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A proposed scheduling algorithm for real time application in 5G networks

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ABSTRACT

The third-generation partnership developed the fifth-generation specifications to satisfy the expansion of mobile applications and the grown demand for extra data flow. As the real time services in 5G networks are widespread, professional scheduling algorithms are necessary to deal with the assignment of the scarce frequency resources among different categories of applications, ensuring the quality of service and improving the user experience. This paper proposes a real time flow scheduling algorithm by enhancing the scheduling metric to prioritize real time flows such as voice and video, particularly as the packet delay approaches its threshold time. The performance metrics of the proposed algorithm were evaluated and compared to three well-known algorithms, which are the modified largest weighted delay first, the exponential proportional fair, and the logarithmic rule. The simulation results, which was conducted by a dedicated software, showed that the proposed algorithm achieved up to 1.5 times the throughput of the other algorithms and resulted in less than half the video packets loss ratio compared to others, moreover, it offered a higher fairness index between users than other algorithms for video packets.

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1. INTRODUCTION

The International Mobile Telecommunications (IMT-2020) introduced the fifth generation (5G) standards to meet the requirements of the rapid increasing of wireless devices and the new applications with high data rate and low latency requirements [1]. 5G networks can attain up to 20 Gbps of data rates, 20 times than that of the 4G network, and a total time delay as low as 10 ms for real-time packets [2]. Moreover, 5G provide high system spectral efficiency, larger data density per unit area, higher reliability, better energy efficiency [3].

5G networks come with three distinctive use cases, enhanced mobile broadband (eMBB), massive-machine type communication (mMTC), and ultra-reliable low-latency communication (URLLC) services [4]. Employing these use cases necessitates a more robust system since each one of these cases demands specific special requirements [5].

The packet scheduler component of the radio resource management (RRM) gathers the information from users continuously, then applies the intended quality of service (QoS) to each flow type to guarantee the optimum utilization of the resources [6]. RRM is a set of procedures, strategies, and algorithms responsible for managing resource allocation, data rates, modulation and coding scheme among others [7]. The extensive number of deployed devices and the various types of services and flow types complicate RRM designing and deploying. The varied QoS requirements of each application, the spectrum scarcity, the channel information of each user in the network, and the user mobility are factors that affect the allocation of resources in the network. Thus, QoS must be configured to ensure acceptable delay and data rate with the lowest possible loss ratio. To achieve the optimal QoS for all users, the MAC scheduler should be designed to ensure the efficient resource allocation of the various 5G scenarios [8].

Traffic can be real-time (RT) or non-real time (NRT) with different requirements of QoS. Internet users show a huge demand for real time applications like voice over internet protocol (VoIP) and live video streaming. VoIP traffic is a time critical application with high priority and reliability requirements. While, video streaming traffic requires high data rates and low latency serving large number of users whereas guaranteeing adequate network performance [9]. Current real-time interactive applications require more sophisticated QoS targets concerning delay and packet loss ratio (PLR) as performance indicators beside high network performance to ensure VoIP and video quality. These indicators are vital for both user experience and QoS in real-time applications [10].

The available scheduling algorithms can be classified as channel-aware, QoS-aware, channel-and-QoS-aware or none of those. The channel quality is considered to enhance network performance, while the QoS are employed to satisfy the desired service's quality parameters [11]. Nwawelu *et al.* [12], an algorithm is evaluated which gives extra QoS support to the network while increasing the number of supported users. This algorithm predicts the incoming traffic and gathers its channel statistics to help the scheduler to assign the resources to the user with current highest channel state. However, this algorithm has a lower priority for RT flows in case of PLR and throughput. Angri *et al.* [13], an algorithm is proposed for large number of users with high speed. It fulfills the needs of the QoS of RT applications concerning average throughput, packet delay, and PLR. For NRT flows it considers the channel condition and the average data rate. While for RT flows, the delay is taken into consideration using an exponential function giving enhancement for RT flows. This may result in unfair resource allocation between RT and NRT flows. A new scheduler is suggested by Nwawelu and Ani [14] to enhance the priority of RT over NRT services, offering a good performance, and satisfying the QoS requirements of the PLR for RT services. The scheduler divides the users to RT and NRT users, then assigns radio resources to users such that standard network performance metrics are satisfied, which make the algorithm more complex than other algorithms. Latiff *et al.* [9] studied an algorithm performance in 5G network. This algorithm combines the proportional fair metric with the head-of-line (HoL) packet delay, and constrained by the packet delay threshold and the permissible loss packets probability. This algorithm gives a lower priority to RT flows compared to NRT flows.

In this work, an algorithm is proposed to maintain the QoS of RT traffic while keeping the minimum required fairness and throughput for the NRT traffic for users. This algorithm gives a fair priority of RT packets newly buffered in the queues with NRT flows, and assigns higher priority to RT flows spent longer time in buffers, achieving better RT throughput and lower PLR.

2. SCHEDULING TECHNIQUE

Scheduling is the process of allocating radio resources among users. Several factors affect the scheduling, such as channel state information (CSI), buffer status report (BSR), and QoS [15]. Several factors affect the QoS, such as target delays, available resources, channel conditions, and service types (RT or NRT) [8].

5G NR employs the orthogonal frequency division multiplexing (OFDM), where time and frequency domains can be used for scheduling as shown in Figure 1. A resource element (RE) is the basic time-frequency resource unit that consists of one subcarrier in the frequency domain and one OFDM symbol in the time domain. Frequency domain is divided into several carriers called subcarriers. 5G supports different subcarrier spacings which are 15 (as in LTE), 30, 60, 120, and 240 kHz, and a range of channel bandwidths up to 400 MHz [8]. A resource block (RB) has twelve contiguous subcarriers. It is considered the smallest resources unit that can be allocated to a user. Time domain is divided into radio frames, subframes, slots and mini-slot. The duration of a radio frame is 10 ms and is divided into 10 subframes with an interval of 1 ms each [16]. Every subframe has one or more slots of 14 OFDM symbols. A mini-slot can be either 2, 4, or 7 symbols. The slot interval relies on the subcarrier spacing [15].

5G supports two types of duplexing, the frequency division duplex (FDD) which assigns different frequency channels to uplink and downlink traffics, and the time division duplex (TDD) which transmits uplink and downlink packets separated by time through a single channel [17]. User equipment (UE) reports the standard CSI at each transmission time interval (TTI). CSI has several components of information, such as channel quality indicator (CQI), and precoding matrix indicator [18]. CQI is an integer of four bits, representing the data speed a device can manage maintaining an error of 10% or less. The CQI is a function of the signal to interference and noise ratio (SINR), although it depends on the device implementation. According to the reported CQI, the RRM module specifies the modulation and coding scheme (MCS) [19]. Packets are arranged in various queues, then, the packet scheduler allocates the available physical radio blocks (PRBs) to the current flows with the largest metric value [8]. The scheduling decision is made according to several factors, including buffer size, channel conditions, and packet delays. For every user, a transport block (TB) is built containing the data to be transmitted [6].

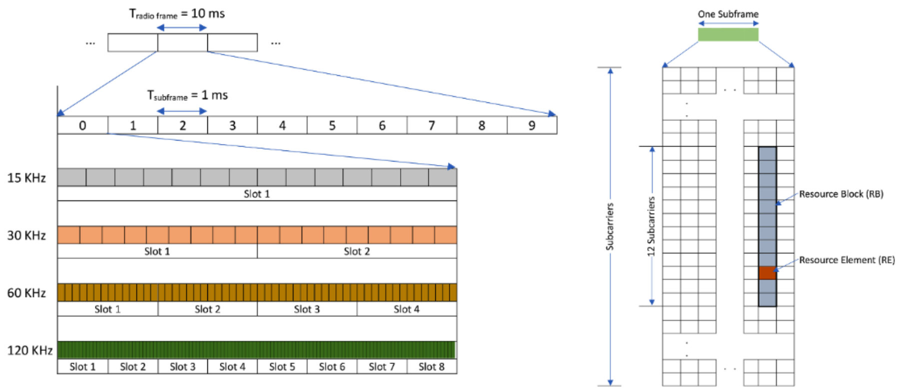

Figure 1. 5G radio frame structure and resource grid

3. EVALUATION PARAMETERS:

Four performance parameters are used to evaluate the algorithms in this paper as follows.

3.1. Throughput evaluation

The throughput is the size of the successfully delivered packets to the destination divided by the delivery time as in (1). It is the main metric employed to evaluate the network performance and to obtain its efficiency. Where R_{rx} is the data rate of the received users' data in bits, and T is time taken to reach the destination in seconds.

$$Throughput = \frac{1}{T}\sum R_{rx} \tag{1}$$

3.2. Packet loss ratio evaluation

The PLR evaluation is an important factor to determine the network performance, particularly for RT applications like VoIP and video. It specifies the ratio of the number of the lost packets to the whole number of the transmitted packets as shown in (2).

$$Packet\ Loss\ Ratio = \frac{N_{tx}-N_{rx}}{N_{tx}} \times 100 \tag{2}$$

where N_{tx} is the number of transmitted packets and N_{rx} is the number of received packets.

3.3. Fairness index evaluation

The fairness index is the measurement of the resource allocation fairness between users. The Jain's fairness index is employed which is a function of the data rates obtained by the users as shown in (3) [18]. This index ranges from (1/N) to 1. The higher the value the higher the achieved fairness between users, with 1 being the optimum fairness.

$$Jain's\ Fairness\ Index = \frac{(\sum_{i=1}^{N} R_i)^2}{N\cdot \sum_{i=1}^{N} {R_i}^2} \tag{3}$$

where R_i is the i^{th} user throughput, and N is the whole number of users.

3.4. Packet latency evaluation (delay)

Latency is the time spent by a packet to be transmitted from its source to the destination. It has a noticeable influence on time critical applications like VoIP and video which requires low delay [20]. The average delay is the sum of the packet delays divided by the total number of received packets as illustrated in (4).

$$Average\ Delay = \frac{1}{N_{rx}}\sum_{i=1}^{N_{rx}}(T_{rx,i} - T_{tx,i}) \quad (4)$$

where N_{rx} is the number of received packets, $T_{rx,i}$ is the time the i^{th} packet received, and $T_{tx,i}$ is the time the packet transmitted.

4. SCHEDULING ALGORITHMS

Scheduling algorithm makes a decision for allocating an available resource (j^{th} RB) to the (i^{th}) user by evaluating and comparing the metric values ($m_{i,j}$) for each user, then assigning the resource to the user having the maximum metric value ($w_{i,j}$) [21], as in (5).

$$w_{i,j} = max_i(m_{i,j}) \quad (5)$$

Numerous numbers of scheduling algorithms were developed to maintain the QoS targets and to enhance throughput and fairness [22]. Following is a brief description of some of the scheduling algorithms.

4.1. Round robin (RR)

RR is one of the most basic algorithms which assigns each UE an equal number of resources in a cyclic format without priority to any one of these UEs [23]. The advantage of RR is that it allocates resources fairly between UEs. However, RR doesn't take channel condition into account, which may result into poor network performance and a waste of network resource [24].

4.2. Best channel quality indicator (BCQI)

The best CQI scheduler allocates resource blocks to the users with the best channel conditions. Each TTI, the UEs reports their CQI to the base station. Higher CQI value represents a better channel condition. This scheduler achieves the best network utilization by scheduling the users with the highest CQI values. However, this may result in unfair distribution of resources, since UEs at the cell edge suffering poor channel may not be scheduled [25].

4.3. Proportional fair (PF)

The PF algorithm provides an optimal balance between throughput and fairness by allocating the available resources among users, considering the current data rate which is a factor of channel quality experienced by the user, and the average user's throughput [22]. It intends to reach high level of fairness with acceptable throughput and to improve the QoS for various levels of traffic load conditions [21]. As shown in (6), the metric $m_{i,j}$ determines the ratio between $r_{i,j}(t)$ and $R_i(t)$, where $r_{i,j}$ is the UE instantaneous data rate taking into consideration the CQI value stated by the (i^{th}) UE on the (j^{th}) RB, and $R_i(t)$ is the average data rate of the (i^{th}) UE [26].

$$m_{i,j} = \frac{r_{i,j}(t)}{R_i(t)} \quad (6)$$

The previous average data rate of a user (i) represents the history of the user's allocated resources. It enhances the fairness of resource distribution between users by prioritizing users who had low throughput. Every TTI, the achieved instantaneous average data rate $R_i(t)$ is updated as in (7):

$$R_i(t) = \left(1 - \frac{1}{t_c}\right)R_i(t-1) + \frac{1}{t_c}r_{i,j}(t) \quad (7)$$

Such that $R_i(t-1)$ is the past average data rate, and t_c is the constant time window length, used as an averaging filter [27].

4.4. Modified largest weighted delay first (M-LWDF)

This is a channel and QoS aware algorithm, it considers the delay, fairness, and network performance, and handles RT and NRT flow types differently by enhancing real time flows with the highest delay to be transmitted before reaching the threshold time. The metric is specified in (8) and (9):

$$m_{i,j} = a_i D_{HoL,i}\frac{r_{i,j}}{R_i} \quad (8)$$

$$a_i = -\frac{log(\delta_i)}{\tau_i} \tag{9}$$

where $r_{i,j}$ and R_i are the same as those in the proportional fair metric. $D_{HoL,i}$ is the packet (i) head of line delay which is the time the packet spent in the buffer before transmission, τ_i is the delay threshold of the (i) real-time flow as the packet is considered lost after passing this time in the buffer, and δ_i indicates the maximum probability to allow for packets to exceed the threshold time [15].

4.5. Exponential proportional fairness (EXP/PF)

The EXP/PF enhances the real time traffic of the multimedia services. It intends to enhance RT flows priority over NRT flows by using the average fixed maximum time of all active RT flows. For RT flows, the metric priority is increased when the HoL packet delays reach the delay threshold time. For RT flows [27], the metric is employed as illustrated in (10) and (12):

$$m_{i,j} = exp\left(\frac{a_i D_{HoL,i} - X}{1+\sqrt{X}}\right)\frac{r_{i,j}}{R_i} \tag{10}$$

$$X = \frac{1}{N_{rt}}\sum_{i=1}^{N_{rt}} a_i D_{HoL,i} \tag{11}$$

where *Nrt* is the number of RT flows.

4.6. Logarithm rule (LOG-rule)

This scheduler fulfills the QoS requirements of the network. It gives an enhanced priority to flows with high rate [12]. The metric is defined in (12) and (13):

$$m_{i,j} = b_i \ log\left(c + a_i D_{HoL,i}\right)\frac{r_{i,j}}{R_i} \tag{12}$$

αi, bi, c could be set as (13)

$$a_i = \frac{5}{0{\cdot}99\ \tau_i}, b_i = \frac{1}{E\left(\frac{r_{i,j}}{R_i}\right)}, c = 1 \cdot 1 \tag{13}$$

Table 1 summarizes a comparative information about the key methods mentioned above, in case of factors used and complexity.

Table 1. Channel and QoS factors for some scheduling methods

Scheduling methods	Channel factors			QoS factors			Complexity
	CSI	Average data rate	Target rate	Service type	Target delay	HoL delay	
Round robin				NRT			Basic
Best-CQI	✓		✓	NRT			Simple
PF	✓	✓	✓	NRT			Simple
M-LWDF	✓	✓	✓	RT	✓	✓	Moderate
EXP/PF	✓	✓	✓	RT, NRT	✓	✓	High
LOG-rule	✓	✓	✓	RT, NRT	✓	✓	High

5. PROPOSED SCHEDULER

The proposed algorithm takes into consideration RT and NRT flows. For NRT flows, the CQI is employed to achieve the highest throughput for users. So, $r_{i,j}$ is defined as the instantaneous data rate that the i^{th} user could obtain on the j^{th} RB at current time. Users with higher CQI will get higher priority, which results in unfair resource distribution. Thus, the metric should consider the user previous average data rate R_i, to add a fair assignment of resources to users. The metric for NRT is then as in (14):

$$m_{i,j} = \frac{r_{i,j}}{R_i} \tag{14}$$

For RT three parameters need to be considered. The head of line delay (D_{HoL}) which is the time spent by the packet in the buffer for the i^{th} user, the threshold time that is the maximum permissible time delay (τ_i)

for the RT packet before being dropped, and the maximum probability (δ_i) allowed for packets delay to exceeds the threshold time. The final metric for RT flows is illustrated in (15):

$$m_{i,j} = \frac{r_{i,j}}{R_i} \exp\left(-log(\delta_i)\frac{D_{HoL,i}}{1-\frac{D_{HoL,i}}{\tau_i}}\right) \quad (15)$$

when the D_{HoL} parameter is zero, the metric behaves like the NRT metric, as the D_{HoL} increases towards τ_i, the metric priority increases exponentially resulting in more RT flows to be scheduled before reaching the threshold time to prevent being dropped. As shown from the metric, this algorithm has less computational complexity than EXP/PF and LOG-Rule algorithms, but higher complexity than M-LWDF.

6. SIMULATION MODEL

The simulation is conducted using 5G-air-simulator [28]. In this simulation, the more realistic Single Cell with Interference configuration was employed. As shown in Figure 2, the model contains 7 cells, each cell has a radius of 1 Km, a base station at the center serving users, surrounded by six base stations that do not serve users, but produce inter-cell interference which impact the metrics in the primary cell. Users move in random direction with a constant speed of 3 km/h inside the cell using the random direction mobility model. The urban macro-cell channel model is used [29].

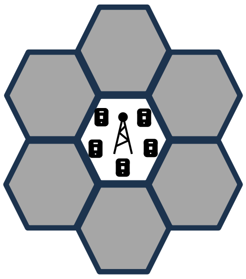

Figure 2. Simulation model, a primary cell surrounded by 6 cells causing interference

Three traffic models are used in this simulation. The best effort (BE) modeled by infinite buffer model offers infinite supply of. The VoIP model uses the G.729 model which generates packets of constant rate and size at different times imitating the way human speak. The TraceBased emulates the video streaming traffic which generated from real video file with full size and time information of every frame.

Each user has one BE flow, and either a video stream or a VoIP stream depending on the simulation scheme. To study the effect of varying traffic load, the simulation is applied to different number of users, from 5 to 50 users, with a step of 5 users. Values of 0.1 s for the max delay threshold and 0.005 for the drop probability are considered acceptable values for the VoIP and video traffic [13].

The proposed algorithm beside other three algorithms which are the LOG-Rule, M-LWDF and EXP-PF schedulers are evaluated and their performance metrics are compared to each other in terms of throughput, PLR, fairness, and delay. The simulation parameters are shown in Table 2.

Table 2. Simulation parameters' values

Parameter	Value
Carrier frequency	2.1 GHz
Bandwidth	5 MHz
Frame structure	FDD
UE speed	3 Km/h
Radius	1 Km
Number of base stations	7
Simulation duration	46 second
Simulation flow duration	40 second
Channel model	Urban macro-cell
Max delay threshold (τ)	0.1 second
Drop probability (δ)	0.005
Video bit-rate	242 kbps
Number of users	5, 10, 15, 20 ,25, 30, 35, 40, 45, 50

7. SIMULATION RESULTS AND DISCUSSION

Following is the evaluation and comparison of the performance of the proposed algorithm and the other algorithms. This simulation considers the evaluation metrics of throughput, PLR, fairness index, and delay as the Y-axis. Different number of users are studied which represent various loads as the chart's X-axis.

7.1. Throughput for VoIP flows

The throughput of the VoIP flows against different number of users illustrated in Figure 3. It shows a close data rate of the four evaluated algorithms for all the number of users. VoIP flows are of low data rate compared to video flows, which make the algorithms to have almost equal values of data rates.

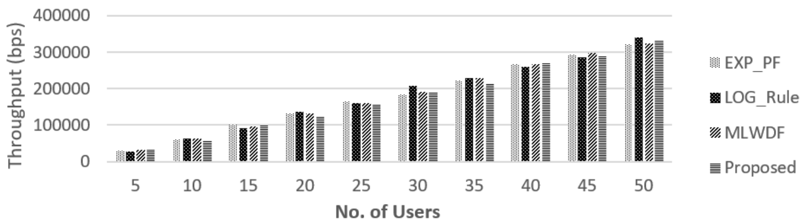

Figure 3. The throughput for VoIP flows

7.2. Throughput for video flows

Figure 4 illustrates the video packet throughput for various number of users. It is obvious that the proposed algorithm outperforms other scheduling algorithms, and is more noticeable when having high number of users with high traffic load. At 50 users, the proposed algorithm achieved throughput of over 6.9 Mbps much more than the next algorithm, EXP-PF, which reached 4.5 Mbps. This is due to the enhancement of the metric to prioritize the real time flows, which results in higher video bit rate.

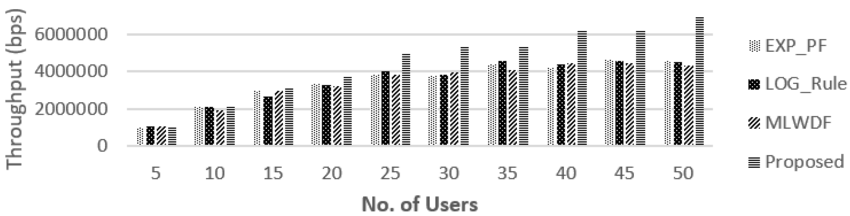

Figure 4. The throughput for video flows

7.3. PLR for the VoIP Flows

The PLR depicted in Figure 5 shows that the proposed algorithm generally presents PLR than other algorithms for the different number of users. A range of 0.7 to 3.3 percent which is much lower than the target PLR of 10%. This low PLR of the proposed algorithm is due to the high priority the metric gives to the RT flows, preventing them from being lost.

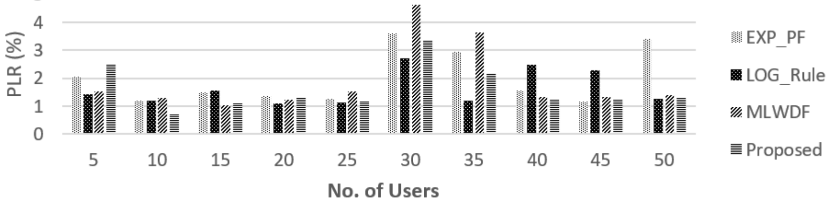

Figure 5. The PLR for VoIP flow

7.4. PLR for the video flows

The PLR for video flows given in Figure 6 illustrates a very low PLR of the proposed algorithm, less than all the other algorithms for different number of users. As the number of users increases, the proposed algorithm achieves 26%, almost half the loss packets of the other algorithms which achieved more than 48%. This is the result of the design of the metric which prioritizes the RT flows particularly video data, which improves the performance of the proposed algorithm for video flows.

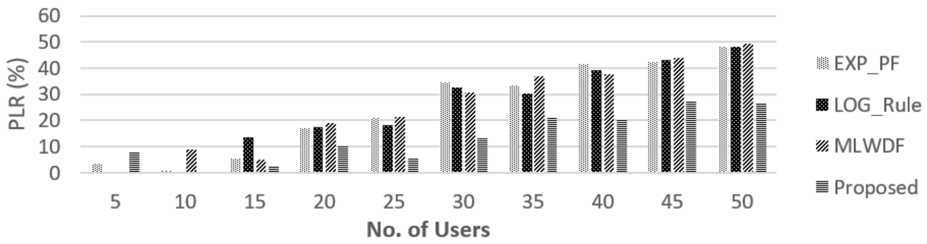

Figure 6. The PLR for video flow

7.5. Fairness Index for VoIP flows

The fairness index of the VoIP flows illustrated in Figure 7 shows a close fairness index values for all the evaluated algorithms. And the values decrease with the increase of the number of users. The proposed algorithm index ranges from 0.7 to more than 0.9.

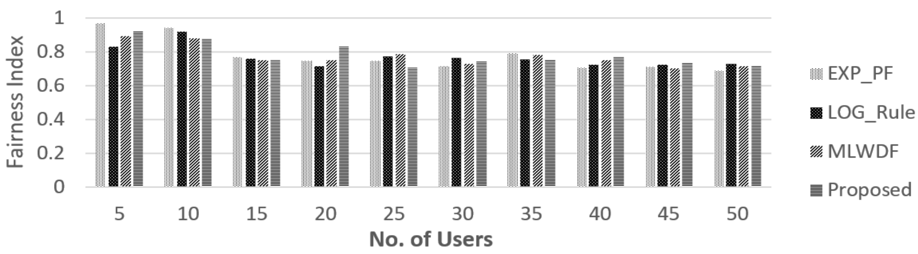

Figure 7. The fairness index for VoIP flows

7.6. Fairness index for video flows

As illustrated in Figure 8, the fairness index of the video throughput shows higher values for the proposed algorithm as the number of users rises compared to the other algorithms. It achieves values of 0.8 up to 0.91 for the different number of users which improves the user's quality of experience. The metric aims to prioritize RT flows equally resulting in higher fairness between users.

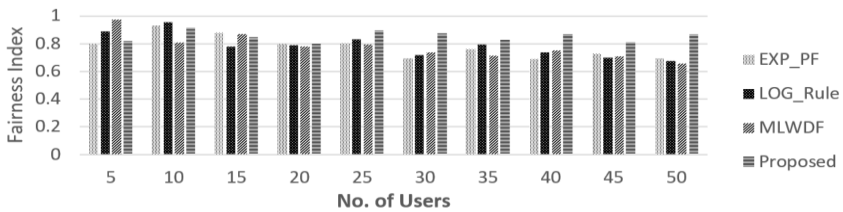

Figure 8. The fairness index for video flows

7.7. Delay for the VoIP Flows

The Figure 9 shows that the proposed algorithm results in a short duration of delay for different number of users. The same is observed for the other algorithms. The maximum value achieved by the proposed algorithm is 2.3 ms which is much less than the threshold time of 100 ms.

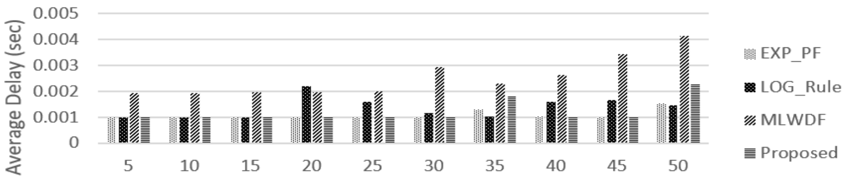

Figure 9. The average delay for VoIP flows

7.8. Delay for the video flows

As we can see in Figure 10, the average video delay of the proposed algorithm is not lower than other algorithms, the maximum delay obtained by the proposed algorithm ranges from 1 ms up to 75 ms, which is within the acceptable adopted range in this simulation where the maximum threshold delay is 100 ms. The advantage of the higher data rates and fairness and lower PLR comes at the expense of the higher delay. Nevertheless, it is still less than the maximum allowed value.

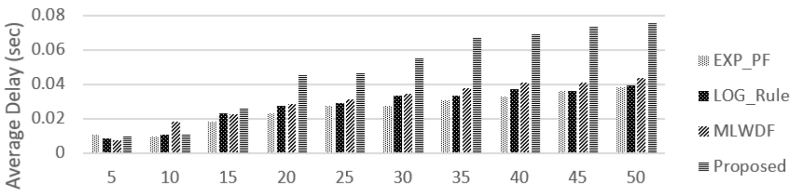

Figure 10. The delay for video flows

8. CONCLUSION

A new proposed algorithm for 5G networks is evaluated and its metrics performance is compared to others which are the LOG-Rule, EXP/PF, and the MLWDF, for various number of users which represents different traffic load. This analysis evaluated the algorithms in terms of throughput, PLR, fairness index, and delay. The results showed: i) for throughput, the simulation exposed that for VoIP traffic, the values of data rates increase gradually from almost 30 kbps for 5 users up to more than 300 kbps for 50 users, for each algorithm. While for video traffic, the throughput increases form almost 1 Mbps for 5 users to approximately 4 Mbps for 25 users and keeps near values for higher number of users, and that is because of the full utilization of the available resources at 25 users due to high traffic of video flows compared to VoIP traffic; ii) the PLR results for VoIP show that the values of all the algorithms are very small of less than 5% with the extreme case obtained at 30 users, that is due to the small size of the voice data. While for video traffic the values start very low at 5 users, then increase gradually as the number of users increases, since video flows are bigger in size and this causes a lot of data packet loss; iii) the fairness index values for VoIP and video flows start to be high at 5 users of as high as 0.97, then decrease gradually as the number of users increases down to almost 0.65 for MLWDF; iv) the delay results show almost near values for different number of users for VoIP flows of less than 5 ms. The voice flows are small in size which take less time to be delivered. On the other hand, video flows are of large sizes which take longer time to reach distention, values of less than 76 ms were recorded. Therefore, the proposed method had a higher delay than other algorithms as a consequence of the higher achieved throughput of the video flows than other algorithms; and v) lastly, the results expressed, and due to the careful design of the proposed algorithm metric parameters, that the algorithm achieved a better throughput, a significant lower PLR and higher fairness index than other algorithms, and the results were superior for video flows than for VoIP flows particularly for higher number of users. The average delay for VoIP flows is close to other algorithms, while for video flows is higher, but still within the acceptable range of the maximum threshold values of the delay, which is trade-off of the higher video throughput achieved that leads to a higher video packet delay. Moreover, the proposed algorithm metric is designed to be less computational complex than EXP/PF and LOG-rule, since they use extra computations such as summation and averaging functions.

As a future work, this algorithm can be used in multi-level schedulers which have multiple algorithms used depending on the types of the flows, where this algorithm is used to schedule real time flows especially video traffic. Moreover, the proposed algorithm can be studied in different scenarios with different simulation parameters to show the effect changing the parameters values on the proposed algorithm compared to others.

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AUTHOR CONTRIBUTIONS STATEMENT

This journal uses the Contributor Roles Taxonomy (CRediT) to recognize individual author contributions, reduce authorship disputes, and facilitate collaboration.

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C : **C**onceptualization
M : **M**ethodology
So : **So**ftware
Va : **Va**lidation
Fo : **Fo**rmal analysis
I : **I**nvestigation
R : **R**esources
D : **D**ata Curation
O : Writing - **O**riginal Draft
E : Writing - Review & **E**diting
Vi : **Vi**sualization
Su : **Su**pervision
P : **P**roject administration
Fu : **Fu**nding acquisition

CONFLICT OF INTEREST STATEMENT

Authors state no conflict of interest.

DATA AVAILABILITY

The authors confirm that the main data supporting the findings of this study are available within the article. Any further information that may support the findings of this study is available from the corresponding author [Moaath Saleh Abdulrahman], upon reasonable request.

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