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A Packet Scheduling Scheme for Improving Real-time Applications Performance in Downlink LTE–advanced

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ABSTRACT

Quality of Service based packet scheduling is a key-feature of LTE-A mandating selection and transmission of individual user packets based on their priority. HARQ Aware Scheduling, Retransmission Aware Proportional Fair, Chase Combining Based Max C/I Scheduling and Maximum- Largest Weighted First (M-LWDF) are popular Packet Scheduling Algorithms (PSAs) developed to meet QoS requirements. In highly erroneous LTE-A cannel, M-LWDF is considered to be one of best PSA. To validate the performance of M-LWDF for the LTE-A channel, Mean User Throughout, and Fairness performance measures were evaluated for 3 different PSAs designed based on M-LWDF algorithm in this paper. A C++ based simulation results indicate the superiority of the PSA3 algorithm within the threshold of the performance measures against benchmarks. It has shown more efficiency and the performance of RTA traffic was enhanced. Results show that PSA3 is superior to its benchmark PSA2 by 12% in Mean User Throughput and 11% in Fairness. PSA2 performed the worst because it prioritizes new users and it allocated all available RBs to the scheduled user leaving the rest to wait in the buffer. PSA3 maintians good Mean User Throughput and fairnessdue to scheduling each user on its RB which leads to multi-user diversity.

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1. INTRODUCTION

Wireless communication is one of the most active areas of technology development of present time. This development is being driven primarily by the transformation of what has been largely a medium for supporting voice telephony into a medium for supporting other services, such as the transmission of video, images, text, and data. Thus, the demand for new wireless capacity is growing at a very rapid pace from Third Generation (3G) technology to Fourth Generation (4G). However, the more developed verison of Long Term Evolution (LTE) Long Term Evolution–Advanced (LTE-A) (which is known as release 10 by the Third Generation Partnership Project (3GPP), it became the leading 4G technology due to some limitations in LTE such as not having a high data rate and its vulnerability to interference and scrambling in the physical layer.

LTE-A aimed to meet the demands of higher data rate within the quality of service (QoS) required by the ITU-R with more coverage as compared to LTE. This is achieved through the usage of improved packet scheduling algorithms, such as the Carrier Aggregation (CA) technique, the enhanced usage of multi-antenna techniques and support for relay nodes [1][2]. In LTE-A system, the core network is divided as figure 1. Enhanced NodeB (eNB) is a combination of NodeB and Radio Network Controller (RNC) which interconnects the User Equipment (UE) and is able to serve more than one cell at a time while home eNB serves a femtocell. Furthermore, Enhanced Packet Core (EPC) consists of serving gateway (S-GW) for routing packets between UE and Packet Data Network (PDN), whilst Mobility Management Entity (MME) copes UE access and mobility, and PDN Gateway (PDN GW) is a gateway to PDN [3], [4].

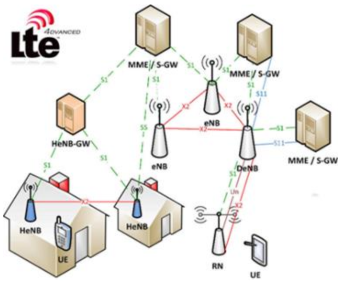


Figure 1. LTE-A Network architecture

In LTE-A, Orthogonal Frequency Division Multiple Access (OFDMA) is used for downlink transmission and Single Carrier Frequency Division Multiple Access (SC-FDMA) is used for uplink transmissions. The OFDMA symbols are grouped into Resource Blocks (RB), the RBs have a total size of 180 kHz in the frequency domain and 1 ms in the time domain. Each user is allocated a number of RB and each Componet Carrier (CC) contains a number of RBs available for usage. However, each RB can merely be assigned to one user only during each 1ms Transmission Time Interval (TTI) [5]. The LTE-A employs a simplified Evolved Universal Terrestrial Radio Access Network (E-UTRAN) architecture that only consists of eNBs' which links users to the core network and performs all Radio Resource Management (RRM) functions. Packet scheduling is one of the key RRM functions and it's uses become vital as the LTE-A delivers both delay-sensitive Guaranteed Bit Rate (GBR) and loss-sensitive Non-GBR multimedia applications using packet switching technology [1]. In a non-real time (NRT) service environment, channel condition is the most common scheduling criterion, but in real-time (RT) service environment, mean user throughput, Packet Loss Ratio (PLR) and fairness are the common scheduling criterion [6]. If the network overloads with packets, the scheduling algorithms plan the order of the packet transmission under various QoS requirements from different users and allocates them to different resources so that it offers larger capacity and higher data rates [7]. The generalized model of packet scheduling is shown in figure 2 while Table1 compares between LTE and LTE—A.

The process of scheduling a downlink LTE system is as follows: it is at 1 ms interval (as known as Transmit Time Interval, TTI) which consists of 2 time slots, or resource-block-pair basis (RB, one sub frame of 0.5ms over 180 kHz). This specific TTI a user is assigned two consecutive RBs [7]. Nevertheless, once these Signal to Interference plus Noise Ratio (SINR) values in each RB are determined, it will be sent to the serving eNodeB in each TTI [9]. These SINR values that are received by each user will help to find out the modulation and coding scheme (MCS) that is appropriate for downlink packet transmission. After which the data rate comes into play as it is the number of bits that a user can support in two consecutive RBs in a TTI. The user's priority in channel-dependent scheduling is also found here and is helped determined by the effective SINR value [10]. The rate also determines the number of bits that a user can have in two consecutive RBs in a TTI.

Moreover, a buffer is assigned to each user and any packet arriving into the buffer will be consequently time stamped and queued for transmission on a First-in-First-out (FIFO) basis. This is all done at eNodeB. Furthermore, the time difference between the present time and the arrival time of a packet is computed in the queue for each packet at the eNodeB buffer. This is known as the Head of Line (HOL) packet delay. LTE uses three modulation schemes: QPSK (4QAM), 16QAM, and 64QAM. A mobile station or eNodeB will choose the selection of modulation and channel coding schemes (MCS) to match the channel conditions. If the channel quality is good, it will use a good quality MCS to transmit at the highest data rate [11]. The Multi-Input Multi-Output (MIMO) basically employs multiple antennas at transmitter (network) and receiver (terminal) side allowing simultaneous transmission of multiple data streams over a single radio link. Either of the two MIMO schemes is chosen: spatial multiplexing or transmit diversity, depending on which one is more suitable in the channel conditions present [12].

Table 1. Comparison between LTE vs. LTE-A [8]

Technology	LTE	LTE—A
Peak data rate Down Link (DL)	150 Mbps	1 Gbps
Peak data rate Up Link (UL)	75 Mbps	500 Mbps
Transmission bandwidth DL	20MHz	100 MHz
Transmission bandwidth UL	20MHz	40 MHz (requirements as defined by ITU)
Mobility	Optimized for low speeds (<15 km/hr) - High Performance at speeds up to120 km/hr - Maintain Links at speeds up to 350 km/hr	Same as that in LTE
Coverage	Full performance up to 5 km	- Same as LTE requirement - should be optimized or deployment in local areas/micro cell environments.
Scalable Band Widths	1.3,3, 5, 10, and 20 MHz	Up to 20-100 MHz
Capacity	200 active users per cell in 5 MHz.	3 times higher than that in LTE
Bandwidth	Symmetric	Asymmetric
MIMO	DL: 2x2, 4x2, 4x4 UL:1x2,1x4	DL: up to 8x8 UL: up to 4x4
Coordinates multipoint	No	Yes
Relaying	No	Yes

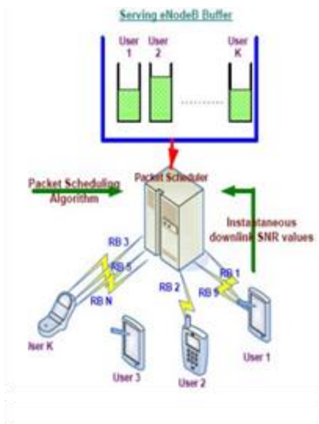


Figure 2. Packet scheduling operation

LTE-A is backward compatible with LTE and also shares it's features too. However, it has some technical improvements over LTE. CA is the clearest way to use in order to speed up the peak data rate to meet the requirements of IMT-Advanced [13] and an upside is that network providers can use all the available spectrums that they were assigned from the government regulator for LTE-A. LTE-A will apply MU-MIMO techniques which are an improvement from the current SU-MIMO. All this greatly improves the peak spectrum efficiency, system data rate, capacity (e.g., overall throughput), as well as cell-edge user throughput [14]. Multi-point transmissions features in LTE-A with a single transmitter with antennas which are geographically separated, boosts LTE-A as the cell-edge user throughput is increased, as well as the coverage is also increased for the deployment flexibility [15].

A HAS algorithm is used in GBR services for downlink LTE system, as it reduces the number of lost packets and keeps a low queuing delay while still holding a maximum effective user throughput [16]. Packets which require retransmission that wait in the buffer long and exceed the buffer delay threshold will be discarded, thus HAS algorithm gives highest priority for users with packets that require retransmission that has been in the buffer the longest. This means those users are prioritized over other user with packets that require retransmission.

RAPF is an algorithm whose purpose is to meet the fairness standards and it uses the following equation to choose the highest priority users with new packets or retransmission:

$$u_i(t) = \frac{r_i(t)}{R_i(t)} \quad (1)$$

For a good mobile channel condition, the value of $r_i(t)$ (instantaneous data rate for user *i* at scheduling interval *t*) will be high so it will result in high priority value. On the other hand, a bad mobile channel condition will result into a low $r_i(t)$ thus low priority value, which means that there will be small chance of transmitting. However, $R_i(t)$ (average throughput of user *i* at scheduling interval t) will be low, this will lead to a high priority value. This means that RAPF algorithm gives users good chance to transmit even though the mobile channel conditions are bad. The packet scheduling algorithm that utilized the channel condition and retransmission information to be used in the scheduling process was investigated in [9].

2. MODEFIED MAXIMUM-LARGEST WEIGHTED DELAY FIRST ALGORITHM

2.1. Maximum-Largest Weighted Delay First Algorithm (M-LWDF)

M-LWDF algorithm is proposed to support ral time applications (RTA). It was developed for a single carrier mobile cellular system that transmits packets to the user by using only one CC and across the whole bandwidth of CC. M-LWDF algorithm is suitable for satisfying the QoS requirements of GBR applications but it does not take packet delay into consideration [5]. In each TTI the scheduler schedules a user that has the highest priority according to equation 2 to receive it's packet in each TTI. The scheduling criteria metric $u_i(t)$ is defined as follows:

$$u_i(t) = \alpha_i * W_i(t) * \left(\frac{r_i(t)}{R_i(t)}\right) \quad (2)$$

$$\alpha_i = -\frac{(log\delta_i)}{T_i} \quad (3)$$

$$R_i(t) = \left(1 - \frac{1}{t_c}\right) * R_i(t-1) + I_i(t) * \frac{1}{t_c} * r_i(t) \quad (4)$$

where $u_i(t)$ is the priority of user *i* at scheduling interval *t*, α_i is the QoS requirement of user *i*, $W_i(t)$ is the delay of HOL packet of user *i* at scheduling interval *t*, $r_i(t)$ is the instantaneous data rate of user *i* at scheduling interval *t*, $R_i(t)$ is the average throughput of user *i* at scheduling interval *t*, δ_i is the application dependent Packet Loss Ratio (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 [5].

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. Since M-LWDF jointly considers HOL packet delay along with PF properties, it obtains a good throughput and fairness performance along with a relatively low PLR. The simulation has verified that M-LWDF algorithm is efficient for usage in downlink in LTE-A for maximizing the system capacity while providing satisfactory QoS of GBR services [17].

2.2. Proposed Modified M-LWDF Algorithms

Maximum- Largest Weighted Delay First algorithm (M-LWDF) is one of the best packet scheduling algorithms for supporting GBR applications, hence its being considered for this research. M-LWDF algorithm was developed for mobile cellular systems that preform packet scheduling in a single CC and allocate all the available bandwidth to a user with highest priority [5]. In each time interval, the priority of each user is computed according to the following equation:

$$u_i(t) = \alpha_i * W_i(t) * (\frac{r_i(t)}{R_i(t)}) \quad (5)$$

Where $u_i(t)$ is the priority of user *i* at scheduling interval *t*, α_i is the QoS requirement of user *i*, $W_i(t)$ is the delay of HOL packet of user *i* at scheduling interval *t*, $r_i(t)$ is the instantaneous data rate of user *i* at scheduling interval *t*, $R_i(t)$ is the average throughput of user *i* at scheduling interval *t* [5].

Due to the characteristic of the mobile cellular systems, some packets that are transmitted to users may be received in error. As such, these packets need to be retransmitted. This research considers the situation in which the LTE-A system consists of users with new packets and packets that require retransmission.

There is more than one CCs considered in this research and packet scheduling is performed in time and frequency domain. The M-LWDF algorithm is implemented in this paper using three different algorithms named PSA1, PSA2 and PSA3. Each algorithm is described next. PSA1 prioritizes users with retransmission of TBs over users with new transmissions, PSA2 prioritizes users with new packets of TBs over users with packets that require retransmission while PSA3 gives equal opportunity to all users.

In each Transmit Time Interval (TTI) and on each CC, PSA1 schedule the retransmission users first then if any resources left it will be directed to the new users. Among all the retransmission users in a cell, it selects the first retransmission users for retransmitting TB that maximize $\mu_{i,j}(t)$ in the following equation

$$\mu_{i,j}(t) = \alpha_i * W_i(t) * \frac{r_{i,j,k}(t)}{\sum_{j=1}^{j=CC_{max}} R_{i,j}(t)} \tag{6}$$

Where $\mu_{i,j}(t)$ is the priority of user *i* on CC j at scheduling interval *t*, α_i is the QoS requirement of user *i*, $W_i(t)$ is the delay of Head-of-Line (HOL) packet of user *i* at *t*, $r_{i,j,k}(t)$ is the data rate of user *i* on CC *j* on RB *k* at *t*, $R_{i,j}(t)$ is the average throughput of user *i* on CC *j* at *t*, CC_{max} is the maximum number of CC available.

This equation shows that PSA1 schedules retransmission users first based on the data rate of each user on each CC. Then it will schedule the user on each RB and this allows multi-user diversity to be exploited for retransmission. The remaining RBs will be assigned to new users. After scheduling retransmission users, PSA1 schedule users with retransmission packets which has the highest priority based on the previous equation. It can be observed that the algorithm determines priority on each user on each RB and it aggregates the average throughput across all CCs for each user.

PSA2 schedules the new users first based on the following equation. New user which has the highest priority will be scheduled first. However, PSA2 allocates all of the available RBs to the selected user. PSA2 does not take data rate of user on each RB into consideration when determining the priority of the new user, it take the average data rate. The following equation shows PSA2 using average data rate of all RB in determining the user priority instead of the data rate in each RB. This algorithm also aggregates the average throughput across all CCs for each user. After completing scheduling the new users then it schedules the retransmission users.

$$\mu_{i,j}(t) = \alpha_i * W_i(t) * \frac{avg - r_{i,j}(t)}{\sum_{j=1}^{j=CC_{max}} R_{i,j}(t)} \tag{7}$$

Where $\mu_{i,j}(t)$ is the priority of user *i* on CC *j* at scheduling interval *t*, α_i is the QoS requirement of user *i*, $W_i(t)$ is the delay of Head-of-Line (HOL) packet of user *i* at *t*, $avg - r_{i,j}(t)$ is the average data rate on user *i* on CC *j* at *t*, $R_{i,j}(t)$ is the average throughput of user *i* on CC *j* at *t*, CC_{max} is the maximum number of CC available.

PSA3 which is the proposed algorithm, is similar to PSA2 as this algorithm selects user with highest priority calculated by the following equations. The difference is that PSA3 selects new packet and packets that need transmission/retransmission by applying two different equations based on the user packets type, or in simple terms the algorithm gives equal opportunity to all users. Users with the highest priority will be selected for transmission of TB. After selecting the user, PSA3 will determine the type of packets, of its retransmission packets Then, required RB will be assigned randomly and the packets will be retransmitted. On the other hand, if the user selected has new packets then the RB which has the best channel quality will be assigned to this packet then transmitted to the user.

$$\mu_{i,j}(t) = \begin{cases} \alpha_i * W_i(t) * \frac{avg - r_{i,j}(t)}{\sum_{j=1}^{j=CC_{max}} R_{i,j}(t)} & i \in \ new\ user \\ \alpha_i * W_i(t)_{retrans} * \frac{avg - r_{i,j}(t)}{\sum_{j=1}^{j=CC_{max}} R_{i,j}(t)} & i \in \ retransmission\ user \end{cases} \tag{8}$$

Where $\mu_{i,j}(t)$ is the priority of user *i* on CC *j* at scheduling interval *t*, α_i is the QoS requirement of user *i*, $W_i(t)$ is the delay of Head-of-Line (HOL) packet of user *i* at *t*, $W_i(t)_{retrans}$ is the delay of Head-of-

Line (HOL) retransmission packet of user i at t, $avg - r_{i,j}(t)$ is the average data rate on user i on CC j at t, $R_{i,j}(t)$ is the average throughput of user i on CC j at t, CC_{max} is the maximum number of CC available [5].

A series of C++ programing code is developed for computer simulation to evaluate the performance of the proposed packet scheduling algorithm PSA3 and the other two PSA1 and PSA2 for validation purposes for a delay sensitive GBR applications in the downlink of LTE-A. All of the parameters are summarized in Table 2.

Table 2. System Parameters [6], [18]

Parameters	Values
Bandwidth	3 MHz
Carrier frequency	2 Ghz
Number of RB	15 RBs
Channel quality	Error-free
Buffer delay	20 ms
Maximum No of users	60
Maximum No of retransmission	3
Users speed	30 Km/h
Cell radius	500 m

Mean user throughput is defined as the amount of data (packets) moved successfully from one place to another (from sender to receiver) in a given time period and its measured in bits per second (bps). The mean user throughput can be represented as:

$$\text{Mean user throughput, } T_{user} = \frac{1}{N}\frac{1}{T}\sum_{i=1}^{N}\sum_{t=1}^{T} Prx_i(t) \tag{9}$$

Where, $Prx_i(t)$the total size of correctly received packets (in bits) of user i at time t, N: the total number of users, and T: the total simulation time.

Fairness is defined as a metric used to determine whether users get their fair share of the system resources. Jain's fairness index rates the fairness of a set of values with n users where the results range from 1/n (worse case) to 1 (best case). There is a lot of different type of fairness index but we are only interested in Jain's fairness index because it used on system throughput [18]. It can be expressed as below:

$$\text{Jain's Fairness, } F = \frac{\left(\sum_{i=1}^{N} X_i\right)^2}{N * \sum_{i=1}^{N} {X_i}^2} \tag{10}$$

Where, X_i: the system throughput of user i, N: the total number of users, and T: the total simulation time. The objective of this proposed algorithm is to maximize the performance while it does not degrade the performance of others users. The mathematical representation for maximum T_{user} subject to user (N) and packets received correctly ($Prx_i\ (t)$), while PLR<=10^{-3} and F>= 0.5 as:

$$T_{user\ max} = \frac{1}{N}\frac{1}{T}\sum_{i=1}^{N}\sum_{t=1}^{T} Prx_i(t)\ , PLR \leq 10^{-3}\ , F \geq 0.5 \tag{11}$$

Where $\mu_{i,j}\ (t)$ is the priority of user i on CC j at scheduling interval t, α_i is the QoS requirement of user i, $W_i(t)$ is the delay of Head-of-Line(HOL) packet of user i at t, $W_i(t)_{retrans}$ is the delay of Head-of-Line(HOL) retransmission packet of user i at t, $avg\text{-}r_{i,j}(t)$ is the average data rate on user i on CC j at t, $R_{i,j}(t)$ is the average throughput of user i on CC j at t, CC_{max} is the maximum number of CCs available [6].

Our proposed scheme is divided into four steps. The first step is to identify whether the user's packets are new or needs retransmission. Once this is done the algorithm checks the possibility of accommodating the retransmission packets based on their RBreq, if there is a possibility the packets are added to an active user's list. Moreover, step two starts by determining the users list of packets, whether they are new or retransmission packets, afterwards the user with the highest priority based on two equations for both new and retransmission packets is selected, based on that selection if the user has retransmission packets step three is selected, if the user has new packets step four is selected. Step three starts with randomly assigning RB to the user then retransmitting it, then user buffer and RB lists are updated. Step four goes somewhat similar to step three, an RB with the best channel quality is selected then assigned to the user, once that is done the packets are transmitted, then similar to step three user buffer and the list of available RB are updated. Step three and four are repeated until the system is out of RB or users to process, as shown in flowchart Figure 7.

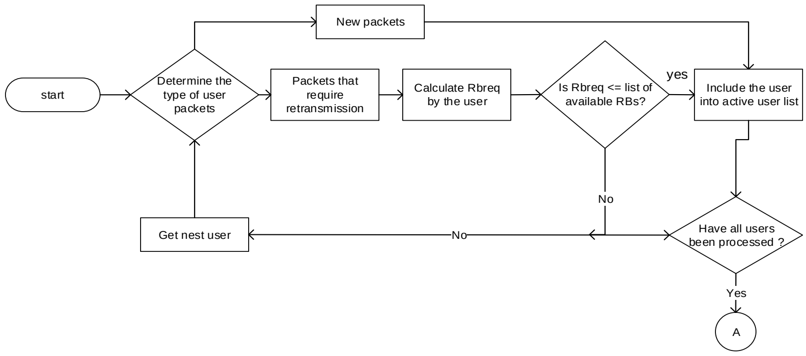
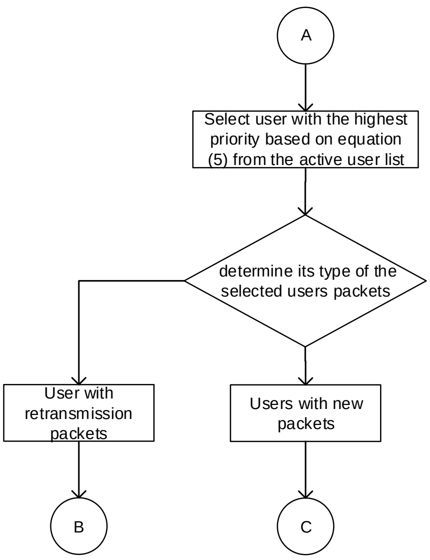
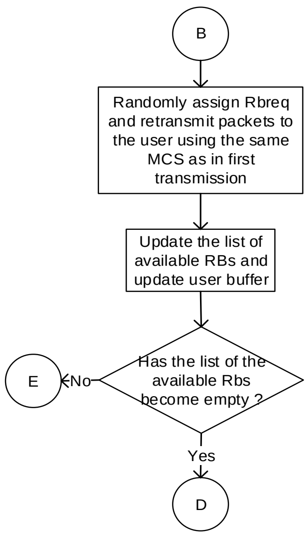


Figure 3. Step 1

Step 1 will be repeated till all users have been processed into the active list of users or discarded the users that the algorithm can not process (RB_{req}>list of available RB).





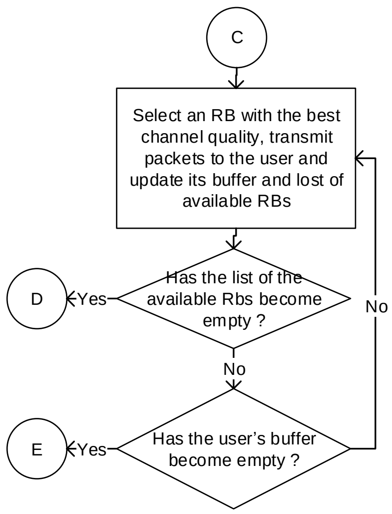


Figure 4. Step 2 Figure 5. Step 3 Figure 6. Step 4

Step 2 is shown in Figure 4. It starts by applying the priority equation (15) to find out which user has the highest priority to be transmitted at this point. Step 3 in Figure 5starts after choosing which user maximize (highest priority) equation (9) then the algorithm will randomly assign RBreq and retransmits the packets to the user using the same MCS as in the first transmission.

After step 3 the list of available RB is checked if there is still some more then remove the user from the active user list the algorithm checks if there are more users if yes then we repeat from the beginning of step 2 if there is no more users then we move on to the final step which is checking if all CC have been processed if not then the algorithm starts again, if all CC have been processed then the algorithm comes to an end. During step 4 the algorithm checks if there is more available RB if yes then proceed to check the user buffer and if its empty then we proceed to remove the user from the active user list and go through the same process as explained before for step 3, but if the RB list is empty in both steps 3 and 4 (see figure 6) then proceed straight to check if all CC have been processed and end the algorithm if they are still empty. So, the final process is like the following.

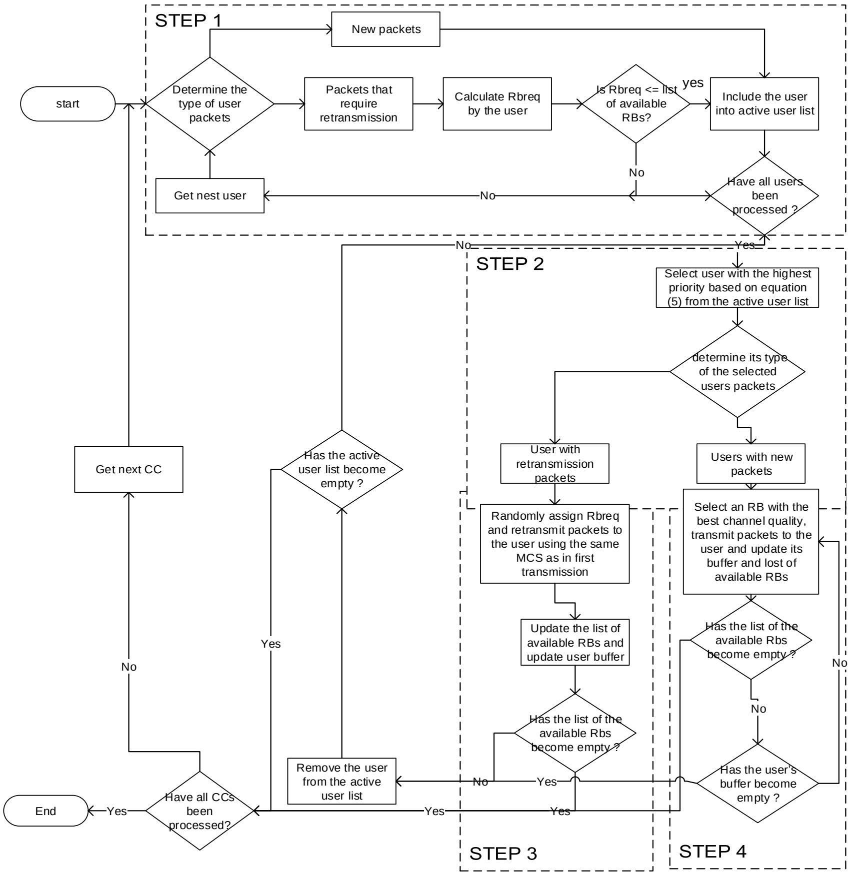


Figure 5. Step 4

3. RESULTS AND ANALYSIS

In this results and analysis section of this paper, the performance metrics are Mean User Throughput, and Fairness for our proposed algorithm.

3.1. Mean User Throughput

Figure 8 presents the values of mean user throughput against the system capacity. Since the average data rate used in this system is 256 kbps then the maximum mean user throughput can't be more than that as it can be observed that the highest value for mean user throughput is 256 kbps at zero users which is the highest system capacity possible.

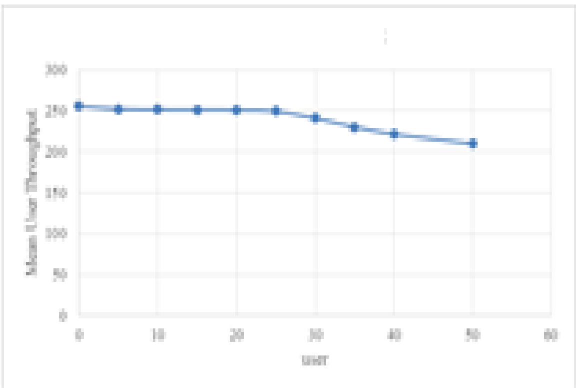


Figure 6. Mean user throughput vs system capacity for PSA 3

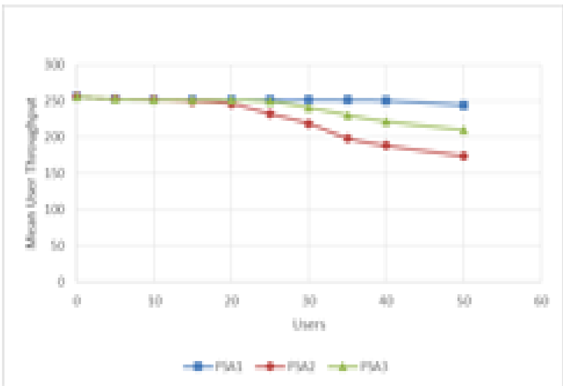


Figure 7. Mean user throughput vs system capacity

Figure 9 shows the mean user throughput against system capacity that is obtained from PSA1, PSA2 and PSA3. As we can see all algorithms start at 256 kbps at zero users and decreased as the number of user increased for both PSA2 and PSA3, while PSA1 performance does not degrade much. This is a disadvantage for new new users as the donot get a chance to be schudled as PSA1 gives priority to retrsmision users. This is clearly reflected in Figure 11 for the fairness.

3.2. Fairness

Figure 10 shows a graph that displays fairness values against system capacity for PSA3. It can observe that this system gives very acceptable fairness with increasing the system capacity up to 50 users. It starts with a fairness value of 1 at zero users then it keeps a steady performance at almost maximum value of 1 till it reaches 25 users. Then it starts to decrease slowly with more users competing for the channel untill it reaches almost 0.95 at 50 users. However, what is interesting in the comparison between PSA2 and PSA3, where PSA3 shows much better achievable mean user throughput compared to PSA2. This is due to the fact that PSA3 schedules users with the highest priority for transmission TB to be successful, hence, leading to a better throughput performance as depicted in figure 8 and figure 9. Figure 11 depicts the fairness of all algorithms against the system capacity. They all start at fairness value of 1 at zero users then they all maintain approximately 1 up till 20 users. As the channel becomes congested, PSA2 performance starts then to decrease rapidly with the increasing number of users. That means PSA1 has the best fairness out of all three algorithms which expected because it has a better throughput.

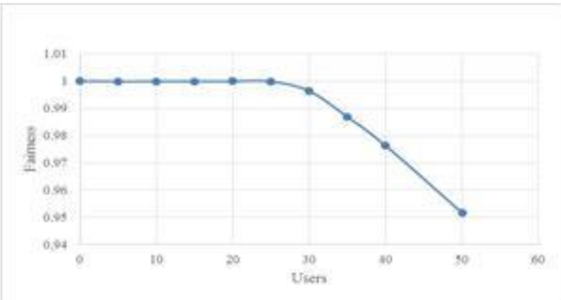


Figure 8. Fairness vs system capacity for PSA3

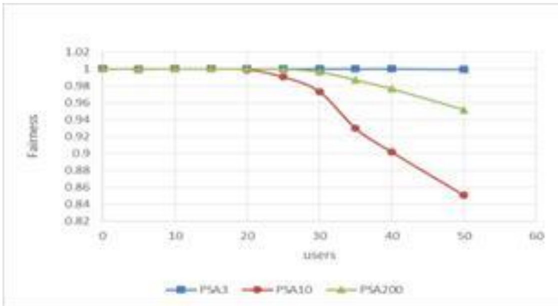


Figure 9. Fairness vs system capacity

However, PSA2 shows the least fairness values out of all three which is expected since it focuses on users with new packets and leaves the other users to wait either for their turn if it came up or to be discarded for vaulting the buffer delay threshold, second comes PSA3 which not shocking giving that the Mean User Throughput values come second to PSA1, but still it's much more fair than PSA2, now PSA1 comes on top in terms of fairness because it has the best Mean User Throughput values throughout the whole system run, regardless of this ranking all of three algorithms don't cross the fairness threshold which means they are fair to their users. So, we can come up with a conclusion that this algorithm works and meets the three performance measure that we choose to validate it, in terms of mean user throughput it gives almost perfect throughput up till 25 users but even then the decrease after that is not that big in value and as observed from the fairness graph this algorithm is fair to all users. Referring to the obtained results it can be mentioned that PSA3 is much better improvement over its successor PSA2 12% in Mean User Throughput and 11% in Fairness, while PSA3 still relatively great packet scheduling algorithm it couldn't beat PSA1 in terms of performance measures, PSA2 performed the worst because it prioritizes new users and it allocated all available RBs to the scheduled user leaving the rest to wait in the buffer, then comes PSA1 it has great value Mean User Throughput and fairness which is because it schedules each user on its RB which leads to multi-user diversity. Table 3 compares the achievable fairness for our PSAs to recent work in [19]. It shows that our PSA3 achieves much better performance in terms of fairness for the same type of traffic.

Table 3. Fairness Index Comprison between PSAs and Recent Works

No. Users	PSA1	PSA2	PSA3	[19]
10	1	1	1	0.85
20	1	1	1	0.83
30	0.975	0.997	0.9985	0.79
40	0.95	0.955	0.997	0.75
50	0.85	0.95	0.997	-

4. CONCLUSION

LTE-A is evolutionary path of LTE that is being developed by 3GPP to meet the requirements of people demands. LTE-A enhances the capability of LTE by providing much higher data rate, low latency and higher spectrum efficiency. It can be observed that the best algorithm is M-LWDF and that's why it was decided that the proposed algorithm will be M-LWDF for its great performance and satisfying the QoS for real time applications. This research aims to find a packet scheduling algorithm over M-LWDF to improve the real time multimedia performance in LTE-A, It can be concluded that the proposed algorithm is a good algorithm given its Mean User Throughput and fairness values it's a big improvement over PSA1 which is the objective of this paper that is to improve a packet scheduling algorithm and it does satisfy the QoS requirements for real time multimedia applications, which makes a great real-time multimedia performance.Nevertheless, researching other algorithms beside M-LDWF algorithm and see where their limitation is and figure out a possible method to develop it into a better packet scheduling algorithm to compete with the ever evolving algorithms to suit the futuristic technologies and applying the proposed algorithm technique on PSA1 and see where that could lead in terms Mean User Throughput and Fairness or see if there are other ways of developing the PSA3 itself beside giving equal opportunity would be a suggestion as future work.

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