Resource Allocation for Real-time Multimedia Applications in LTE's Two-level Scheduling Framework

Ang Ee Mae, Wee Kuok Kwee, Pang Ying Han and Lau Siong Hoe

***Abstract*— Long-Term Evolution (LTE) was brought in to fulfill the Quality of Service (QoS) demands and needs of users. Existing buffering technique do not have QoS factors and does not support live feed streaming. There are various solutions for problems in the downlink packet scheduling, but there is still room to have a higher QoS to provide greater satisfaction to users. A better user experience requires real-time applications to have minimal amount of delay. Thus, three novel schemes, that regards real-time applications, are proposed and developed. The performances of our proposed algorithms are compared with the latest two-level scheduling algorithm, Frame Level Scheduler and Maximum Throughput (FLS-MT). Our proposed frameworks, 1) FLS and Enhanced Exponential with MT (FE2M) downlink scheduler, 2) FE2M and Exaggerated Earliest Dateline First (E2DF) with MT as weight and best effort packets using the basic MT rule (FE2E2M_I), and 3) FE2M and E2DF with MT as weight and best effort packets using the Proportional Fairness (PF) rule (FE2E2M_II), have exploited innovative approaches based on packet delay at the lower level. The upper level of the scheduling algorithm remains as it is based on discrete-time linear control. A comparative study of recent existing scheduling strategies implemented in LTE are presented. Based on extensive experimental tests, the results generated clearly show that the simulation results of our proposed lower-level scheduling frameworks, FE2M, FE2E2M_I, and FE2E2M_II, outperform the existing two-level scheduling algorithm in terms of throughput, delay, spectral efficiency, and packet loss rate for real-time multimedia applications.**

***Index Terms*—Downlink, LTE network, Quality of Service, real-time, scheduler**

I. Introduction

As the number of mobile broadband users keep increasing from year to year, quality of service (QoS) is strongly taken into consideration to give a much better network quality. This include using devices for online games, video streaming, website browsing, and so on. Mobile social networks are expanding as more users exploit these integrated technologies [1]. In [2] and [3], the authors provide a brief introduction in regard to the various network architectures that lead towards the 4G network. The 3rd Generation Partnership Project (3GPP) [4] has established Long-Term Evolution (LTE) to meet and satisfy the increasing demands for broadband services and requirements. An overview of the former 3GPP technologies interaction with LTE was given in [5].

LTE is one of the fourth generation (4G) standards which have recent updates of digital signal processors as well as modulation schemes to enable sustainability of high speed communications [6][7][8]. The evolution towards LTE offers higher throughput rate to mobile terminals, as service providers strive to deliver advanced mobile broadband services [9]. It is also recognized as the Evolved Universal Terrestrial Access Network (E-UTRAN) [1][6][10]. The LTE network consists of two sections, which include the core network, also known as the main component, and the radio access network. The core network is provided by the Evolved Packet Core (EPC) or the Service Architecture Evolution Core (SAEC), whereas the radio access network is provided by E-UTRAN. EPC consists of 3 components which are Packet Gateway (PGW), Service Gateway (SGW), and Mobility Management Entity (MME). See Figure 1 for the network architecture of LTE. In this paper, our enhancement focuses on the downlink packet scheduling which takes place from eNodeB (eNB) to the user equipment (UE).

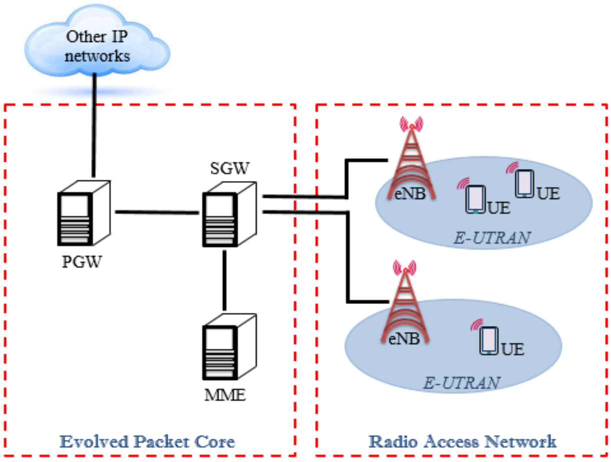

Fig. 1. LTE Network Architecture.
(Adapted with permission from F. Capozzi, G. Piro, LA. Grieco, G. Boggia, P. Camarda, "Downlink Packet Scheduling in LTE Cellular Networks: Key Design Issues and a Survey", *IEEE Communications Surveys & Tutorials*, vol. 15, no. 2, 2012, pp. 678-700. Copyright 2012, IEEE.)

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Proper QoS designation is vital. Improper use of QoS, may prime towards denial of service and low data bandwidth [11]. Several QoS factors are taken into consideration, including packet delay, packet loss, throughput and spectral efficiency, which is also known as bandwidth efficiency [12][13]. The delay may affect the packet loss rate (PLR) because as the delay increases, the queue will overflow, leading to the discarding of packets. In the case of a packet being discarded, the throughput is also affected. Lastly, the spectral efficiency is measured to see whether the data rate is transmittable over the communication channel. Our interest lies in the computation of metric for packet scheduling, which is situated in the Media Access Control (MAC) layer [7][14][15]. This can be viewed in the architecture given in Figure 2.

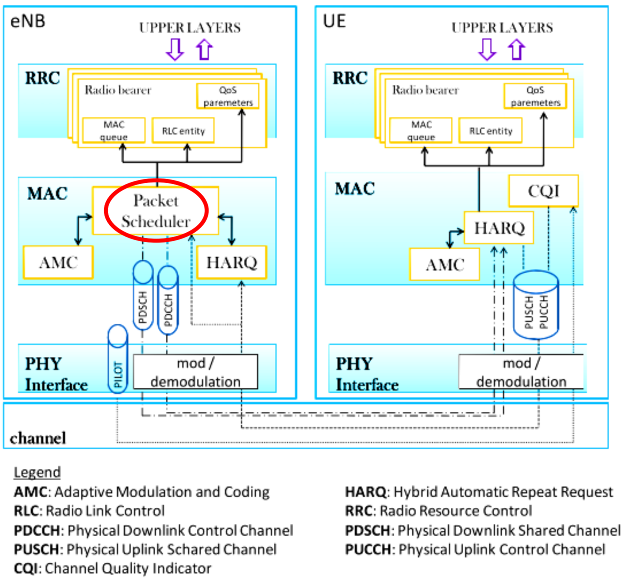

Fig. 2. The RRM features interaction.

(Adapted with permission from F. Capozzi, G. Piro, LA. Grieco, G. Boggia, P. Camarda, "Downlink Packet Scheduling in LTE Cellular Networks: Key Design Issues and a Survey", *IEEE Communications Surveys & Tutorials*, vol. 15, no. 2, 2012, pp. 678-700. Copyright 2012, IEEE.)

Currently, there are several existing scheduling algorithms developed by other researchers but there is still room for further research and improvement. This is especially the case for real-time traffic which involves video and Voice over Internet Protocol (VoIP) transmissions, as lower packet loss and delay rates are required for a better quality of experience (QoE) [16]. In addition to real-time applications, [17] stated that the quality of video's rely on several network factors which include packet loss, while [18] mentioned that VoIP clients experience signal degradations like jitter and packet end-to-end delay. Areas of improvement comprise of QoS provisioning [13][19][20]. It can be done by increasing the transmissions throughput and fully utilizing the network bandwidth efficiency at the same time minimizing the delay and PLR. Since there is much real-time mixing with non-real-time traffic transmission, there is still room to maximize the network's throughput. Buffering is required when the QoS is unable to satisfy the user needs. Best effort (BE) traffic is used to send data based on its availability and, unlike in the LTE network, no QoS is implemented for it [8][20]. The existing service provider (SP) provides only BE traffic flows for all packet transmissions; hence buffering does not work if users stream live feeds, like video conferencing. Another issue is the delay factor that buffering would cause [21].

The contributions of this work can be summarized as follows: Prior to this research, we found that the two-level scheduling algorithm proposed in [22] has several weaknesses, including the fact that its lower layer uses only Maximum Throughput (MT) for real-time traffic and the Proportional Fairness (PF) scheme for non-real-time traffic (details are further elaborated in Section 2). Considering only the maximum throughput of a channel is not sufficient for real-time multimedia applications. Furthermore, the MT scheme does not consider the delay parameter of the traffic. Thus, we have proposed three scheduling schemes that utilizes the two-level scheduling framework with enhancements on the lower level. All of our proposed two-level scheduling frameworks impose packet delay as the top priority component in our scheduling frameworks for real-time multimedia traffic. As a result, these schemes are able to improve the performance of real-time traffic in the network in terms of delay, PLR, throughput, and also spectral efficiency compared to the two-level scheduling algorithm in [22]. One of our scheduling schemes that stands out is the FE2E2M_I (details are further elaborated in Section 3), which significantly contributes towards the improvement in efficiency of the network's performance in terms of delay and PLR of real-time traffic. The experimental results have proven that our proposed scheduling scheme provides a better QoE and QoS at the same time, leading to satisfaction of the end users of multimedia services as it is able to satisfy user demands for an efficient and more reliable wireless network.

The paper consists of several sections. Section 2 consists of several reviews on the latest implementation of existing scheduling algorithms. Section 3 elaborates the details of our proposed scheduling schemes. The scenarios and parameters for the simulation are described in Section 4 along with the results and discussion based on the simulation performed. Finally, in Section 5, concluding observations are expressed.

II. Literature Review

Several recent works on the scheduling algorithm for both real-time and non-real-time applications that are applied in the LTE network were studied and briefly elaborated in the following sub-sections. Refer to Table I to obtain the notation meanings.

TABLE I
Scheduling Metrics Notations

Expression	Meaning
$m_{i,k}$	Common metric for the k^{th} resource block for the i^{th} user
$d_k^i(t)$	The achieved data rate at period t by the i^{th} user
$\bar{R}^i(t)$	The achieved average throughput in the past by the i^{th} user up till time t
$D_{HOL,i}$	The i^{th} packet's head-of-line delay
τ_i	The i^{th} user's threshold delay
$w_{i,k}$	The channel's weighted spectral efficiency for the i^{th} user in the k^{th} resource block
δ_i	The i^{th} user's acceptable rate for packet loss

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As stated in Table I, the metric expression is given as $m_{i,k}$. Its role is to determine which packet should be transmitted first. Based on the traffic flow, the eNB shall assign each sub-channel to the metric with the highest value [23]. There are several scheduling schemes of which its functionality stands out and most commonly used by other researchers in their works. The following are the schemes arranged in sub-sections based on channel and QoS awareness.

A. *Channel and QoS Aware*

QoS parameters play a role in the scheduling behavior of the data in guaranteeing a minimal requirement for the performance be it in terms of packet delays in delivery as well as assured data rates. It is highlighted that QoS-awareness does not significantly mean QoS provision [8]. This is due to the fact that it relies on each flow requirement to yield the allocation decision. Hence, it does not have to comply with guaranteeing such requirements as it can be unfeasible due to unimplemented procedures in admission control. The implementation of QoS guarantee strategy is said to be challenging as QoS violations are prone to occur due to low reliability in the wireless channels as well as inconsistent strength of the wireless signal [24].

1) *Modified-Largest Weighted Delay First (M-LWDF)*

Currently existing schemes include the M-LWDF, which is a scheduling strategy that is aware of the channel as well as QoS. It is commonly referenced by researchers because of its ability to distinguish between non-real-time and real-time flows by providing priority to packets in the real-time transmission that have the highest waiting period with reference to the packets' head of line (HOL) and the most desirable channel state [25]. The scheme's metric is expressed by (1):

$$m_{i,j}^{M-LWDF} = \alpha_i D_{HOL,i} \gamma_{i,j} \tag{1}$$

where the metric is represented by the symbol $m_{i,j}^{M-LWDF}$. The data rate functions by weighing the metric and obtaining a metric with strong conditioning, whose allocation is greatly preferable, like the packet expiry date and acceptable loss rate; it is given in the form of α_i, while the HOL delay for the i^{th} packet is given as $D_{HOL,i}$, and $\gamma_{i,j}$ is the weighting feature that assures that users in very poor channel conditions will be served within a given time constraint [8].

A more elaborate explanation of the data rate, α_i, is given by (2):

$$\alpha_i = -\frac{\log \delta_i}{\tau_i} \tag{2}$$

where the threshold value for the packet's waiting period, τ_i, along with the probability of packet loss, δ_i, is categorized as the highest probability intended for the packet delay's HOL, $D_{HOL,i}$, which exceeds the delay value's threshold.

With regard to the MAC queue, real-time transmission packets are erased from the queue if the packets expire after the given transmission period. This scheme is characterized to be throughput-optimal; it can deal with all of the queues steadily and can also deal with the conditions, and therefore various methods are able to perform viably with it [26].

2) *Exponential Rule (EXP Rule)*

The EXP Rule is known to be an enhancement of the EXP/PF and is also considered to be more robust than the Logarithm Rule [8][27]. Moreover, this method is channel-aware. In [8], the delay of a user is in some way normalized over the total amount of delay of all the experienced users. Hence, it is known that the EXP Rule performs scheduling decisions based on the packets' actual delay. Further detailed explanation in regards to the mathematical computation of the EXP Rule is provided in Section 3.1.

3) *Logarithmic Rule (LOG Rule)*

The LOG Rule is known to be almost the same as the EXP Rule, as it is sensitive towards the condition of the channel [8]. As the queues keep increasing in size, the LOG Rule is able to schedule them in a way that reduces the prominence of queue-balancing in order to increase the overall weighted service rate [28].

The symbol $m_{i,k}^{LOG\ Rule}$ is the metric value for LOG Rule and the equation is shown in (3) [8]:

$$m_{i,k}^{LOG\ Rule} = b_i \log_{10} (c + \alpha_i D_{HOL,i}) \Gamma_k^i \tag{3}$$

where a classification of several favorable parameters, which include α_i, b_i, and c, are specified as $\alpha_i \in [\frac{5}{(0.99\,\tau_i)}, \frac{10}{(0.99\,\tau_i)}]$, $b_i = \frac{1}{E[\Gamma^i]}$, and $c = 1$. The character Γ^i can be replaced by the k^{th} channel's spectral efficiency value, which ranges between 1 and N for the i^{th} user.

Mathematically, the role of the Logarithm is somewhat similar to that of the exponential function. The difference is that the Logarithmic function performs the inverse of the exponential function [29].

4) *Frame Level Scheduler (FLS)*

This scheme is also known as the Guaranteed Delay Scheduler (GDS) [30][31]. According to [32], the FLS functions by calculating the number of data, frame by frame, that are to be transmitted for every real-time transmission to fulfill the delay constraint's conditions. It was also stated that the FLS has a very low computational complexity and that the channel status is not accounted for.

Reference [22] elaborated that the requirements are evaluated by the FLS scheduler right from the start of every single frame for all queues in the LTE transmission. Hence, to evaluate these requirements, the volume of data, $u_i(k)$, is computed using a control law [22][32]. The control law that is used to compute $u_i(k)$ provides a delay boundary to the sent frames. This is further elaborated in (4):

$$u_i(k) = h_i(k) * q_i(k) \tag{4}$$

where $h_i(k)$ signifies the pulse response that is passed in the course of a time-invariant filter throughout the signal at the present queue length, i, which is represented by $q_i(k)$. The time-invariant linear filter uses the discrete time convolution function that is represented by the symbol '*'.

In order to guarantee the queuing delays and also the system stability, Bounded Input Bounded Output (BIBO) is fully utilized in the design to make sure that the resources are not assigned to an infinite bandwidth by the FLS scheduler [22]. The FLS scheme has advantages in guaranteeing bounded delays to real-time flows at the same time as offering a minimal PLR [33]. In short, the FLS computes the amount of data to be transmitted.

B. Channel Aware and QoS Unaware

By utilizing the CQI feedbacks from the UEs that are being periodically delivered to the eNB through the use of ad hoc control messages, the channel's quality can be approximated by the scheduling scheme by every UE. Thus, making it possible to foresee the reachable amount of maximum throughput [8].

1) Proportional Fairness (PF)

The advantage of the PF scheduler is that it distributes resources fairly amongst its users [34]. This is achieved through the allocation of similar proportions of resources, also known as time slots, to every single user, and at the same time, observing the channel's quality and previous user's throughput, in order to give priority to better channel states [23][35].

Besides guaranteeing fairness, the PF scheduler makes full use of the overall network throughput and also the spectral efficiency [8][16]. The PF metric, which is given as $m_{i,k}^{PF}$, is expressed by (5):

$$m_{i,k}^{PF} = \frac{d_k^i(t)}{\bar{R}^i(t-1)} \tag{5}$$

where $d_k^i(t)$ is the achieved data rate and $\bar{R}^i(t-1)$ is the attainable throughput that is anticipated at the t^{th} time transmission interval (TTI) for the i^{th} user based on the total bandwidth and the respective k^{th} resource block (RB) [8].

The preceding average throughput acts as a weighting factor so that the users that are in bad conditions will be served within an assured period. The aim of the PF scheduler is to maximize the throughput value based on certain LTE system constraints. However, the drawback of this scheme is that it does not prioritize urgent traffic.

2) Maximum Throughput (MT)

As its name states, the MT strategy aims to utilize the overall throughput fully. This is performed by allocating each RB to the user that can attain the highest throughput value in that particular TTI [36][37]. However, the downside of MT is that it is unable to provide fairness in the sharing of resources [8]. The metric for MT is given by (6):

$$m_{i,k}^{MT} = d_k^i(t) \tag{6}$$

where $d_k^i(t)$ has the same meaning as in the PF metric. Furthermore, to obtain the throughput value that is achieved at i, the channel-capacity, also known as the Shannon mathematical expression, is used. The expression is given in (7):

$$d_k^i(t) = log\,[1 + SINR_k^i(t)] \tag{7}$$

C. Architecture of Two-Level Scheduling Algorithm

1) Upper Level Scheduler: FLS

The two-level scheduling algorithm involves priority in [38], which offers strict boundary for delays in real-time transmission [39]. In this two-level scheduling scheme's upper level, resource computation is performed by the FLS, which was designed using a theory called linear discrete-time control. It is assigned to evaluate the allocation of data using a closed control loop process [22][32]. Also, it is a channel-aware scheme that takes precautions regarding the performance of the data sources and the durable limits for delays [22].

2) Lower Level Scheduler

The lower level is functioned to assign RBs to real-time application traffic. Based on the structural design, the lower level interacts mutually with the upper level scheduler in correspondence to assigning radio resources actively. Traffic that passes through the lower level scheduler is real-time traffic only, which includes only video and VoIP transmissions.

For the lower level, there are two existing tests, PF and MT. The MT technique allocates RBs based on the data with higher priority in order to fully utilize the system throughput. PF is a scheduling method that is applicable for RB scheduling as it can provide fairness measures, but it is less efficient for real-time clients [32][33].

In short, at the lower level, RBs are first allocated to users that have a better channel quality, and then the remaining RBs are allocated to other users.

3) Details of Existing Two-Level Scheduler

In this existing method, which was proposed by [32], the author used the two-level scheduling approach whereby FLS is applied at the upper layer while the lower layer uses PF for real-time traffic (video and VoIP). The non-real-time traffic, which are the BE packets, are scheduled using the PF scheme.

In [32], the author stated the implementation of MT scheduling algorithm to be employed at the lower level to maximize the throughput [36]. MT is able to make full use of the throughput by assigning RBs to UE which has the possibility of attaining the highest value for throughput in the recent TTI. The MT metric, $m_{i,k}^{MT}$, is given as in (6) [8]. However, no experimental findings regarding FLS-MT were provided in the paper [32].

III. The Proposed Schemes

This scheme, known as the two-level scheduling architecture (Section 2.3), is comprised of two levels: an upper layer as well as a lower layer scheduler [22]. These two levels provide services supporting both real-time and non-real-time transmission flows [32]. This section elaborates on the enhanced scheduling architecture at the lower level of the two-level scheduling algorithm, where the RBs are scheduled based on TTI, after data has been computed by the FLS scheme at the upper level. The complete proposed scheduling architecture can be seen in Figure 3.

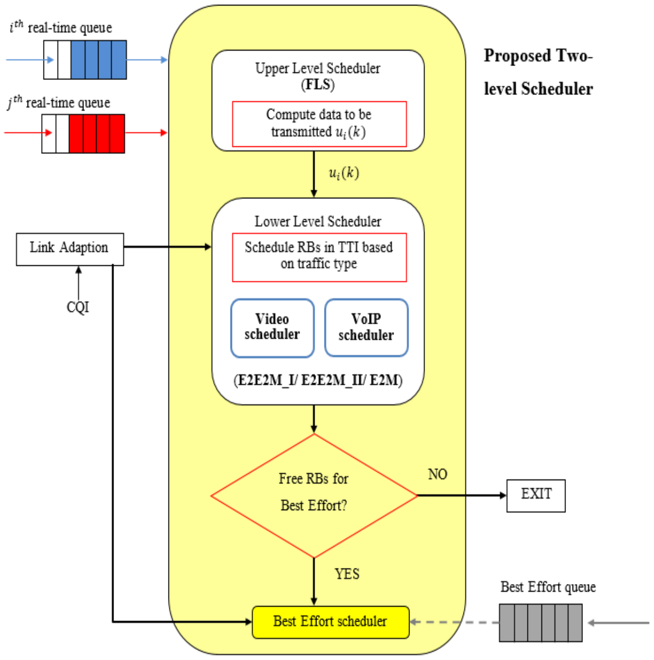

Fig. 3. Proposed Two-Level Scheduling Architecture

The lower layer assigns RBs based on the constraints by the FLS, which is at the upper layer, for every TTI. Based on our previous work in [40], we derived the packet scheduling metric without having a framework to compute the amount of data to be transmitted. By computing the amount of data, the packet scheduling will be more stable. Also, an efficient scheduling scheme with good QoS performance for a particular traffic shall provide more focus to the traffic needs. Therefore, we propose scheduling schemes which distinguishes between real-time and non-real-time traffic among the transmitted packets. The proposed architecture is able to increase system efficiency and provide better QoS.

1) The Enhanced Frame Level Scheduler (FE2M)

FE2M is a result based on the combination of three packet scheduling schemes, namely FLS, MT and enhanced EXP Rule with MT (E2M) Rule. The real-time traffic are directed towards the FLS scheme, where real-time packets are subjected to a control law to obtain a controlled amount of computed data to be transmitted before being subjected towards the packet scheduler, E2M Rule. E2M Rule is derived from EXP Rule and a small part of mEXP Rule [40].

With reference to [8], the EXP Rule's metric computation is given as shown in (8), where the fixed parameter values are given as $\alpha_i \in [\frac{5}{(0.99\,\tau_i)}, \frac{10}{(0.99\,\tau_i)}]$, $b_i = \frac{1}{E[\Gamma^i]}$, and $c = 1$.

$$m_{i,k}^{EXP\ Rule} = b_i\ exp\ \left(\frac{\alpha_i D_{HOL,i}}{c + \sqrt{\left(\frac{1}{N_{RT}}\right)\sum_j D_{HOL,j}}}\right) w_{i,k}^{PF} \qquad (8)$$

where N_{RT} represents the amount of real-time traffic and $D_{HOL,i}$ is the HOL delay for the i^{th} user for the k^{th} RB. The weight, $w_{i,k}^{PF}$, given in (9), is the channel's weighted spectral efficiency based on the PF scheduling algorithm [8]:

$$w_{i,k}^{PF} = \frac{d_k^i(t)}{\bar{R}^i(t-1)} \qquad (9)$$

where $d_k^i(t)$ is the available data rate, as described in (7), and $\bar{R}^i(t-1)$ represents the average data rate estimation.

The modified EXP (mEXP) Rule is applied at the lower level of the scheduler [40][41]. Its mathematical computation can be expressed by (10), where the metric for the modified EXP Rule is given as $m_{i,k}^{mEXP\ Rule}$ [8][41][42]:

$$m_{i,k}^{mEXP\ Rule} = exp\ \left(\beta_i D_{HOL,i}\right) w_{i,k}^{PF} \qquad (10)$$

where the equation for β_i is somewhat similar to that for M-LWDF's β_i, which did not have several optimal parameters. Hence, based on [41], the modified version is given as (11).

$$\beta_i = -\ ln\left(\frac{\delta_i}{0.99(\tau_i)}\right) \qquad (11)$$

where δ_i is the acceptable weighted probability of the users dropping the packet due to packet deadline expiration. The value of δ_i is given as 6, based on the optimal parameter that was suggested in [22]. The probability value for packet loss has been increased by replacing log with the ln function. This is so that the chance of losing a packet is smaller. The weight of the metric determines the priority of the packet, thus as the probability of losing a packet increases so does the weight. Hence, increasing the packets priority to be served. This scheduling scheme is a deadline-based scheduler [43][44].

Based on our research, the weight is substituted with a heavier weight, known as MT, which maximizes the throughput value in the current TTI. Hence, (12) shows the mathematical computation of our E2M Rule's metric, $m_{i,k}^{E2M\ Rule}$:

$$m_{i,k}^{E2M\ Rule} = exp\frac{\beta_i D_{HOL,i}}{c + \sqrt{\left(\frac{1}{N_{RT}}\right)\sum_j D_{HOL,j}}} w_{i,k}^{MT} \qquad (12)$$

where $w_{i,k}^{MT}$ is the weight using the MT scheme. The details are elaborated in (6).

In this scheduling architecture, real-time traffic (video and VoIP) passes through the FLS scheme and the E2M Rule, while non-real-time traffic (BE) pass through the PF algorithm. The PF designates resources fairly amongst its users that have good channel condition at the moment.

2) Details of the Proposed Scheduler (FE2E2M)

In this research, we separate the video, VoIP, and BE to pass through different schedulers. Thus, two packet scheduling schemes which differ at the lower level of the proposed scheduling framework are proposed. The scheduler utilizes the following scheduling schemes for different traffic flows.

<u>General description of FE2E2M</u>

The FE2E2M is where the upper layer consists of the FLS, which computes the amount of information that is to be transmitted by video and VoIP traffic only.

Based on Table II, it shows the scheduling scheme that is assigned for different traffic flows. Video traffic uses our enhanced scheduler, the E2M Rule, and VoIP traffic uses the Exaggerated Earliest Dateline First (E2DF)-MT.

TABLE II
SEPARATE TRAFFIC FLOW SCHEDULERS

Type of Traffic	Scheduling Scheme
Video	E2M Rule
VoIP	E2DF-MT

In [45], E2DF is an enhanced version of the Earliest Dateline First (EDF) [8][43][46] scheduler, which aims to avoid the packet expiration deadline. The priority of the EDF metric is calculated based on the HOL and the highest delay. The equation is given as (13):

$$m_i^{EDF} = \frac{1}{(D_{HOL,i} - \tau_i)} \quad (13)$$

where τ_i is the i^{th} user's delay threshold.

In [43], the author exaggerated the metric value in order to increase the weight of the metric. This, increases the metric's value and priority. The E2DF expression is given as (14):

$$m_{i,k}^{E2DF} = exp(\frac{1}{(D_{HOL,i} - \tau_i)}) \frac{d_k^i(t)}{\overline{R^i}(t-1)} \quad (14)$$

In this test, we replaced the weight by using MT instead of PF. Therefore, the equation is given as (15):

$$m_{i,k}^{E2DF} = exp(\frac{1}{(D_{HOL,i} - \tau_i)})\, w_{i,k}^{MT} \quad (15)$$

Hence, there is a better chance that a packet with a closer deadline expiration will be served first, thus reducing the PLR.

a) Best Effort Scheduling Scheme Using PF

Based on the details of the proposed scheduler, the type of scheduling scheme that is used for video and VoIP traffic has been stated. For the BE packets in non-real time transmissions, we performed tests using PF, which was proposed in [23] and [33].

Thus, the FE2E2M that applies PF as it's BE scheduling method is named FE2E2M_II.

b) Best Effort Scheduling Scheme Using MT

We then tested the use of MT in order to utilize the BE traffic's throughput fully despite the fact that it does not provide fairness in resource sharing amongst its users [8].

Therefore, the FE2E2M that applies MT as it's BE scheduling method is named FE2E2M_I.

IV. PERFORMANCE EVALUATION

1) *Simulation Parameters*

The simulator used to perform simulations is the 5th version of the LTE-Simulator (LTE-Sim) [47]. The computation of metrics governs the scheduling of packets. Therefore, prioritized metrics are metrics that comprises a greater significance in terms of its value are hence prior to be scheduled followed by the next highest metric computed and so on [48]. Until every listed packet is scheduled, the scheduling process continues [23]. Several simulation parameters have been fixed for the simulation tests performed. An average was attained based on the five sets of conducted simulation results. Table III briefly describes the simulation parameters that were used.

TABLE III
SIMULATION PARAMETERS

Parameters	Values
PHY	OFDMA
Bandwidth/Frame Length	5 MHz / 10 ms
Frame Structure	TDD with uplink-downlink configuration number 1, and periodicity of 5 ms.
Simulation Scenario	Single cell with an interference
Modulation	QAM, 4-QAM, 16-QAM
Simulation Duration	60 s
Traffic Models	Real-time: Video and VoIP; Non-real-time: Best Effort
Mobility	eNodeB: Constant Position; UE: Random Direction
Speed	3 km/h
Number of UEs	5–70
UE Interval	5
Traffic Type	A video, a VoIP and a BE
Video Data Rate/Traffic	128 kbps/ foreman_H264 data
VoIP Traffic	Generated by G.729 codec, an audio data compression scheme

The video traffic uses the QCI value 2 of which the packet delay budget is given as 150 ms [49][50]. As for the VoIP traffics, it is generated by G.729 codec, an audio data compression scheme [47]. The algorithmic delay for G.729 codec is 15 ms per frame. G.729 codec has a look-ahead delay of 5 ms [51][52]. Hence, the VoIP traffics utilizes QCI value 1 whereby the packet delay budget is 100 ms.

2) *Results and Discussion*

Based on the simulations that were executed in the LTE-Sim, the performances are evaluated based on several factors, which include delay, spectral efficiency, PLR, and throughput. This was performed for all traffic, which included video, VoIP, and BE traffic. It was stated in [53] that BE targets web services, hence does not guarantee QoS. The delay for BE packets are constantly fixed at 1 ms as it is the maximum delay threshold, and therefore it is not further discussed. Results are obtained based on an average of five sets of simulations.

In contrast to the existing two-level scheduling architecture aforementioned in Section 2.3.3, even though the PF is able to provide fairness and maximize networks throughput [32], it is only focused on BE services. Our focus is mainly on real-time services hence, based on our experimental test, we found that the FLS-MT has a much better performance on its throughput rate while its fairness measures is approximately the same with the FLS-PF. The FLS-MT has a much more efficient use of bandwidth compared to the FLS-PF, and it contributes to a lesser PLR as well as delay for video traffic. Thus, results of the proposed schemes are compared and discussed based on the mentioned scheme the FLS-MT [32] instead of the FLS-PF [22][32].

There are three graphs that illustrate the PLR for BE (Figure 4), video (Figure 5), and VoIP (Figure 6) traffic. In Figure 4, it can be clearly seen that the PLR is constant for the FLS-MT, FE2E2M_II, and FE2M at approximately 2.8 to 3.9%. As for the FE2E2M_I, the PLR is quite high for 5 to 20 users, at about 3 to 7%. However, the FE2E2M_I shows an improvement in PLR of less than 3% from the 25th user onwards. The lowest PLR occurs at the 40th user, where it is 0.86%. The higher the number of users, the lower the packets' actual delay. This is due to the PF function which does not guarantee real-time transmission flows, hence allocates more bandwidth for BE flows.

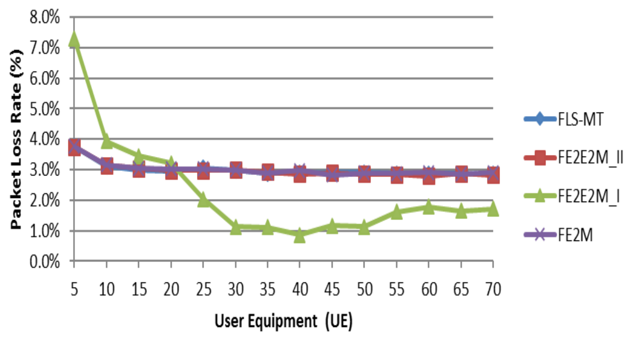

Fig. 4. PLR for BE Transmission

In Figure 5, the PLR for video is low for all schemes for 5 to 35 users, being less than 0.6%. Conversely, there is a sudden increase in PLR for all schemes from 40 to 70 users. The FE2E2M_I and the FE2E2M_II have higher PLRs compared to the FLS-MT and the FE2M. The change ranges between approximately 1 and 3%. When the number of users decreases, the packets' actual delay is very low. This is due to the metric value, which is pushed up by the exponential function, which exaggerates the metric value. Thus, this leads to more resources being allocated to video traffic.

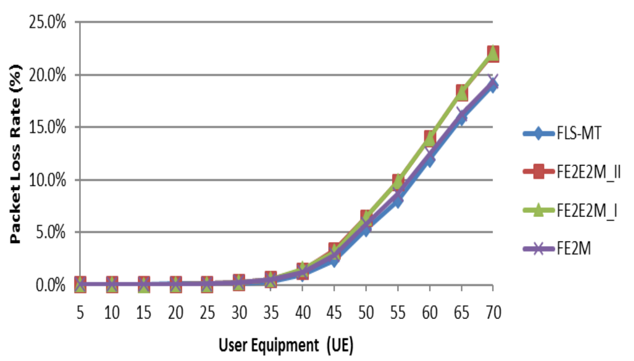

Fig. 5. PLR for Video Transmission

For VoIP traffic, Figure 6 shows that the FE2E2M _I and the FE2E2M have very good performance in terms of PLR compared to the existing FLS-MT and also the FE2M. The PLR for the FE2E2M_I ranges from 0 to 0.001%, while for the FE2E2M_II it ranges from 0 to 0.007%. The performance of the FE2E2M_II is approximately 8%, which is much better than the FLS-MT and represents an improvement of approximately 12% compared to our previous scheme, FE2M. The reason for this improvement is similar to that described in the above paragraph based on Figure 6. The FLS can guarantee the packets' bounded delays for real-time transmissions, thus providing the lowest PLR [34].

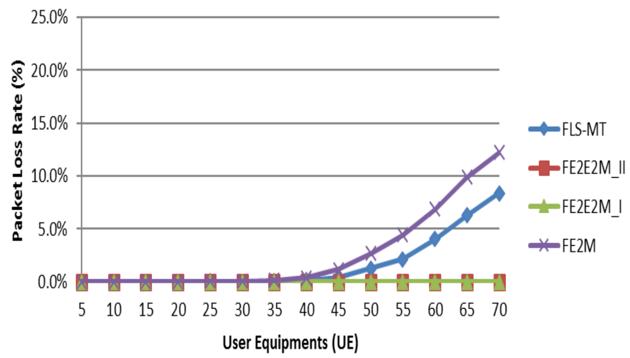

Fig. 6. PLR for VoIP Transmission

In terms of throughput for BE, Figure 7 shows that the FE2E2M_I has a higher throughput value compared to the other schemes that were tested. The peak throughput is at 10 users, where the rate is 9.96 Mbps. The percentage improvement between the performances of the FLS-MT and the FE2E2M_I is estimated to be 9 to 45%. Meanwhile the other three schemes, FLS-MT, FE2M, and FE2E2M_II, have peak throughputs of around 2 to 6 Mbps. Based on the figure, all schemes have peak throughput values at 10 users. The rate then steadily decreases from 10 user's right up to 70 users. The decrease in throughput as the number of users increases is due to the low availability of bandwidth that is freely left over for the BE flows.

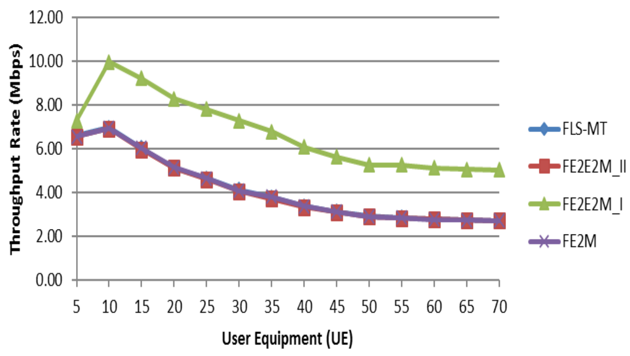

Fig. 7. Throughput for BE Transmission

The throughput rate for video traffic can be seen in Figure 8, where the rate increases as the number of users increases. The FLS-MT and the FE2M have higher throughput compared to the FE2E2M_I and the FE2E2M_II. The percentage change is in the range of 0 to 5%, which is a rather small change. Based on the results obtained for all schemes, for 5 to 45 users the throughput is between 0.4 and <3 Mbps. For 50 to 70 users, the throughput ranges above 4 Mbps.

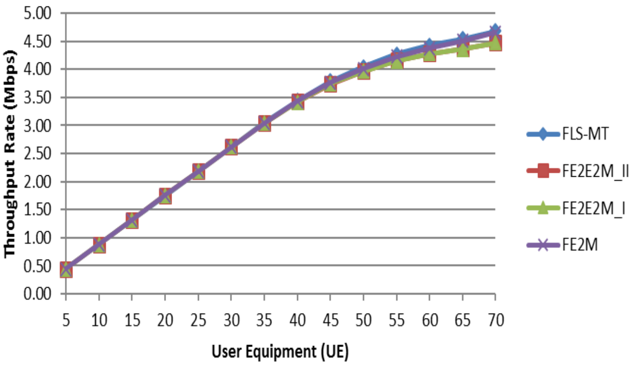

Fig. 8. Throughput for Video Transmission

Figure 9 shows the throughput results for VoIP traffic. As can be seen from the graph, the throughput values keep increasing as the UEs increase. Our FE2E2M_II and FE2E2M schedulers perform much better compared to FLS-MT and our previous scheduler FE2M. This is especially so for between 55 and 70 users, where the highest throughput value is achieved by FE2E2M_II for 70 users. In addition to that, the improvement obtained by FE2E2M_II is approximately 10% when compared to the existing FLS-MT and less than 1% when compared to FE2E2M_I. As mentioned, the throughput values for both video and VoIP increase as the number of users increases because there is a smaller PLR and hence more packets can be sent out.

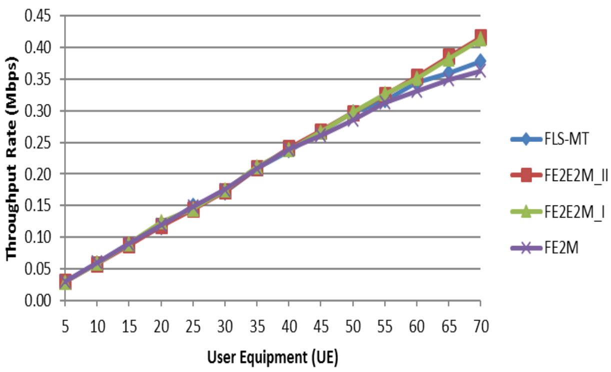

Fig. 9. Throughput for VoIP Transmission

Figure 10 illustrates the delay for video traffic. There is an improvement of approximately 0.1 to 5.64% in the delay from the 50th user to the 70th user for the FE2E2M_II, and an enhancement of approximately 6% for the FE2E2M_I compared to the existing FLS-MT. The FE2M performs much better than the FLS-MT from the turning point of 40 users onwards. The change is in the range of approximately 0 to 11%.

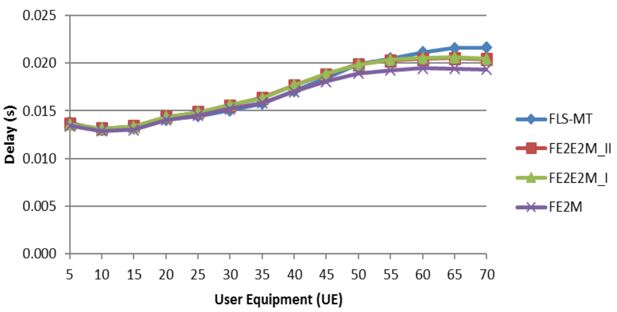

Fig. 10. Delay for Video Transmission

As illustrated in Figure 11, the FE2M shows a better performance in terms of delay than the FLS-MT for 50 to 70 users. Its maximum improvement is a change of approximately 18%. Moreover, the FE2E2M_I and the FE2E2M_II outperform both the FLS-MT and the FE2M. This is especially so for FE2E2M_II at 30 to 65 users, where the improvement is from 0.16 to 3.09% in comparison to the FE2E2M_I. In contrast to FLS-MT, the improvement in the delay is between 8 and 53%, which is also twice as good as FLS-MT. The delay with the FE2E2M_I and the FE2E2M_II is approximately constant for video transmissions as users share out the bandwidth fairly and make full use of its resources.

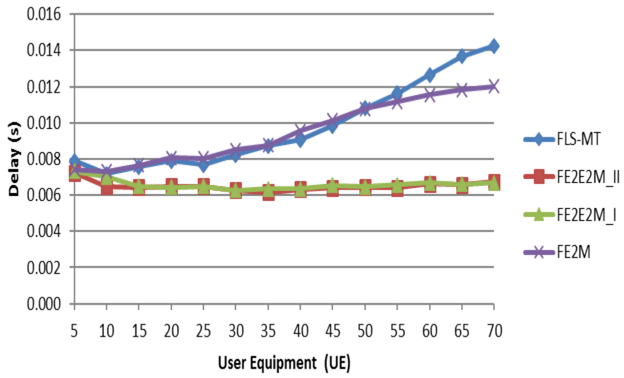

Fig. 11. Delay for VoIP Transmission

Lastly, Figure 12 shows the results for the simulations' spectral efficiencies. As can be seen very clearly in the figure, the FE2E2M_I has the highest spectral efficiency compared to the other three scheduling schemes. FE2E2M_II, FLS-MT, and FE2M have values of spectral efficiency that are constant, on average, with minimal changes of approximately 0 to 2.4%. In contrast to the FLS-MT, the FE2E2M_I shows an improvement ranging from 9 to 46%, where the greatest improvement in performance is at the 30th user and is 45.64%. Based on the better performance of FE2E2M_I on the several QoS factors elaborated in this section, which include the improvement of PLR, delay, and throughput for real-time services, there is an enhancement on the bandwidth's efficiency because resources are allocated efficiently amongst its users.

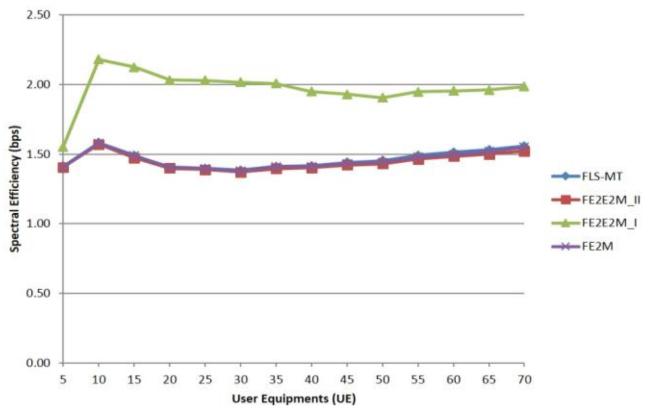

Fig. 12. Spectral Efficiency

V. Conclusion

Based on the gathered results and our observations, it can be concluded that the FE2E2M_I shows better performance compared to the FLS-MT in terms of PLR, throughput, and delay for BE as well as VoIP. The PLR for VoIP is very good as the test results show that it comprises a minimal amount of packet loss of almost 0%. The throughput for BE in the FE2E2M_I increases by 9 to 46% in contrast to the FLS-MT. Regarding the VoIP delay, the delay obtained by the FE2E2M_I is approximately 2.5 to 114% smaller.

In regards to the video traffic that was scheduled in the FE2E2M_I, there is a slight improvement in the delay. However, this was not the case for the throughput, but the variance is not very significant as the throughput values for this scheduling scheme vary slightly between 0 and <5% in comparison to the FLS-MT. Also, the performance is good for spectral efficiency as it overcame the other scheduling schemes that were tested. The percentage improvement ranges between 9 and 46%. As for the delay in video, the FE2E2M_I has an enhancement of up to <6%, especially for 50 to 70 users.

The results for the FE2E2M_II are somewhat similar to those for the FE2E2M_I. The dissimilarity is that it has better features for VoIP based on factors which include PLR, delay, and throughput when compared to the FLS-MT. The throughput of the FE2E2M_II is 0.14 to 9.28% higher than that of the FLS-MT scheduler. Regarding the delay in VoIP, the FE2E2M_II shows improvements that range between approximately 8 and 53%. Similarly to the FE2E2M_I, the FE2E2M_II has an almost perfect PLR. The lowest PLR value is 0.001%. Regarding the spectral efficiency, there is a slight improvement, with the highest improvement being up to 2.33%.

The number of users range from 5 to 70 with an interval of 5. Therefore, we are able to compute an average QoS performance based on the number of classes. Table IV (BE), V (video) and VI (VoIP) shows the average result for the given QoS factors on each scheduling scheme. Thus, an average is obtained by computing the sum of the results obtained by each class divided by the total number of classes, in this case 14 classes.

Table VII gives an overview of the results that were obtained based on the benchmark FLS-MT. The symbols ‘I’, ‘IS’ and ‘D’ stands for improvement, improve slightly, and degraded performance, respectively.

Hence, it can be concluded that the performance of our proposed scheduler the FE2E2M_I is much better than that of the existing scheduler FLS-MT as well as the FE2M and the FE2E2M_II. As discussed above, it can be clearly seen that the FE2E2M_I performs well and can satisfy the QoS requirement factors, which include delay, PLR, throughput, and spectral efficiency.

TABLE IV
AVERAGE RESULTS FOR BEST EFFORT TRAFFIC

QoS Factors	Type of Schemes				
	FLS-PF	FLS-MT	FE2E2M_I	FE2E2M_II	FE2M
PLR (%)	3.00	3.00	3.00	2.99	3.01
Through-put (Mbps)	3.931979	4.132932	6.724284	4.098751	4.127878

TABLE V
AVERAGE RESULTS FOR VIDEO TRAFFIC

QoS Factors	Type of Schemes				
	FLS-PF	FLS-MT	FE2E2M_I	FE2E2M_II	FE2M
Delay (s)	0.021256	0.017111	0.017113	0.017093	0.016467
PLR (%)	6.58	4.61	5.46	5.45	4.83
Through-put (Mbps)	2.642624	2.959053	2.900123	2.900765	2.941961

TABLE VI
AVERAGE RESULTS FOR VOIP TRAFFIC

QoS Factors	Type of Schemes				
	FLS-PF	FLS-MT	FE2E2M_I	FE2E2M_II	FE2M
Delay (s)	0.008421	0.009791	0.006589	0.006505	0.009483
PLR (%)	0.03	1.62	0.00	0.00	2.71
Through-put (Mbps)	0.223665	0.216159	0.222078	0.221978	0.212354

TABLE VII
EXPERIMENTAL RESULTS OVERVIEW

Schemes		QoS Factors			
		Delay	PLR	Throughput	Spectral Efficiency
FE2E2M_I	VoIP	I	I	I	I
	Video	IS	IS	D	
	BE	-	I	I	
FE2E2M_II	VoIP	I	I	I	D
	Video	IS	IS	IS	
	BE	-	IS	D	
FE2M	VoIP	IS	D	IS	D
	Video	IS	D	D	
	BE	-	IS	D	

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