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Research Article

Adapted Packet Scheduling Algorithm for Robust Real-time Multimedia

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Abstract: This study aims to provide a good real-time multimedia performance under the presence of mobile cellular channel impairments by proposing a novel packet scheduling algorithm that adapts the well-known single-carrier Maximum Largest Weighted Delay First (M-LWDF) algorithm into the multi-carrier downlink Long-Term Evolution-Advanced (LTE-A). The proposed algorithm prioritizes packets that require retransmission as compared to new packets. Packet scheduling of new packets is performed per mobile cellular channel basis. Simulation results demonstrate the efficacy of the proposed algorithm where it is capable in providing good real-time multimedia experience for more users and is more robust towards the impact of mobile cellular channel impairments as compared to a benchmark algorithm.

Keywords: Channel quality information, hybrid automatic repeat request, long term evolution-advanced, mobile cellular channel impairments, orthogonal frequency division multiple access, quality of service

INTRODUCTION

Long Term Evolution-Advanced (LTE-A) is an emerging mobile cellular technology standardized by the Third Generation Partnership Project (3GPP) organization in its attempt to meet the International Mobile Telecommunications Advanced (IMT-Advanced) requirements. The LTE-A is expected to support 100 Mbps peak data rates for highly mobile users and 1 Gbps peak data rates for low mobility users. A high peak data rate in LTE-A is rationalized via carrier aggregation. Carrier aggregation is a method that aggregates two or more Component Carriers (CCs) of the same or different frequency spectrums. LTE-A has a simplified architecture that contains only enhanced Node B (eNodeB) at the Evolved Universal Terrestrial Radio Access Network (E-UTRAN). The eNodeB connects users to the core network and performs all Radio Resource Management (RRM) functions.

The LTE-A uses Orthogonal Frequency Division Multiple Access (OFDMA) and Single-Carrier Frequency Division Multiple Access (S-CFDMA) for downlink and uplink transmissions, respectively. The S-CFDMA will not be discussed any further as this study focuses on the downlink. The minimum transmission unit in the downlink LTE-A that is being used for (re)transmission of packets to users in each CC are known as Resource Blocks (RBs). An RB has 12 subcarriers each of 15 kHz bandwidthin the frequency domain. In the time domain, an RB extends to 1 ms duration.

The advancement of mobile cellular technologies has accelerated demand for real-time and non real-time multimedia applications. These applications have diverse QoS requirements where real-time packets are more sensitive to delays whereas the non real-time packets are more sensitive to packet loss. It should be noted that packets are discarded and considered as lost packets if the packets do not arrive at the user's end within the end-to-end delay threshold (Poudyal *et al.*, 2011). The LTE-A delivers real-time and non real-time multimedia packets using packet-switching technology. As such, packet scheduling, which is another RRM function, is of paramount importance in this system. However, providing Quality of Service (QoS) of multimedia applications comparable to fixed line is challenging because of the nature of mobile cellular channels that are subject to various impairments including imperfect Channel Quality Information (CQI) report and adequate CQI reporting rate as well as expensive mobile cellular channels.

Packet scheduling has been an interesting area of research and abundant research studies attempted to optimize throughput of mobile cellular technologies without degrading the QoS of multimedia performance (Andrews *et al.*, 2001; Cheng *et al.*, 2013; Cho *et al.*,

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2014; Jalali *et al*., 2000; Liu *et al*., 2010; Miao *et al*., 2014; Mnif *et al*., 2014; Pokhariyal *et al*., 2007a, 2007b; Ramli, 2014; Ramli *et al*., 2011; Sandrasegaran *et al*., 2010; Torabzadeh and Ajib, 2010; Tsybakov, 2002; Yafeng and Hongwen, 2003; Yuanye *et al*., 2010). However majority of these studies mostly focus on either delay-tolerant non real-time multimedia applications (Cho *et al*., 2014; Liu *et al*., 2010; Pokhariyal *et al*., 2007b; Tiwana *et al*., 2014; Yafeng and Hongwen, 2003; Yuanye *et al*., 2010) or the legacy single-carrier mobile cellular technologies (Andrews *et al*., 2001; Jalali *et al*., 2000; Tsybakov, 2002) and made a common assumption that the mobile cellular channels are free from any impairments (Andrews *et al*., 2001; Ramli, 2014; Ramli *et al*., 2011; Rindzevicius *et al*., 2008; Sandrasegaran *et al*., 2010). Note that the legacy single-carrier mobile cellular technology contains only one CC and transmission of packets to a user in each scheduling interval utilizes all of the available bandwidth which is in contrast to multi-carrier downlink LTE-A where packet scheduling is performed in each CC and accounts both time and frequency domains.

Since delay-sensitive real-time multimedia applications are getting increasing demand among mobile cellular users and a large portion of mobile cellular resources has to be allocated to support real-time multimedia (re) transmission, this study proposes a novel packet scheduling algorithm that adapts a well-known single-carrier packet scheduling algorithm into multi-carrier downlink LTE-A taking mobile cellular channels impairments into consideration. It should be noted that transmitted packets may be received in error at a user end due to various impairments and correctly received packets in sub-sequent intervals cannot be delivered to application layer until the erroneously received packets are retransmit and decode correctly at the user's end. Therefore, packets that require retransmission (retransmitting packets) need to be given a higher priority as compared to packets of first transmission (new packets) to avoid these packets from being discarded for delay violation and hence minimizing the Packet Loss Ratio (PLR).

The remainder of this study is organized as follows: Section materials and methods provides a detailed description of the adapted well-known single-carrier packet scheduling algorithm as well as discussed the environments of simulation. Results obtained via simulation are discussed in detailed in section results and discussion and finally, conclusion section concludes the paper.

MATERIALS AND METHODS

Adapted well-known single-carrier packet scheduling algorithms: This study adapts the well-known single-carrier Maximum-Largest Weighted Delay First (M-LWDF) (Andrews *et al*., 2001) algorithm into the downlink LTE-A. The aforementioned algorithm shows an excellent performance for providing good real-time multimedia experience in the legacy single-carrier mobile cellular technologies. In each scheduling interval, the M-LWDF selects new packets of a user according to (1) and allocates all of the available bandwidth for transmission of new packets to the selected user:

$$\mu_i(t) = a_i * W_i(t) * \frac{r_i(t)}{R_i(t)} \tag{1}$$

$$a_i = \frac{(\log \delta_i)}{T_i} \tag{2}$$

$$R_i(t) = \left(1 - \frac{1}{t_c}\right) R_i(t-1) + I_i(t) * \frac{1}{t_c} * r_i(t) \tag{3}$$

where, $\mu_i(t)$ is the priority of user i at scheduling interval t, a_i is the QoS requirement of user i, $W_i(t)$ is the delay of the Head-of-Line (HOL) packet of user i at scheduling interval t, $r_i(t)$ is the instantaneous data rate (across the whole bandwidth) of user i at scheduling interval t, $R_i(t)$ is the average throughput of user i at scheduling interval t, $\delta_i$is the application-dependent PLR threshold of user i, T_i is the application-dependent buffer delay threshold of user i, $I_i(t)$ is the indicator function of the event that packets of user $i$are selected for transmission at scheduling interval t and t_c is a time constant. Note that the HOL packet of a user is the packet that has resided the longest in its buffer at the base station while the buffer delay threshold is defined as the maximum allowable waiting time of a packet at the base station buffer.

The present state of the M-LWDF algorithm may not be suited for implementation in multi-carrier downlink LTE-A because it does not account for the situation where multiple CCs are available and scheduling is performed in both time and frequency domains. Additionally, the indicated algorithm assumed that all transmitted packets are correctly received at users' end and hence retransmitting packets are not available. In our attempt to address the stated situation, the M-LWDF is adapted such that packet scheduling is performed in more than one CCs and accounts both time and frequency domains. Moreover, the adapted M-LWDF considers the situation where the downlink LTE-A contains both new and retransmitting packets. It should be noted that the adapted M-LWDF is developed under the constraint that retransmitting packets of a user have to use the same number of RBs and similar Modulation and Coding Scheme (MCS) as in the first transmission. MCS is determined based on the CQI reported by the user. Additionally, it is assumed in this

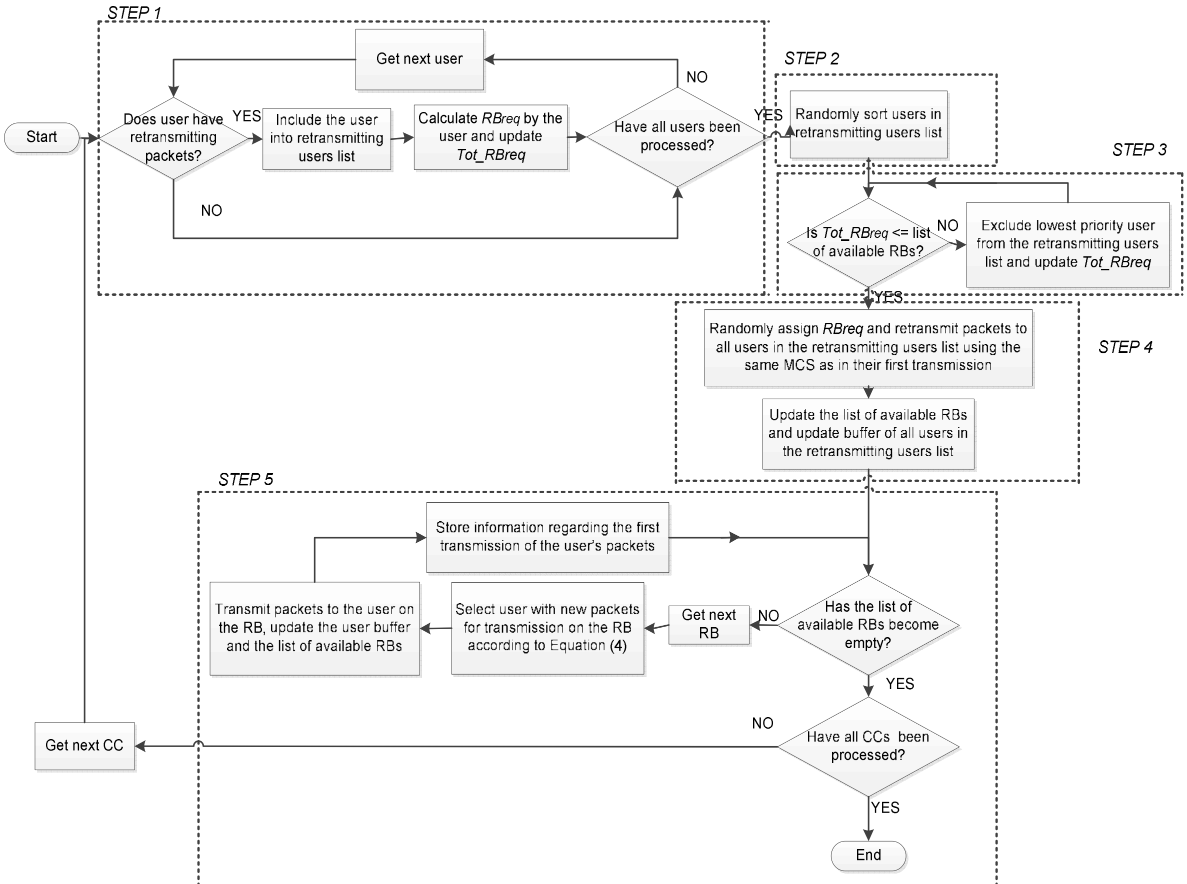

Fig. 1: Flowchart of the adapted-M algorithm

study that a user can either receive new packets or retransmission packets (not both) in each scheduling interval. Detailed description of this adapted M-LWDF algorithm, referred to as Adapted-M, is given next.

The Adapted-M proposed in this study is divided into five steps (Fig. 1). Step 1 until Step 4 of the Adapted-M is responsible to schedule retransmitting packets whereas Step 5 is only executed if there are remaining RBs after completion of Step 4. When compared with new packets, the Adapted-M prioritizes retransmitting packets which are more urgent. This is because it is highly likely that retransmitting packets will be discarded for delay violation. In each scheduling interval and on each CC, Step 1 begins by determining the type of each user packets that needs to be (re)transmit. If a user has retransmitting packets, then Step 1 includes the user into retransmitting user list, computes its required number of RBs (RB_{req}) which is based on first transmission (see Step 5) and update total number of RBs required for retransmission (Tot_RB_{req}) in the CC. Step 1 is repeated until all retransmitting users have been determined. Thereafter, all users in the retransmitting user list are sorted according to a random priority in descending order in Step 2. A user is randomly sorted in this algorithm as it is not necessary to prioritize a user with good channel quality for packets retransmission given that retransmitting packets are more likely to be correctly decoded at user's end compared to packets of the first transmission. This is due to the combining gain of multiple retransmissions.

Step 3 checks if Tot_RB_{req} is more than the maximum available number of RBs in the list of available RBs. If yes, it indicates that RBs in the CC are insufficient for packets retransmission. Therefore a user at the lowest priority is excluded from packets retransmission and the list of retransmitting users and Tot_RB_{req} are updated. Step 3 repeats until Tot_RB_{req}<= the maximum available number of RBs in the list of available RBs. Step 4 is responsible to randomly assign the required number of RBs and retransmit packets to all users in the list using the same MCS as in the first transmission. It was shown in Ramli *et al*. (2013) that random assignment of RBs for packets retransmission led to throughput improvement. The RBs that have been used for packets retransmission are removed from the list of available RBs and the buffers of all users in the list of retransmitting user are updated after completion of Step 4.

Table 1: Simulation parameters

Parameters	Values
Type of CA	Inter-band Non-Contiguous CA
Number of CCs	2
Bandwidth	3 MHz each
Carrier Frequencies	2 GHz and 900 MHz
Number of RBs	15 RBs each
eNB Transmit Power	43.01 dBm
Radio Propagation	Path loss: Hata model for an urban environment Shadow fading: a Gaussian log-normal distribution Multi-path fading: Frequency-flat Rayleigh fading
Channel Quality Reporting	Error-free
HARQ Type	Type II HARQ with Chase Combining
HARQ Feedback	Error-free with 4 ms delay
Maximum Number of Retransmission	3

Step 5 is executed if remaining RBs are available after completion of Step 4. This step is responsible to select new packets for transmission where users who are not in the retransmitting user list are considered. At each scheduling interval, on each CC and on each RB, Step 5 chooses new packets for transmission to a user that maximizes (4). Thereafter, the packets of the selected user are transmitted on the RB and the user buffer is updated. Additionally, Step 5 updates the list of available RBs. Step 5 of the Adapted-M algorithm completes after all information regarding first transmission of the new packets are stored. This information is needed in sub-sequent scheduling intervals if the packets are erroneously received and require retransmission. Step 5 is repeated until the list available RBs become empty:

$$\mu_{i,j,k}(t) = a_i * W_i(t) * \frac{r_{i,j,k}(t)}{\sum_{j=1}^{j=CC_{\max}} R_{i,j}(t)} \tag{4}$$

$$R_{i,j}(t) = \left(1 - \frac{1}{t_c}\right) R_{i,j}(t-1) + I_{i,j}(t) * \frac{1}{t_c} * rtot_{i,j}(t) \tag{5}$$

where, $\mu_{i,j,k}(t)$ is the priority of user i on CC j on RB k at scheduling interval t, $a_i$is the QoS requirement of user i (as defined in (2)), $W_i(t)$ is the delay of the HOL packet of user i at scheduling interval t, $r_{i,j,k}(t)$ is the instantaneous data rate of user i on CC j on RB k at scheduling interval t, $R_{i,j}(t)$ is the average throughput of user i on CC j at scheduling interval t, $rtot_{i,j}(t)$ is the total data rate of user i on CC j at scheduling interval t, $I_{i,j}(t)$ is the indicator function of the event that packets of user i are selected for transmission on CC j at scheduling interval t, CC_{max} is the maximum available number of CCs and t_c is a time constant.

Simulation environment: This performance evaluation uses PLR and mean user throughput metrics. Each metric is described as following:

$$PLR = \frac{\sum_{i=1}^{N}\sum_{t=1}^{T} pdiscard_i(t)}{\sum_{i=1}^{N}\sum_{t=1}^{T} psize_i(t)} \tag{6}$$

$$mean\ \ user\ \ throughput = \frac{1}{N}\frac{1}{T}\sum_{i=1}^{N}\sum_{t=1}^{T} prx_i(t) \tag{7}$$

where, $pdiscard_i(t)$ is the total size of discarded packets (in bits) of user i at time t, $psize_i(t)$ is the total size of all packets (in bits) arrive into the eNB buffer of user i at time t, $prx_i(t)$ is the total size of correctly-received packets (in bits) of user i at time t, N isthe total number of users and T is the total simulation time.

The performance evaluation was conducted within a micro-cell of 250m radius. It is assumed that all users move at 60 km/h and run video application with average data rates of 512 kbps. It should be noted that video streaming is one of real-time multimedia application. The buffer delay threshold is *set* to 100 ms 3GPP acceptable threshold. The PLR threshold is capped at 10^{-3} threshold (3GPP, 2009). This PLR threshold is considered as the maximum threshold where the QoS requirement of the video streaming is satisfied (i.e., video users to experience good streaming experience). Moreover, to allow a user to run 2 min video streaming session without its buffer running dry (if the size of de-jitter buffer is assumed to be 10 s when a user starts its session), the minimum user throughput is assumed to be maintained above 469 kbps. Other parameters used in this performance evaluation are summarized in Table 1 (Ramli and Sandrasegaran, 2013).

RESULTS AND DISCUSSION

The performance of the Adapted-M algorithm is evaluated and compared with another adapted M-LWDF algorithm (Ramli and Riezman, 2015) (referred to as Benchmark algorithm hereafter). This algorithm was considered as the well-known M-LWDF algorithm developed in Andrews *et al*. (2001) does not account the situation where (i) multiple CCs are available and scheduling is performed in both time and frequency domains and (ii) retransmitting packets are not available. The Benchmark algorithm contains four steps. When compared with the Adapted-M algorithm that prioritizes retransmitting users, the Benchmark algorithm gives equal opportunity to users with new and retransmission packets. In each scheduling interval and on each CC, Step 1 of the Benchmark algorithm

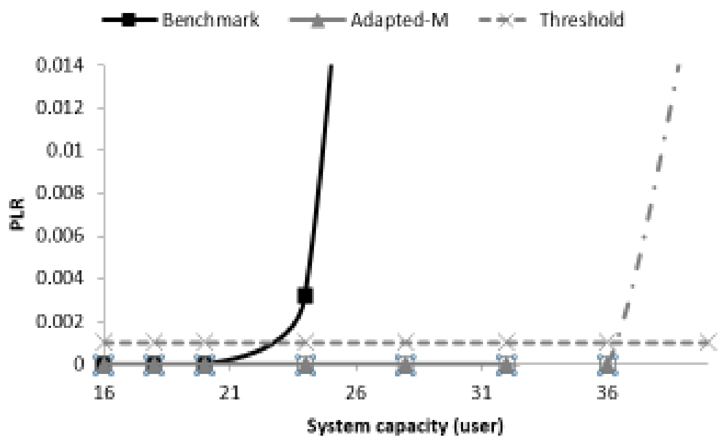

Fig. 2: PLR vs. system capacity

Table 2: Maximum system capacities to satisfy good real-time multimedia experience

Packet scheduling algorithm	Maximum system capacities	Percentage of improvement in Adapted-M over Benchmark (%)
Benchmark	22	63.6
Adapted-M	36	

begins by determining the type of each user packets that needs to be (re)transmit. If a user has retransmitting packets; then, this step computes the number of RBs required (*RBreq*) by the user. The user will only be included into active user list if its *RBreq* is less than the list of available RBs in the CC.

A user with new packets to be transmitted will automatically be included into active user list in Step 1. Thereafter, Step 2 selects a user (from the active user list) that maximizes (8):

$$\mu_{i,j}(t) = a_i * W_i(t) * \frac{avg\_r_{i,j}(t)}{\sum_{j=1}^{j=CC_{\max}} R_{i,j}(t)} \quad (8)$$

$$avg\_r_{i,j}(t) = \frac{1}{RB_{\max}} * \sum_{k=1}^{k=RB_{\max}} r_{i,j,k}(t) \quad (9)$$

where, $\mu_{i,j}(t)$ is the priority of user i on CC j at scheduling interval t, $a_i$is the QoS requirement of user i (as defined in (2)), $W_i(t)$ is the delay of the HOL packet of user i at scheduling interval t, $avg\_r_{i,j}(t)$ is the average data rate of user i on CC j at scheduling interval t, $R_{i,j}(t)$ is the average throughput of user i on CC j at scheduling interval t, $r_{i,j,k}(t)$ is the instantaneous data rate of user i on CC j on RB k at scheduling interval t, CC_{max} is the maximum available number of CCs and RB_{max} is the maximum available number of RBs.

If a user with retransmitting packets is selected, Step 3 will randomly assign RB_{req} and retransmits the packets to the user using the same MCS as in the first transmission. The list of available RBs in the CC and the user buffer are then updated after completion of this step. Step 4 will only be executed if a user with new packets is selected in Step 2. In this step, an RB with the best channel quality will be selected and new packets will be transmitted to the user on the selected RB. Subsequently, the user buffer at the eNodeB is updated and the selected RB is removed from the list of available RBs. Step 4 is repeated until the user does not have any new packets in its buffer or the list of available RBs becomes empty. The information regarding the first transmission of the packets are stored at the end of Step 4. If the list of available RBs in the CC and the active user list are not empty after completion of Step 3 or Step 4 then Step 2 will again be repeated.

The PLR performances of the Adapted-M and Benchmark algorithms with increasing system capacity are illustrated in Fig. 2. The CQI delay is set at 1 ms duration in this performance evaluation. With increasing system capacity, more packets will be discarded for delay violations as there are insufficient RBs available in each CC to schedule all the packets. This leads to PLR degradation with increasing system capacity. If a good real-time multimedia experience is to be satisfied, then it can be observed in Table 2 that the Adapted-M has approximately 63.6% system capacity improvement over the Benchmark algorithm.

Similarly, if the minimum mean user throughput requirement is to be satisfied at 469 kbps, then it is illustrated in Fig. 3 and Table 3 that the Adapted-M is capable to support more than 36 users as compared to the Benchmark algorithm that can only support up to 28 users.

Figure 4 and 5 show the PLR and mean user throughput performances of both algorithms with increasing CQI delay. These performance evaluations were conducted to show the robustness of the proposed algorithm in minimizing the detrimental effect due to mobile cellular channel impairments, generally and due

Table 3: Maximum system capacities to satisfy mean user throughput at 469 kbps

Packet scheduling algorithm	Maximum system capacities
Benchmark	>36
Adapted-M	28

Table 4: Maximum tolerable CQI delays to satisfy good real-time multimedia experience

Packet scheduling algorithm	Maximum CQI delay	Percentage of improvement in Adapted-M over Benchmark (%)
Benchmark	17	52.9
Adapted-M	26	

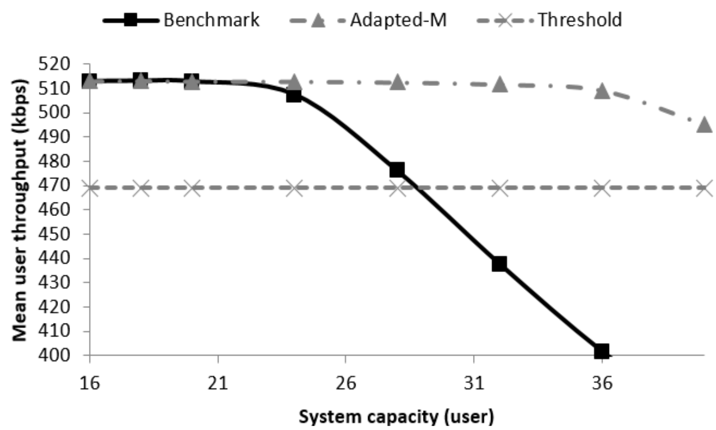

Fig. 3: Mean user throughput vs. system capacity

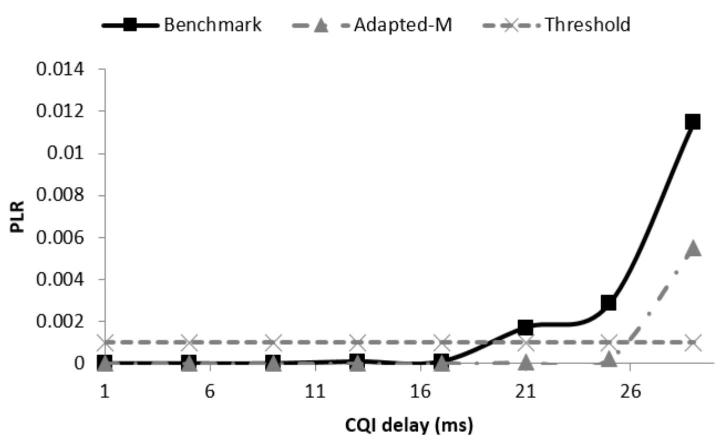

Fig. 4: PLR vs CQI delay

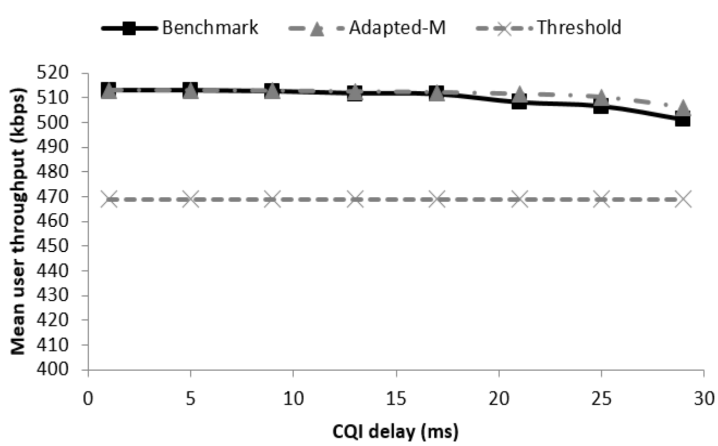

Fig. 5: Mean user throughput vs. CQI delay

to outdated CQI report, specifically. It can be observed in Fig. 4 that the PLR degrades with increasing CQI delay. This is because the CQI that is being used to determine the MCS for the packets before they were (re) transmitted are not up-to-date as the mobile cellular channels vary rapidly due to user speed which is at 60 km/h in this performance evaluation. The packets that arrive in error at the users' end may need to be retransmitted. However, given the strict delay of the real-time multimedia, majority of the packets are discarded for delay violations and some due to exceeding maximum number of retransmissions.

It is demonstrated in Table 4 that the Adapted-M algorithm is more robust as compared to the Benchmark algorithm where if a good real-time multimedia experience is to be satisfied at 10-3, then the maximum tolerable CQI delay in Adapted-M can be as high as 26 ms but limited to 17 ms in the Benchmark algorithm. Additionally, though a slight degradation in terms of mean user throughput is achieved in Benchmark over the Adapted-M, the algorithm cannot provide a good real-time multimedia experience when CQI delay is above 17 ms.

The significant achievements of the Adapted-M over Benchmark algorithm can be explained as follows. (i) The Adapted-M chooses packets of a user for transmission per RB basis. After the packets have been transmitted to the selected user, its buffer status is updated. This allows the algorithm to always schedule the most urgent packets on each RB. Additionally, given that decision is made per RB basis, this allows the Adapted-M algorithm to exploit multi-user diversity on each RB and on each CC. On the other hand, the Benchmark algorithm selects a user first before assigning RBs to the selected user for packets transmission. There is highly likely some packets transmitted to the selected user are not urgent packets, given that buffer status is not updated in each RB. This leads to these packets being discarded in subsequent scheduling intervals for delay violations and hence degrading the Benchmarks performances, (ii) since the Adapted-M prioritizes retransmission as compared to new packets, this allows the algorithm to further optimize the real-time multimedia performance as majority of retransmitting packets have resided longer at the eNodeB highly likely to be discarded for delay violation.

CONCLUSION

To minimize detrimental effects due to mobile cellular channel impairments on the QoS of real-time multimedia applications, this study proposes a novel packet scheduling algorithm called Adapted-M. This algorithm adapts the well-known single-carrier M-LWDF so as to allow the M-LWDF algorithm to perform packet scheduling in the multi-carrier downlink LTE-A and taking the impacts of mobile cellular channel impairments into consideration. When compared to new packets, this algorithm prioritizes packets that require retransmission. Simulation results demonstrate that the Adapted-M is efficient in providing good real-time multimedia experience for 63.6 % more users and can tolerate up to 52.9% outdated CQI (CQI delay) over the Benchmark algorithm. Future studies will investigate the impacts of other mobile cellular channel impairments on the proposed algorithm when supporting real-time and non real-time multimedia applications.

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