}_i}, 1\right), \quad (6)$$

$$U_{\mathrm{delay},i} = \frac{d_{\mathrm{HOL},i}}{D_{\mathrm{QCI},i}}, \quad (7)$$

$$\mathrm{TD\_}P_i = \max\left(U_{\mathrm{rate},i}, U_{\mathrm{delay},i}\right). \quad (8)$$

The $U_{\mathrm{rate},i}$ is the UE_i's traffic rate, and the $U_{\mathrm{delay},i}$ is the delay level. Moreover, we add the $\mathrm{TD\_}P_i$ as the priority parameter of UE_i, the $\overline{r_i}$ is the average bit rate of UE_i in current TTI period, the GBR_i is the requirement bit rate of UE_i, the $d_{\mathrm{HOL},i}$ is current UE_i's head of line in the packet

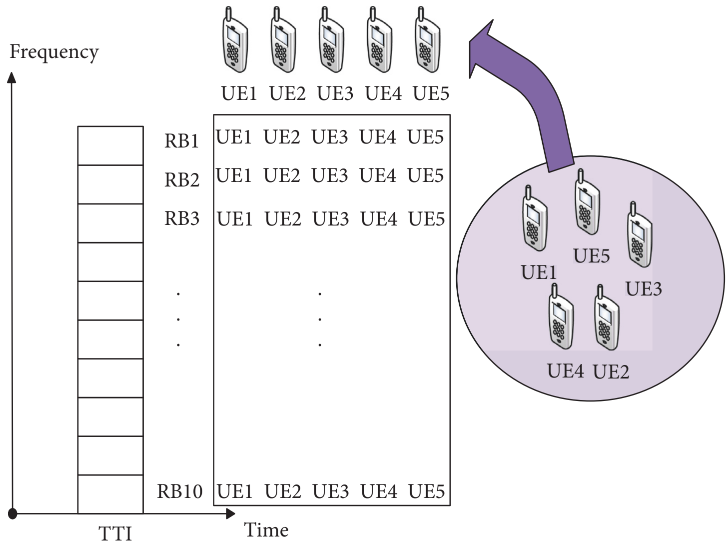

Figure 4: Diagram of one-stage scheduling.

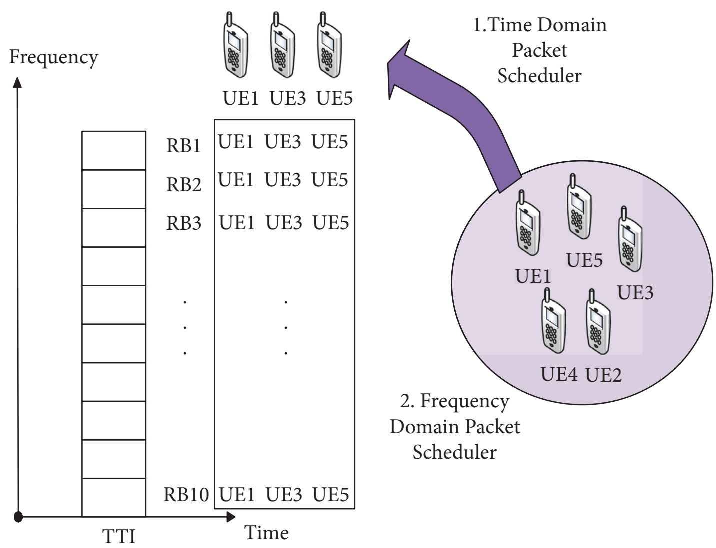

Figure 5: Diagram of two-stage scheduling.

buffer, and the $D_{\mathrm{QCI},i}$ is UE_i's delay budget of UE_i's QCI level. Using these formulas, the larger the TD_P value is, the higher is the emergence level. Thus, a bigger gap between the current rate and requirement rate or delay approaching delay budget raises the value of TD_P. Furthermore, we design a VSL to schedule how many candicate packets from different UE's traffic in a TTI period, one TTI delivery maximum capacity based on VSL_Threshold as equation (9)

$$\mathrm{VSL\_Threshold} = N_{\mathrm{RBG}} * \left(N_{\mathrm{OFDM}}^{\mathrm{TTI}} - N_{\mathrm{OFDM}}^{\mathrm{Ctrl}}\right) * 12\,(\mathrm{subcarriers}) * \mathrm{Eff}\left(\mathrm{CQI}_{\mathrm{max}}\right). \quad (9)$$

In the LTE-A network, two RBs are the *resource block group (RBG)*, and then, the N_{RBG} is the number of RBG in a TTI period. The number of OFDM symbol is $N_{\mathrm{OFDM}}^{\mathrm{TTI}}$ in a TTI period, the number of the OFDM symbol is $N_{\mathrm{OFDM}}^{\mathrm{Ctrl}}$ in the control channel, and then, the $\mathrm{Eff}(\mathrm{CQI}_{\mathrm{max}})$ is the best CQI's efficiency.

To calculate each UE's TD_P value that could sort the priority sequence, a UE traffic buffer with the largest value of TD_P can have the chance to select a packet into the VSL candidate buffer queue as step 1 of Figure 6, and UE1 has the largest value of *0.923*. After the packet has moved from the UE buffer to VSL, each UE's TD_P should be updated. The procedure of the MRDUF scheme would be continuously run until the next selected UE's packet might exceed the VSL_Threshold as the last step of Figure 6. The VSL buffer is the delivery packet list for second-stage FDPS.

3.2.2. FDPS: FCFS. Since the VSL is a selected queue system, our proposed second-stage FCFS scheduler should allocate real RB resources according to the VSL queue sequence

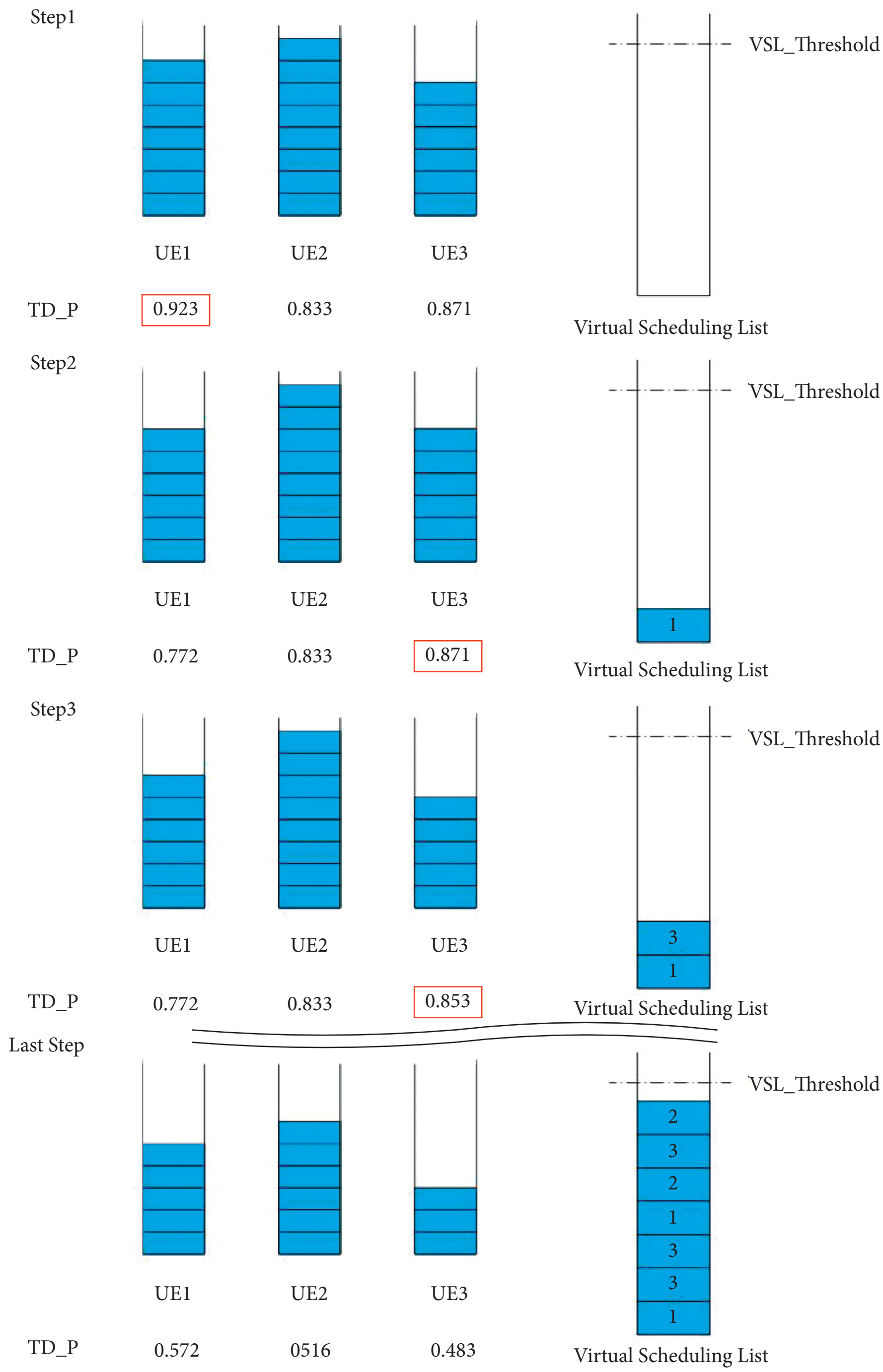

Figure 6: MRDUF approach diagram.

(Figure 7). Then, the packets sorted in VSL are allocated in sequence. In Figure 7(a), the first sequential packet belongs to UE1, which means that UE1 has the highest urgency, so UE1 can preferentially select the optimum RB (Figure 7(b)). The second priority is the packet of UE3 (Figure 7(c)), and the above steps are repeated. There are two finished policies, one is no more UE packets in the VSL queue, and the other one is no more free RB space for allocation (Figure 7(d)).

3.2.3. Enhanced FDPSL: MT + FCFS. Suppose FDPS allocates RB to the users as per the FCFS algorithm, in that case, it will be limited by the priority of the packet in VSL, which degrades the efficiency of RB, resulting in the degradation of overall throughput, as shown in Figure 7(d). Although RB4 still has space for UE3, such space in RB4 can only be wasted because the next urgency sequence is not the packet of UE3.

First of all, the VSL of TDPS is slightly modified by adding a lower threshold ($\text{VSL\_Threshold}_{\text{lower}}$), which could

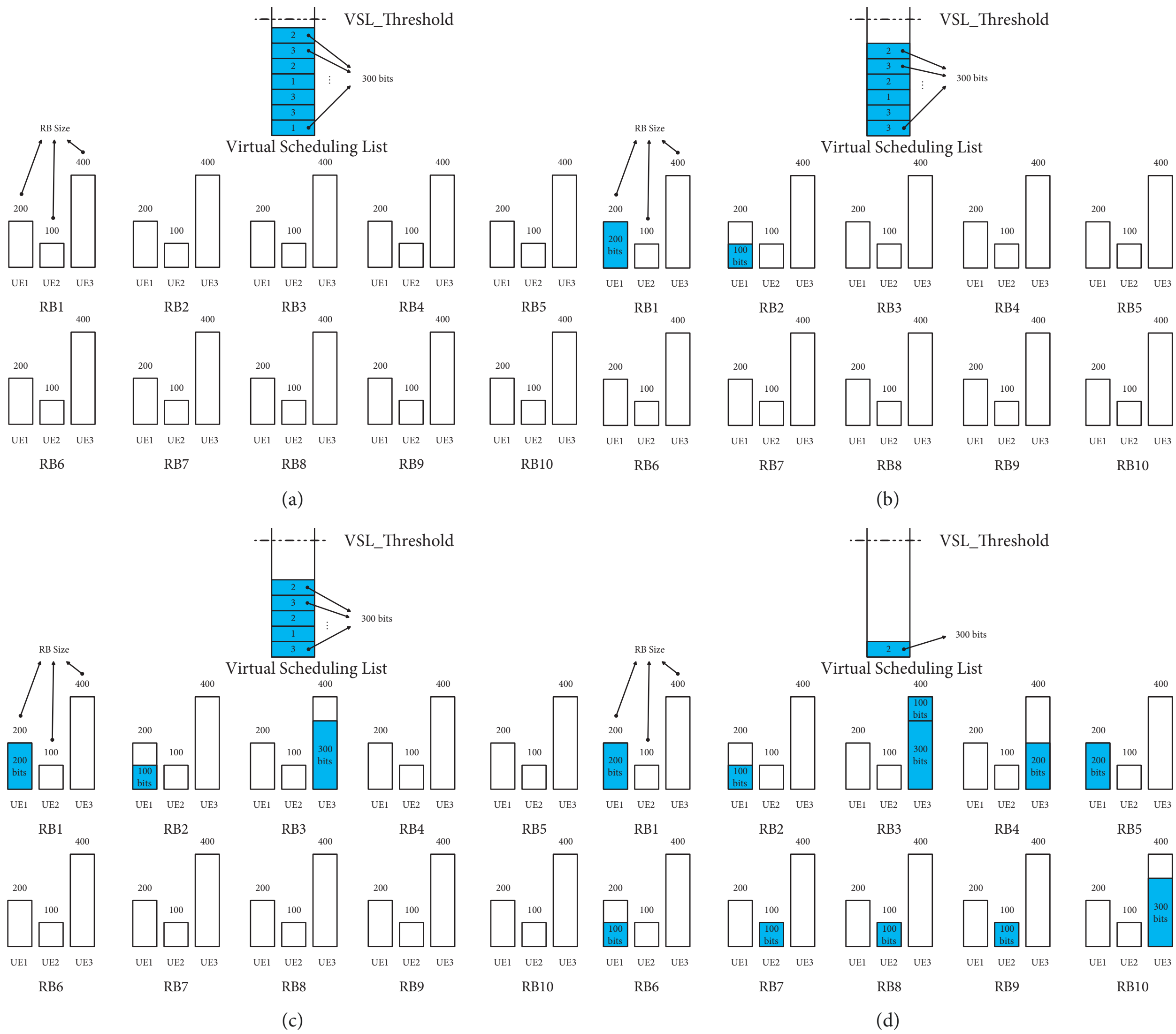

Figure 7: FCFS diagram.

disregard the urgency order of the packet in VSL and is estimated by the current CQI of all UE, as shown in equation (10), where $\text{Eff}(\text{CQI}_{\text{avg}})$ is the average efficiency corresponding to all UE CQI values. The higher threshold is defined as ($\text{VSL\_Threshold}_{\text{upper}}$), which is the original threshold, and the calculation method is as shown in equation (9). Different algorithms calculate these two thresholds in the FDPS.

$$\text{VSL\_Threshold}_{\text{lower}} = N_{\text{RBG}} * \left(N_{\text{OFDM}}^{\text{TTI}} - N_{\text{OFDM}}^{\text{Ctrl}}\right) * 12\,(\text{subcarriers}) * \text{Eff}\left(\text{CQI}_{\text{avg}}\right). \tag{10}$$

The allocation process of the enhanced FDFS will determine the allocation order of RB according to $\text{VSL\_Threshold}_{\text{lower}}$ and $\text{VSL\_Threshold}_{\text{upper}}$, while for packets with a sequence lower than $\text{VSL\_Threshold}_{\text{lower}}$, the resources are allocated according to the MT algorithm in Section 3.1.1 regardless of the urgency. For example, the yellow packets in Figure 8 are irrespective of their sequence; while for packets with a sequence higher than $\text{VSL\_Threshold}_{\text{lower}}$, RB is allocated according to the method specified in Section 3.2.2, such as the green packets in Figure 8.

The allocation process of the enhanced FDFS determines the allocation order of RB according to $\text{VSL\_Threshold}_{\text{lower}}$ and $\text{VSL\_Threshold}_{\text{upper}}$, while for packets with a sequence lower than VSL_E, the resources are allocated according to the MT algorithm in Section 3.1.1, regardless of the urgency. For example, the yellow packets in Figure 8 are irrespective of their sequence, while for packets with sequence higher than $\text{VSL\_Threshold}_{\text{lower}}$, RB is allocated according to the method specified in Section 3.2.2, such as the green packets in Figure 8.

The RB allocation process of the enhanced FDFS is as shown in Figure 9. The MT algorithm is adopted to allocate RBs because the sequence of yellow packets is lower than $\text{VSL\_Threshold}_{\text{lower}}$, and UE3 in the figure has the optimum

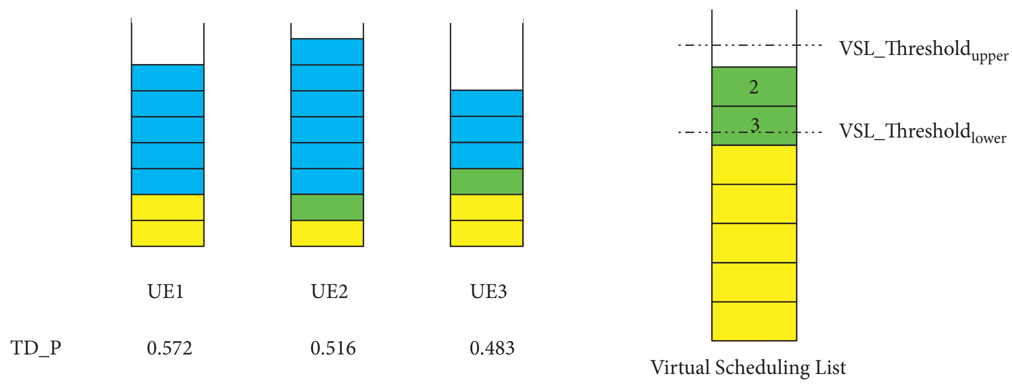

FIGURE 8: Enhanced FDFS virtual scheduling list diagram.

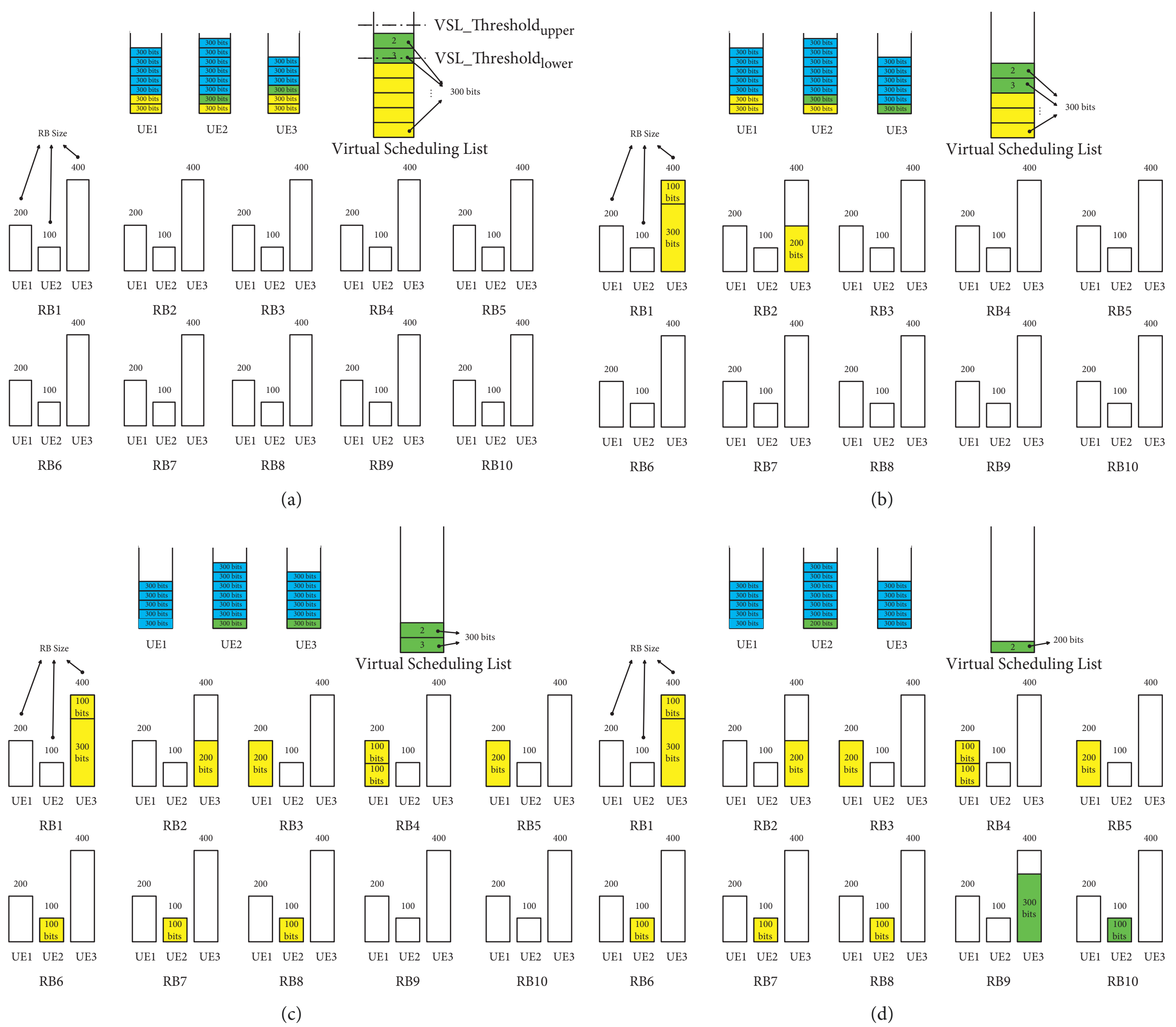

FIGURE 9: Enhanced FDFS resource allocation diagram.

channel quality, so it can preferentially select the optimum RB until the resources for packets of UE3 in VSL with a sequence lower than VSL_Threshold$_{lower}$ are completely allocated as shown in Figure 9(b). The above actions are repeated until all yellow packets (whose sequence is lower than VSL_Threshold$_{lower}$) are allocated as shown in

Figure 9(c). In case of any remaining RBs, they are allocated to green packets (whose sequence is higher than VSL_Threshold$_{lower}$) as per the original FCFS algorithm as shown in Figure 9(d). By comparing Figure 9(d) with Figure 7(d), it can be found that the enhanced FDFS method can transfer 100 bits more packets in UE2, which improves the overall performance.

The two-stage schemes are shown in Figure 10 which have been, respectively, scheduled from time and FD consideration with the VSL concept that is the detail of our proposed satisfaction-based scheduling approach, and it can achieve both rate and delay requirements for different UE traffic behavior.

3.2.4. Discussion of UE's Traffic Behavior. In our proposed approach, it should support both rate and delay requirements for different UE's traffic flows. The audio and video traffic flow types should be assigned, and then, the different aspects of the delay budget should be added as the four real-time traffic types (VoIP, video, online radio, and video conference) for performance evaluation. Poisson-like real-time traffic would be easier to analyze these scheduling schemes. The analysis of different proportions of these four types is also necessary for further investigation.

4. Results and Discussion

4.1. Experimental Setup and Performance Criteria. A custom program developed by Microsoft Visual C++ 6.0 for system-level simulation is extended from the authors' laboratory experiment tools in the LTE environment. There is only one eNB and four real-time traffic types including VoIP, video, online radio, and video conference, as shown in Table 2 in our simulation environment. The proportion of these four types in the basic simulation environments (VoIP: Video: Online Radio: Video Conference) is 1 : 1 : 1 : 1. The four types of real-time traffics are designed as uniform distribution, and the detailed simulation parameters are shown in Table 3. The input load for each UE flow is equal, and the packet arrival process is Poisson. The size of a packet is fixed, and the two levels of packet size, 800 bits for audio traffic type and 8000 bits for video traffic type, are simulated. All UE moves in a random direction within the eNB signal range by means of random way point as shown in Figure 11, whose movement speeds are random within 15 m/s–20 m/s. It is assumed that the number of packets received in each TTI is in Poisson distribution; then, the resources are allocated according to the CQI value reported by the bottom layer. The channel quality for each UE is simulated by directly drawing a random number from the range of CQI values. The number of bits that can be carried in a resource block is calculated according to the efficiency of the CQI value.

The performance evaluation and analysis of MRDUF/FCFS and MRDUF/MT schemes proposed in this study and four existing scheduling algorithms (MT, PF, BET, and EDF) are performed.

The performance analysis criteria are total throughput, rate satisfaction, delay satisfaction, and fairness gain of rate/delay satisfaction. The definition of performance criteria is as follows:

(i) Total throughput: (total received bit per second in all UE).

(ii) Rate satisfaction: (average actual UE's bit rate)/(average UE's requirement bit rate) × 100%, bigger value would indicate higher rate satisfaction level.

(iii) Delay satisfaction: (number of delivery packet under delay budget)/(number of arrival packet) × 100%, bigger value would indicate higher delay satisfaction level.

(iv) Gain of fairness of rate/delay satisfaction: (standard deviation (SD) of MT scheme–SD of other schemes)/(SD of MT scheme) × 100%, the bigger value would indicate fairer rate/delay satisfaction level.

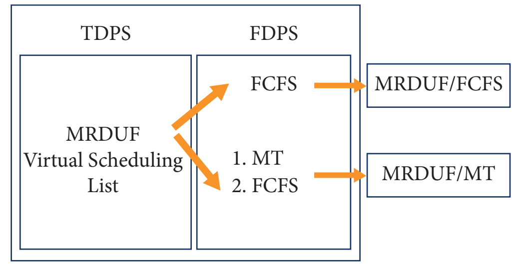

Figure 10: Overview of proposed network user satisfaction-based scheduling.

Table 2: Real-time traffic flow type.

Parameter	VoIP	Online radio
Bite rate	10 kbps	10 kbps
Packet size	800 bits	800 bits
Delay budget	50 ms	300 ms
Parameter	Video	Video conference
Bite rate	250 kbps	250 kbps
Packet size	8,000 bits	8,000 bits
Delay budget	300 ms	50 ms

Table 3: Simulation parameters.

Parameter	Value
Simulation time	10,000 ms
Cell radius	1.7 km
Number of UE	24–96
Number of RBGs/TTI	50
UE mobility	Random way point
UE speed	Random (15 m/s ~ 20 m/s)
CQI reporting type	Wideband CQI

4.2. Experimental Results. The performance evaluation studies have been shown in Sections 4.2.1–5. Initially, the simulation must identify the performance goal for our proposed approach and contrasts. The rate and delay satisfaction are very important for real-time traffic flows, so performance evaluation discusses the fairness of satisfaction,

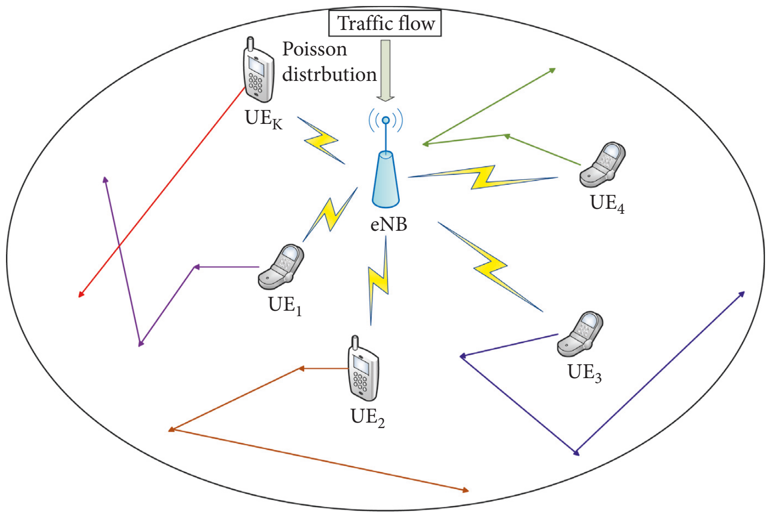

FIGURE 11: Random way point movement in simulation.

throughput, and the rate and delay satisfaction. Moreover, the detailed effect of the number of UE on user satisfaction should be identified.

4.2.1. *Fairness of Satisfaction.*

Our proposed schemes can improve both rates and delay fairness satisfaction. The results are shown in Figures 12–21. First, the performance analysis is discussed for different methods based on the fairness of satisfaction of total UE traffic flows. Since the MT algorithm does not consider the requirements of UE traffic flow but only considers UE channel quality as the allocation criterion of radio resources, it is adopted as the benchmark for performance comparison based on the fairness of satisfaction.

(i) As shown in Figures 12 and 13, MRDUF/FCFS and MRDUF/MT schemes feature better fairness of satisfaction of rate and delay than other scheduling algorithms applied in the control group. The proposed approach considers the rate and delay requirements of different real-time data traffic and schedule according to the urgent priority of rate and delay requirements.

(ii) As shown in Figure 12, the number of UE increases from 32, and as the UE number increases, i.e., the increase of load in the system, a larger difference in effect can be found. As the approach proposed in this study must take into account the overall requirements of all UE and also takes into account consideration fairness, the overall fairness of satisfaction cannot be lost due to the increase in the number of UE. When the number of UE is increased to 96, the gain of fairness of rate satisfaction for MRDUF/FCFS and MRDUF/MT schemes is about 50% and 40%, respectively, compared to that for the MT algorithm.

(iii) For the fairness of delay satisfaction, as shown in Figure 13, as the algorithm is related to the number of packets successfully sent from the delay budget, the performance graph of fairness of delay satisfaction is similar to that of fairness of rate satisfaction. The gain of fairness of delay satisfaction for MRDUF/FCFS and MRDUF/MT schemes is about 50% and 40%, respectively, compared to that for the MT algorithm and is about 10% compared to that of the EDF algorithm.

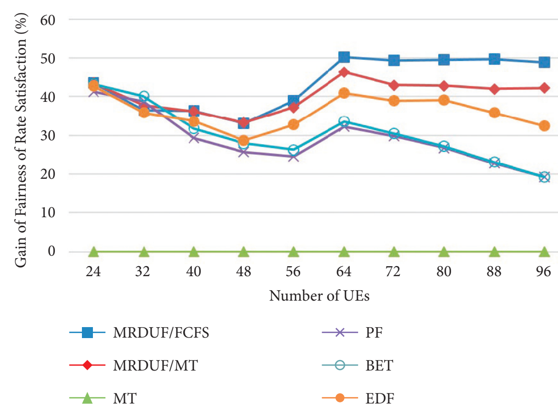

FIGURE 12: Gain of fairness of rate satisfaction.

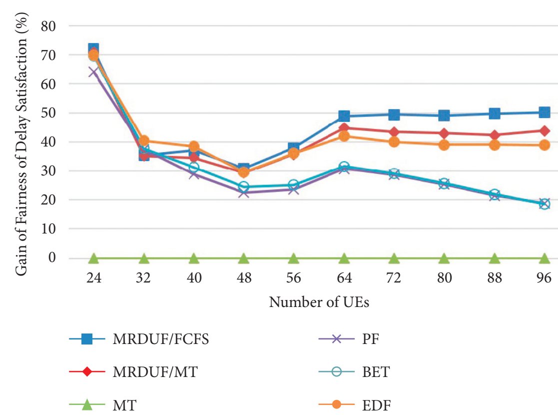

FIGURE 13: Gain of fairness of delay satisfaction.

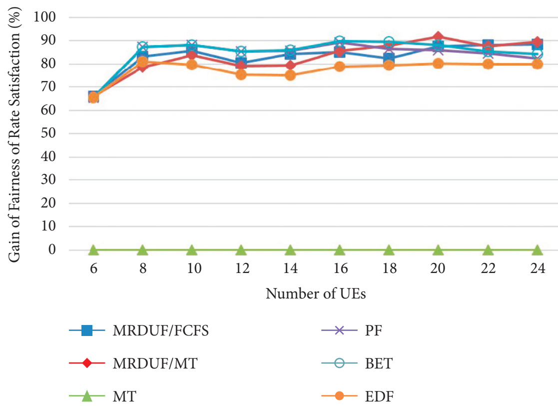

Figure 14: Gain of fairness of rate satisfaction in VoIP flows.

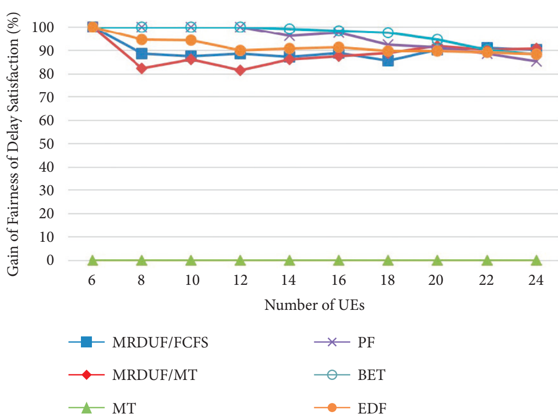

Figure 15: Gain of fairness of delay satisfaction in VoIP flows.

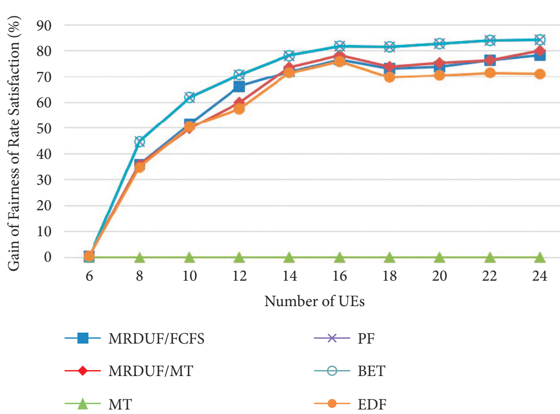

Figure 16: Gain of fairness of rate satisfaction in online radio flows.

(iv) It can be seen from Figures 12 and 13 that the fairness of rate satisfaction and delay satisfaction of the MRDUF/FCFS algorithm is better than that of the MRDUF/MT algorithm because the RB allocation in the FDPS of the MRDUF/FCFS algorithm is based on the urgency of requirements in the TDPS, while the RB allocation in the FDPS of the MRDUF/MT algorithm considers the effective improvement of the utilization rate of RB, so some packets will not be served according to the original urgency priority, and its fairness of satisfaction is relatively lower.

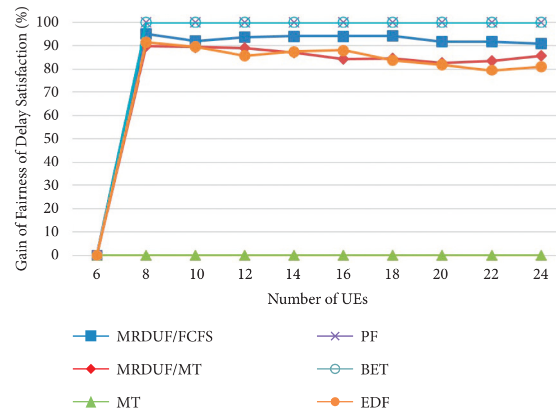

Figure 17: Gain of fairness of delay satisfaction in online radio flows.

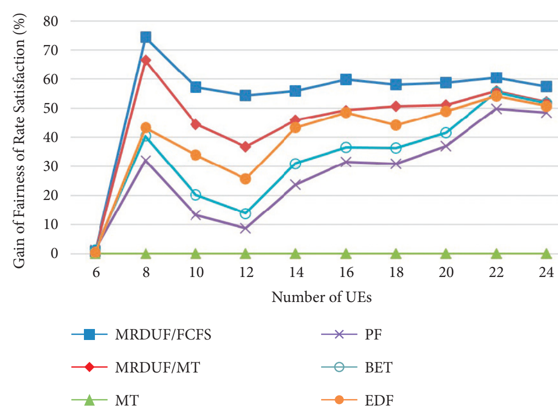

Figure 18: Gain of fairness of rate satisfaction in video flows.

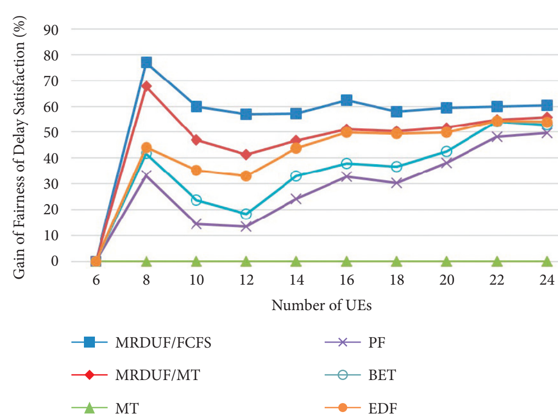

Figure 19: Gain of fairness of delay satisfaction in video flows.

Then, the performance analysis for fairness of satisfaction of different real-time traffic flows is discussed. Figures 14 and 15 show the fairness of rate satisfaction and fairness of delay satisfaction of VoIP, respectively. Figures 16 and 17 show the fairness of rate satisfaction and fairness of delay satisfaction of online radio, respectively. Figures 18

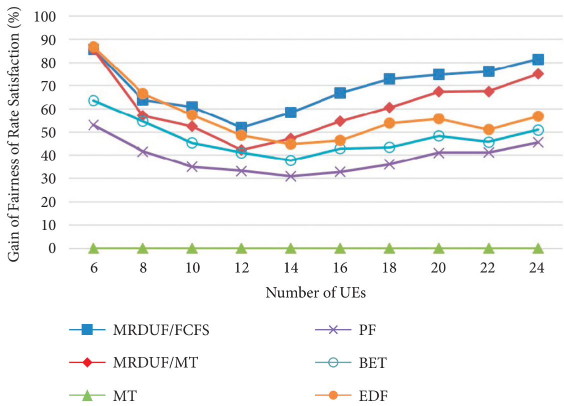

Figure 20: Gain of fairness of rate satisfaction in video conference flows.

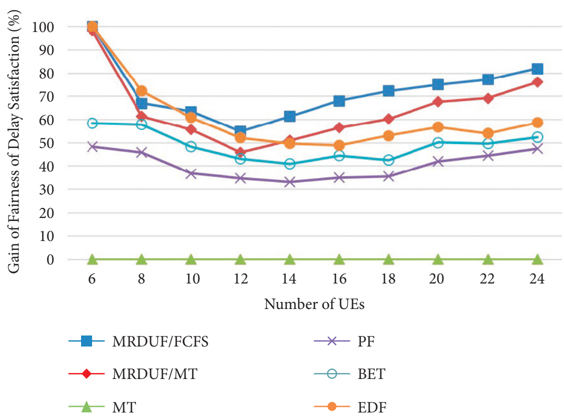

Figure 21: Gain of fairness of delay satisfaction in video conference flows.

and 19 show the fairness of rate satisfaction and fairness of delay satisfaction of video, respectively. Figures 20 and 21 show the fairness of rate satisfaction and fairness of delay satisfaction of video conference, respectively.

(i) As VoIP and online radio have real-time traffic flows with lower bit rates, the approach proposed in this study and PF, BET, and EDF have achieved good results and similar gains for fairness of rate satisfaction and fairness of delay satisfaction as shown in Figures 14–17.

(ii) However, as video and video conferences have real-time traffic flows with larger bit rates, there are obvious performance differences in the fairness of rate satisfaction and fairness of delay satisfaction, as shown in Figures 18–21. When the MRDUF/FCFS and MRDUF/MT schemes are operated under heavy load, their fairness of satisfaction is the optimum among all scheduling algorithms in the simulation environment.

Compared to MT, the PF and BET have the equal fairness gain from light to heavy load. The EDF could have better fairness gain in heavy load compared to the PF and BET schemes. However, our MRDUF/FCFS and MRDUF/MT schemes have a higher than 10% gain than the EDF scheme. So our proposed approach outperforms the contrasts, especially in heavy load.

4.2.2. Throughput. In Figure 22, the MT scheme has the best performance due to the maximum throughput specification. However, the PF and BET should consider the fairness of all UE traffic flows, so the rate sacrifices cannot be avoided. The PF is a little bit better than BET for total throughput due to the UE's channel quality consideration. Meanwhile, the EDF scheme only focuses on delay requirements, so the rate performance is the worst. However, our proposed MRDUF/FCFS and MRDUF/MT concern not only rate fairness but also delay fairness; the throughput might be a little bit lower than MT, and the performance is similar to PF and BET schemes and higher than the EDF scheme.

4.2.3. Rate and Delay Satisfaction. In Figures 23 and 24, the rate and satisfaction both decrease with increasing traffic load. The PF has the best rate of satisfaction due to only focusing on per traffic rate requirement. The lower bit rate traffic flows (e.g., VoIP and online radio) might have a 100% satisfaction value with the average calculation benefit effect. Our proposed approach considers both UE traffic rate and delay requirement and only has the highest satisfaction in very light load; however, our proposed approach has better fairness gains than other contrasts. A little bit of sacrifice cannot be avoided in the rate and delay satisfaction aspect.

4.2.4. Comparison of Average of Satisfaction and Fairness of Satisfaction with Number of UE. Our proposed MRDUF/FCFS and MRDUF/MT schemes focus on UE satisfaction and fairness of satisfaction. In Tables 4–7, the performance of the algorithms proposed in this study is compared with the four algorithms in the control group by a different number of UE 24, 48, and 96 to indicate light, medium, and heavy traffic loads, with parameters as follows: fairness of rate satisfaction, fairness of delay satisfaction, average rate satisfaction, and average delay satisfaction. The values in each cell in the table indicate the gains of MRDUF/FCFS and MRDUF/MT schemes and the control scheduling algorithm, respectively. For example, in Table 4, when the number of UE is 96, the fairness of rate satisfaction of MRDUF/FCFS and MRDUF/MT schemes is higher than that of the MT algorithm by 49% and 42%, respectively, while the average rate satisfaction of MRDUF/FCFS and MRDUF/MT schemes is lower than that of the MT algorithm by 9% and 9%, respectively. Thus, our proposed MRDUF/FCFS and MRDUF/MT schemes are optimum in the fairness of satisfaction while optimal for rate satisfaction and delay satisfaction only under light load. Therefore, for MRDUF/FCFS and MRDUF/MT schemes, the number of UE may be limited in eNB by admission control in the future so that satisfaction and fairness of satisfaction can be taken into consideration at the same time.

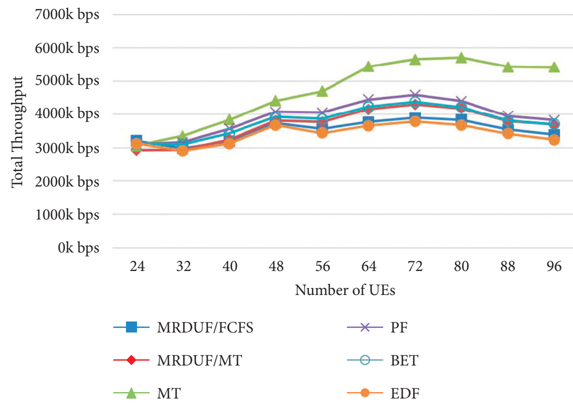

Figure 22: Total throughput.

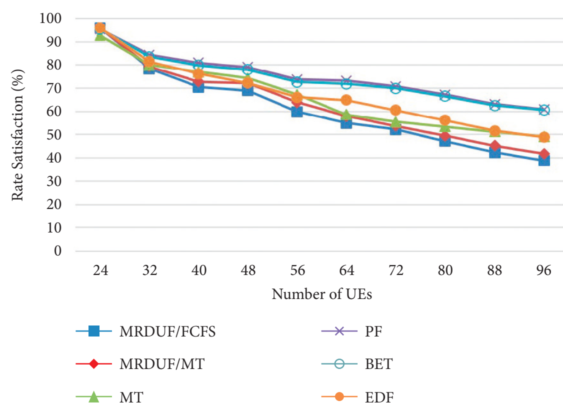

Figure 23: Average rate satisfaction.

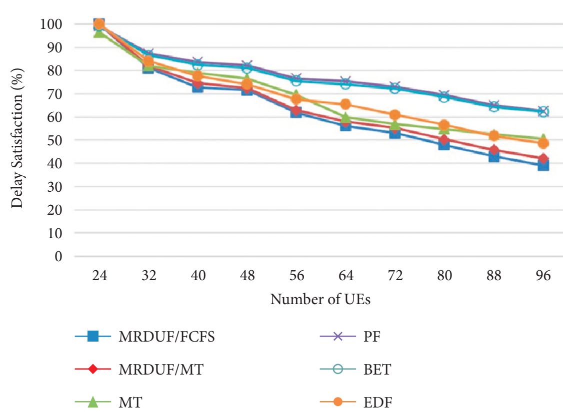

Figure 24: Average delay satisfaction.

Table 4: Proposed approach vs MT with different #UE.

	MRDUF/FCFS and MRDUF/MT vs MT		
	#UE = 24	#UE = 48	#UE = 96
Fairness of rate satisfaction	44%/44%	33%/33%	49%/42%
Fairness of delay satisfaction	72%/71%	31%/29%	50%/44%
Average rate satisfaction	3%/3%	−5%/−4%	−9%/−9%
Average delay satisfaction	4%/4%	−5%/−4%	−10%/−10%

Table 5: Proposed approach vs PF with different #UE.

	MRDUF/FCFS and MRDUF/MT vs PF		
	#UE = 24	#UE = 48	#UE = 96
Fairness of rate satisfaction	2%/2%	8%/8%	30%/23%
Fairness of delay satisfaction	8%/7%	10%/8%	31%/25%
Average rate satisfaction	1%/1%	−10%/−9%	−20%/−19%
Average delay satisfaction	1%/1%	−10%/−10%	−22%/−21%

Table 6: Proposed approach vs BET with different #UE.

	MRDUF/FCFS and MRDUF/MT vs BET		
	#UE = 24	#UE = 48	#UE = 96
Fairness of rate satisfaction	1%/1%	6%/6%	30%/23%
Fairness of delay satisfaction	2%/1%	8%/6%	31%/25%
Average rate satisfaction	1%/1%	−8%/−7%	−19%/−18%
Average delay satisfaction	1%/1%	−9%/−9%	−21%/−20%

Table 7: Proposed approach vs EDF with different #UE.

	MRDUF/FCFS and MRDUF/MT vs EDF		
	#UE = 24	#UE = 48	#UE = 96
Fairness of rate satisfaction	1%/1%	5%/5%	17%/10%
Fairness of delay satisfaction	2%/1%	2%/1%	11%/5%
Average rate satisfaction	0%/0%	−3%/−2%	−9%/−9%
Average delay satisfaction	0%/0%	−2%/−2%	−9%/−9%

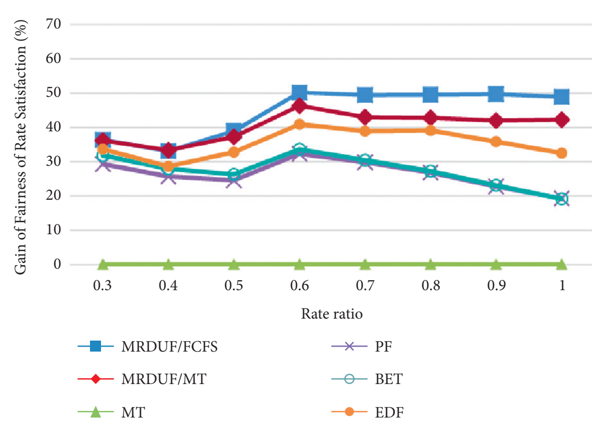

Figure 25: Gain of fairness of rate satisfaction on case A.

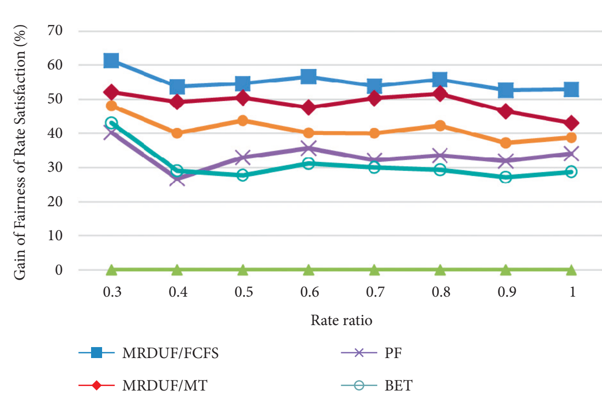

Figure 26: Gain of fairness of rate satisfaction on case B.

Table 8: The proposed approach in different traffic distribution.

	MRDUF/FCFS			MRDUF/MT		
	Case A 1 : 1 : 1 : 1 (%)	Case B 4 : 1 : 4 : 1 (%)	Gap (%)	Case A 1 : 1 : 1 : 1 (%)	Case B 4 : 1 : 4 : 1 (%)	Gap (%)
Fairness of rate satisfaction	49	53	4	42	43	1
Fairness of delay satisfaction	50	54	4	43	45	2
Average rate satisfaction	39	41	2	42	44	2
Average delay satisfaction	39	41	2	42	43	1

4.2.5. Discussion of Average of Satisfaction and Fairness of Satisfaction with Different Distribution of Traffic Flows. To present the traffic impact of these scheduling schemes, case A and case B were designed for further simulation. Case A is the original distribution. That is, the proportion of the four types (VoIP: Video: Online Radio: Video Conference) is 1 : 1 : 1 : 1. Case B is with a higher ratio of voice or audio traffic; therefore, the proportion of the four types (VoIP: Video: Online Radio: Video Conference) is 4 : 1 : 4 : 1. The two cases simulation traffic loads are generated at the same state.

The *X*-axis value has been normalized from light load to heavy load in Figures 25 and 26. Case B has a higher number of lower bit rate traffic flow than case A due to the different traffic distributions. All the scheduling schemes have a higher gain of fairness and average satisfaction in case B. In particular, our proposed MRDUF/FCFS and MRDUF/MT schemes have better performance at light and medium loads (the *rate ratio* from *0.3–0.5*) in case B. Authors have also tried the other simulation criteria, and the results are similar. With the heavy traffic load simulation state, i.e., the *rate ratio* is *1*, case B still has a better performance result in all criteria than case A (Table 8).

5. Conclusions

For current advanced mobile wireless network providers, LTE-A has attracted attention worldwide. To find high network quality satisfaction of MAC scheduling in the LTE-A network is a very hot and important research issue. Some traditional scheduling schemes have been discussed about this idea in many research articles. The common MT, PF, and BET algorithms only consider the requirements for rate, while the EDF algorithm only considers the requirements for the delay. A network user satisfaction-based scheduling approach should consider UE's traffic both rate and delay requirements and regard both the aspects' fairness. Our proposed MRDUF/FCFS and MRDUF/MT schemes can achieve this goal. Moreover, our proposed schemes can support different real-time UE's traffic flows' requirements and achieve higher fairness. Simulation results have shown that the fairness gain in rate and delay has the best performance even though the satisfaction and throughput have less performance. The fairness gain would have a maximum of 50% gain better than contrasts. It is a very important benefit for QoS support.

Data Availability

Data sharing is not applicable to this article as no datasets were generated or analyzed during the current study.

Consent

Not applicable.

Conflicts of Interest

The authors declare that there are no potential conflicts of interest in this paper.

Authors' Contributions

Yi-Ting Mai conceptualized this study, data collection and analysis, literature review, manuscript writing, and experiment design. Cheng-Tao Yang performed the experiments and literature review. All authors read and approved the final manuscript.

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