# Low Complexity in Exaggerated Earliest Deadline First Approach for Channel and QoS-aware Scheduler

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***Abstract***—Long Term Evolution (LTE) is a Quality of Service (QoS) provisioning wireless network for today's technology and as well as for future demands. There is a high demand for better network performance over LTE network, either for real-time or non-real-time traffic. Specifically, the existing scheduling algorithms for real-time application, Exponential/Proportional Fair (EXP/PF), Proportional Fair (PF), and Modified-Largest Weighted Delay First (M-LWDF) have not fully optimized in LTE network. Hence, this paper aims to deliver new scheduling algorithm in the LTE network which overcomes several QoS and channel concerns. Several algorithms were studied, tested and compared which includes EXP/PF, PF, and M-LWDF, which are the popular scheduling algorithms for real-time application in today's deployment. A typical LTE network is simulated and several experiments were conducted. Extensive simulation results showed that our proposed scheduling algorithm, Exaggerated Earliest Deadline First (E2DF), has outperformed the three existing scheduling algorithms. The proposed algorithm is a LTE compliance module and it able to provide great performance improvement as compared to the other algorithms for real-time application.

***Index Terms***—LTE, scheduling algorithm, quality of service, 4G, wireless network.

## I. Introduction

Long Term Evolution (LTE) is the Radio Access Network (RAN) which is called Evolved Packet System (EPS). The network components of LTE, includes User Equipment (UE), Evolved UMTS Terrestrial Radio Access Network (EUTRAN) and Evolved Packet Core (EPC) or System Architecture Evolution (SAE). LTE is designed to be a full IP based network with providing Quality of Service (QoS) and security support. Fig. 1 shows the basic architecture components of LTE, which consists of Enhanced node Bs (eNodeBs), and Mobility Management Entities (MMEs) and Serving Gateways (S-GW) at the EPC. The eNBs interconnect through an interface called X2 interface, while they are connected to entities at the core (MMEs and S-GWs) using the S1 interface [1].

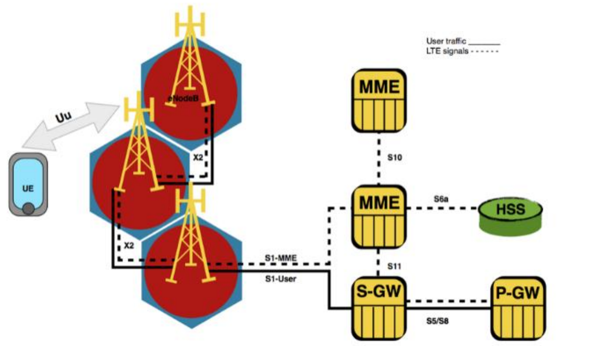


Fig. 1. Architecture of LTE network

The LTE architecture depends on network configuration which is simpler than its predecessor. E-UTRAN, radio access network considerations and decisions are handled by eNodeB, while relevant considerations for the core network are processed at the EPC. Tasks are divided as follow; None Access Stratum (NAS) is handled by entities in core, and Access Stratum (AS) is handled by eNodeB. eNodeB also handles radio access control, scheduling, admission control, mobility control and handover. On the other hands, MME, S-GW and Packet Data Network Gateway (P-GW) have the functionalities such as mobile anchoring, NAS security, mobility, IP address allocation and packet filtering. EPC provides connection with other radio networks and Internet [2].

The main focus of this paper is the scheduling algorithm studies for real-time applications in LTE network. We studied the performance metrics of the existing scheduling algorithms; Proportional Fair (PF), Modified-Largest Weighted Delay First (MLWDF) and Exponential/Proportional Fair (EXP-PF) [3]-[5] in LTE. It is observed that there is a possibility of an enhancement. With our proposed enhanced scheduling algorithm, Exaggerated Earliest Deadline First (E2DF), the performance of LTE network in terms of packet loss ratio and average goodput have been improved while maintaining the latency performance for real-time applications. Particularly, the contributions of this work can be summarized as:

1). This study discovers that the network performance in LTE network could be further improved by adding channel information to the existing Earliest Deadline First (EDF) scheduling algorithm. Meanwhile, the enhanced

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scheduling algorithm considers the important component of its predecessor, EDF, and maintains this component as a parameter during the weighting calculation.

2). This work proposes and verifies new enhanced scheduling algorithm that works in point-to-multipoint (PMP) operation mode of any LTE compliance systems. The proposed scheduling algorithm performs better than the well-known LTE scheduling algorithms for real- time application (MLWDF and EXP-PF).

In Section 2, we review the QoS framework for LTE. Section 3 addresses the previous works in scheduling algorithms used in broadband wireless access, such as LTE and WiMAX. Section 4 elaborates our proposed scheduling algorithm and its strengths over the previous algorithms. Simulation scenario and experiments in LTE are described in Section 5, as well as the experimental evaluation and discussion of our proposed scheduling algorithm. Section 6 concludes this study.

## II. QOS IN LTE NETWORK

The QoS framework of LTE is designed to deliver end-to-end QoS support with its own requirements. Flows in LTE are mapped by QoS Class Identifier (QCI). This virtual bearer is to build an end-to-end QoS [2]. The most important component in the bearer is EPS bearer, which is a bidirectional virtual tunnel. It carries data from a mobile user to the packet gateway (P-GW) with a specific QoS. GTP-C and GTP-U are protocols to modify and configure all these QoS parameters in an EPS bearer. Each bearer has one or more service data flows, and each service data flow has one or more packet flows. For example, a video stream flow has two packet flows; one for audio and one for video. In LTE, all the packet flows in the same bearer will be treated with same QoS parameters [6].

EPS bearer is classified into two; Guaranteed Bit Rate (GBR) or non-GBR bearer. GBR is for long term average data rate which has been assigned to mobile [7]. GBR is suitable for real-time application. On the other hands, non-GBR is for non-real-time applications. The later classification is default or dedicated bearer. Each time, when a UE connects to LTE network, a non-GBR bearer is assigned to the UE. UE uses this bearer to connect to P-GW. Dedicated bearer is assigned based on demand, either non-GBR or GBR bearer. EPS bearer is further divided in 3 parts; radio bearer, S1 bearer and S5/S8, as in Fig. 2.

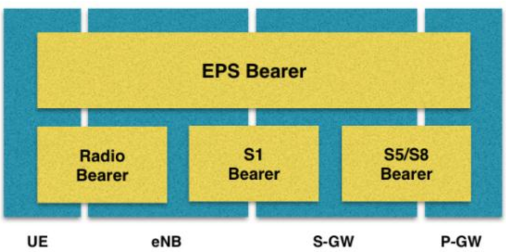


Fig. 2. LTE bearer division

The QoS attributes associated with LTE bearer are listed below [7]:

1) QCI: scalable value from 1 to 9, defined resource type, packet delay budget and packet error lost rate. See Table 1.

2) Allocation and retention priority (ARP): parameter used by admission control and overload control to establish/ release or modify bearer when is needed.

3) Maximum bit rate (MBR): the maximum bit rate for bearer, which should not be exceeded. It is only available for GBR bearers.

4) GBR: minimum traffic rate promises by a bearer.

5) Aggregate MBR (AMBR): the total amount of bit rates of non-GBR bearers. AMBRs values can be defined separately for uplink and downlink between ARP and terminal [6].

TABLE I: LTE GBR AND NON-GBR STANDARDIZED QCI CHARACTERISTICS

| QCI | Priority | Packet Delay Budget | Packet Error Lost Rate | Example Services |
|---|---|---|---|---|
| *1* | 2 | 100 ms | 10 | Conversational Voice |
| *2* | 4 | 150 ms | 10 | Conversational Video (Live Stream) |
| *3* | 3 | 50 ms | 10 | Realtime Gaming |
| *4* | 5 | 300 ms | 10 | Non-Conversational Video (Buffer Stream) |
| *5* | 1 | 100 ms | 10 | IMS Signaling |
| *6* | 6 | 300 ms | 10 | Video Buffer Streaming |
| *7* | 7 | 100 ms | 10 | Voice and video Live streaming |
| *8* | 8 | 300 ms | 10 | Email, FTP, P2P file |
| *9* | 9 | | | video |

## III. SCHEDULING ALGORITHMS IN ADVANCE WIRELESS NETWORKS

Scheduling algorithms in LTE are divided based on their awareness in channel condition and QoS parameter. With this classification, scheduling algorithms are categorized as Channel Non-aware, Channel Aware QoS Non-aware and Channel Aware QoS Aware.

### A. *Channel Non-Aware Schedulers*

Round Robin (RR) is a basic and least complicated scheduling algorithm, which serves each queue in order. Pointer stops at one queue each time, if the queue is not empty. For each non-empty queue, it will dequeue one packet and pointer moves to the next queue [8], [9]. RR is good for equal sized packet with the same QoS parameters. But, when the packet size is different, for instance, in MPEG video application's where packet size is varied, this phenomenon causes RR scheduling algorithm losses its fairness. Weighted Round Robin (WRR) operates on the same basis as RR. However, WRR assigns a weight value to each queue in order to achieve more fairness. Packets are dequeued from each queue base on the weight value assigned to each queue. Sum of the weights of all queues is related to available shared system bandwidth [8]. Since there is a weight value assigned to each queue, it enables prioritization among queues, which makes WRR better than RR. Although WRR fixed the priority problem, different packet size will lead WRR to lose its fairness.

Deficit Round Robin (DRR) scheduling algorithm assigns deficit counter to each queue, which initiates with a value of quantum. In this scheme, each queue dequeues packets according to quantum credit assigned. Fairness of queue for different packet size is achieved through this approach. When pointer of DRR stops at a queue and its deficit value is bigger than zero, packets are served in the queue same amount as a value of the deficit. After deficit value reaches zero, dequeuing process will stop and pointer moves to the next queue. If queue is not empty, deficit counter increases by one quantum on every visit of the queue. If the queue is completely served and reaches zero packets, deficit value will be reset to zero since giving credit without being utilized [8].

Weighted Fair Queue (WFQ) is yet another packet scheduling and queue management technique; it is derived from Generalized Processor Sharing (GPS) scheduling, and it is an alternative way to reduce starvation possibility of queues. WFQ has different queue for each data flow or services with First in First Out (FIFO) technique. It allows guaranteed bandwidth for different services. The weight parameter can be derived by different type of parameters such as packet delay, average data rate, queue length, etc. WFQ has been evaluated in [4] and an enhanced version of WFQ in WiMAX whereby the minimum reserved rate (MRR) is used as weight parameter [10].

In [11], authors examining feasibility of semi persistent scheduling (SPS) for voice over IP (VoIP) by random access and evaluate its performance in terms of throughput of random access and traffic channels, and random access delay. Two scheduling mechanisms were proposed, Dynamic Scheduling and SPS. The first algorithm allocates radio resources dynamically based on each terminal's buffer status and radio channel state information. The later algorithm allocates an uplink traffic channel periodically without any additional control message during a traffic burst. SPS also utilize the allocated traffic channel that would be implicitly released when a certain number of empty transmission slots are found on the allocated traffic channel to prevent it from being released due to some packets being lost over the radio channel.

### B. Channel Aware QoS Non-Aware Schedulers

In LTE, the following three feedback mechanisms are specified its standards:

1) Wideband feedback - Only one Channel Quality Indicator (CQI) value is reported by each UE for the entire bandwidth.

2) UE-selected subband feedback - Each UE sends the indices of its best subbands and only one average CQI value for all the selected subbands

3) subband-level feedback - One CQI value is reported by each UE for every subband.

With this feedback, several proposals were presented. Among these are closed-form expressions for the throughput of the PF and greedy schedulers for the UE-selected subband feedback and subband-level feedback schemes [12]. The analysis quantifies the joint effects of the following three critical components on the overall throughput which are scheduler, multiple-antenna mode, and the quantized CQI feedback scheme. The idea is elaborated as different flow have different QoS variant resulted from various CQI variant and results. This is also include the types of QoS parameter such as, in a video flow, it contains two different parameters; loose delay constraint for streaming video while the other required tight constraint delay for live video. [13] Rate Prediction which is the algorithm is to generate better accuracy CQI report with the uncertainty in the channel gain. The latter is Resource Assignment (RA) that based on the optimization problem gained from the rate prediction algorithm. It is handled by a suitable scheduling policy such as Exp Rule, Log Rule, EDF and etc.

In [14], a scheduling algorithm that is called Improved Frequency Diversity and Selectivity Scheduling (IFDSS) algorithm was introduced. IFDSS is channel state information (CSI) that consider with different delay profile which will result in a better CQI variance result.

Generally, PF scheduling algorithm aims to find a balance between resource fairness and spectral efficiency (effective channel utilization) [5]. Scheduling algorithm should guarantees minimum performance for all UEs, even for several UEs that are categorised as cell-edge users (cell-edge users are those users experiencing bad channel conditions). PF scheduling algorithm is channel-aware strategy, where CQI feedbacks are sent from UEs to eNB periodically. By using these signals, scheduler can estimate the channel quality for each connected UE. Hence, it can predict the maximum achievable throughput. In PF, users are prioritized by their channel quality and their average allocated rate. The PF is formulated as in (1) and (2).

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

$$m_{i,k}^{PF} = \frac{d_k^i(t)}{\overline{R^i}(t-1)} \tag{2}$$

where $d_k^i(t)$ is the expected data-rate for the *i-th* user at $t$ time on the *k-th* resource block and $\overline{R^i}(t-1)$ is past average throughput achieved by data flow of the *i-th* user when scheduled $m_{i,k}^{PF}$ is the metric value for PF algorithm.

### C. Channel Aware QoS Aware Schedulers

A cross-layer optimized video delivery system consists of modules for video application, cross-layer optimization, dynamic resource allocation and wireless delivery was introduced in [15]. This approach considers three design factors for each user (available channel rate of a user on a given resource block, application video packet delay constraints, historical average data rate of each user). The video application module performs video encoding by dynamically adapting to CQI of the RB feedback from the wireless delivery module. Cross-layer optimization module performs optimization to find the best MCS and encoder parameters based on radio resource allocation outcomes and video application characteristics. The resource allocation module uses the historical average data rate to prevent users from holding resources for too long.

In [16], authors proposed Capacity-Driven Resource Allocation (CRA), with the aim of improving joint system capacity of LTE in multiservice scenarios. The architecture of CRA is split into two parts, resource allocation and resource assignment. For Resource Allocation, it defines which flow will be scheduled and also determines the required data rate at current Transmission Time Interval (TTI) while for resource assignment defines which resources will be assigned to selected flows. Flows are ordered according to priority based on satisfaction level (priority to lists that are easier to satisfy). CRA also calculated required data rate the flow needs to transmit in current TTI. The last step is the selection (based on load imposed by each service) of the flows that will receive resource units (RU) in resource assignment part. As for the resource assignment part, it distributes the RUs fairly among selected flows. This part is executed in phases. In each phase, all the flows get one RU. However, the flow that will choose its RU first is the one that has the RU in better channel conditions among all other flows.

Since there is a requirement to delivers the packet within a certain deadline, MLWDF is an example of channel-aware scheduling algorithm which is a QoS-aware scheduling as well [7]. M-LWDF considers packet delay at each time slot and it serves the particular queue based on the head of line's (HOL) delay value of that queue. In M-LWDF, non-real time and real-time flows are treated differently; it uses PF algorithm for non-real time application and weighting metrics for real-time applications.

In each $t$ time slot, M-LWDF serves the $j$ queue that has the HOL delay value of the $j$ queue is maximal. This approach makes M-LWDF scheduling achieves optimal throughput. The weight of the metric, $m_{i,k}^{M-LWDF}$ is calculated in (3) and (4).

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

where $\propto_i$ is the weights of metric, $\delta_i$ is the probability that the packet is dropped due to deadline expiration and $T_i$ is the target delay, in other words, refers to last time when the *i-th* user was served. $D_{HOL,i}$ is the delay of head of line packet.

$$m_{i,k}^{M-LWDF} = \left(\propto_i \times D_{HOL,i}\right) \times \frac{d_k^i(t)}{\overline{R^i}(t-1)} \quad (4)$$

EXP-PF was first developed in multiplexed systems to support multimedia application. Even though, it is not designed for OFDMA in LTE, EXP-PF metric can be used as channel aware and QoS-aware scheduling algorithm [5]. EXP-PF uses both the characteristics of PF and exponential function to minimize the end-to-end delay [3]. EXP-PF is aware of a number of active flows for each user, and the delay of HOL's value. This extra information is used in calculating the weighting metrics. EXP-PF also uses PF for non-real time application and weighting metrics for real-time application. In other words, for non-real-time service (e.g. Best Effort), EXP-PF calculates the metric b0y using PF scheduling algorithm, and for real-time service, the weight is computed as in (5) and (6). However, EXP-PF attempts to guarantee reasonable fairness and goodput for both real-time and non-real-time applications.

$$m_{i,k}^{EXP/PF} = exp\left(\frac{\propto_i \times D_{HOL,i}}{1+\sqrt{\varphi}}\right) \times \frac{d_k^i(t)}{\overline{R^i}(t-1)} \quad (5)$$

$$\varphi = \frac{1}{N_{rt}} \sum_{i=1}^{N_{rt}} \left(\propto_i \times D_{HOL,i}\right) \quad (6)$$

where $N_{rt}$ is the number of active downlink real-time flows. $D_{HOL,i}$ is the delay of HOL packet for the *i-th* user and $\varphi$ is the mean of $\propto_i \times D_{HOL,i}$ for all active sessions and used as weight factor for EXP.

In conclusion, M-LWDF and EXP/PF scheduling algorithm that considers about the waiting time of the packets in the queue [17]. Packet Loss Rate (PLR) for both MLWDF and EXP-PF are also lesser due to the short stay time in the queue.

In LTE network, UEs are placed in filed based on a random position with different channel condition. User mobility is another important criterion in LTE network, so it is more reasonable to use 3 km/h speed to simulate movement of a user [11]. Among the scheduling algorithms, M-LWDF has the best performances among the algorithms for real-time multimedia application [17].

## IV. Exaggerated Earliest Dealine First (E2DF)

EDF is a dynamic scheduling strategy whereby the priority of the packet can be changed according to delay time of the packet [18]. The priority of the packet is inversely proportional to packet deadline. The highest priority belongs to the packet which its deadline is earliest. If two packets are having the same deadline, one of the packets will be chosen in random by EDF scheduling algorithm. In EDF, the priority can be calculated based on HOL and maximum delay. Priority is calculated in (7).

$$m_i^{EDF} = \left(\frac{1}{D_{HOL,i} - \max D}\right) \quad (7)$$

$m_i^{EDF}$ is the metric's priority value, $D_{HOL,i}$ is the delay of the packet located at the HOL and max $D$ is the maximum tolerated delay for a packet.

EDF alone is QoS aware scheduling, which is only aware of delay time. For better scheduling, E2DF is proposed. E2DF is designed based on the idea of having a channel aware and QoS aware scheduling algorithm. Like other existing scheduling algorithms implemented in LTE, packet flows are divided between real-time and non-real-time flows. In non-real-time flow, delay of the packet is not an important element; thus it is scheduled according to the channel condition. In contrast, real-time flow is scheduled with information from channel and QoS requirements. There are two separate weights in E2DF which influence the metric of priority. The first weight, $W_1$ is calculated similar to PF scheduling algorithm. It is based on the channel's condition. The latter weight, $W_2$ is calculated based on QoS requirement which considers of the delay of HOL. $W_1$ and $W_2$ are calculated in (8) and (9).

$$w_1 = \frac{d_k^i(t)}{\overline{R^i}(t-1)} \tag{8}$$

$$w_2 = exp\left(\frac{1}{D_{HOL,i} - \max D}\right) \tag{9}$$

where $d_k^i(t)$ is the expected data rate for the *i-th* user at *t* time on the *k-th* resource block, and is calculated based on (10) and $\overline{R^i}(t-1)$ is the past average throughput achieved by data flow of the *i-th* user, that has been scheduled. $D_{HOL,i}$ is the delay of HOL and *max D* is maximum tolerate delay for the packet.

$$d_k^i(t) = \log[1 + SINR_k^i(t)] \tag{10}$$

E2DF metric is calculated by multiplication of $W_1$ and $W_2$ as in (11), where $m_i^{E2DF}$ is a metric value for E2DF.

$$m_i^{E2DF} = \left(\frac{d_k^i(t)}{\overline{R^i}(t-1)}\right) \times exp\left(\frac{1}{D_{HOL,i} - \max D}\right) \tag{11}$$

When the differences of $D_{HOL,i}$ and max $D$ become smaller or in other words, the delay tolerance of packet is reaching the end, the exponential function exaggerates $W_1$ and $W_2$ increases the weight. This exaggeration increases the value of $m_i^{E2DF}$ and decreases the chance of packet missing the deadline and from being dropped from the queue.

## V. Simulation Results and Discussions

LTE-Sim simulator is used to simulate the network scenario with an eNodeB with radius of 1 kilometer (km) coverage. A number of UEs between 10 and 70 with an interval of 10 are randomly surrounding the eNodeB. In our previous study [17], 3 km/h of mobility is chosen to simulate a moderate speed for this study. Movement is based on a random walk model predefined in LTE-Sim.

The downlink bandwidth is configured at 10 MHz. Other simulation parameters are as in Table II. There are three types of traffic that have been simulated to reflect the end users' activities. Each UE receives 1 H.264 video flow that's been encoded at 128 kbps, 1 VoIP flow and 1 best effort flow with infinite buffer application. Downlink scheduling algorithms (E2DF, PF, M-LWDF and EXP-PF) are evaluated and compared. PLR, throughput and delay are the major network performance metrics that have been evaluated in this study.

TABLE II: Simulation Parameters

| **Simulation Parameters** | |
|---|---|
| ***PHY*** | OFDMA |
| ***Bandwidth*** | 10 MHz |
| ***Frame Structure*** | FDD |
| ***UL/DL Frame Length*** | 10 ms |
| ***Modulation*** | 64QAM, 16-QAM |
| ***Antenna Type*** | Omni-directional |
| ***Simulation Duration*** | 25 s |

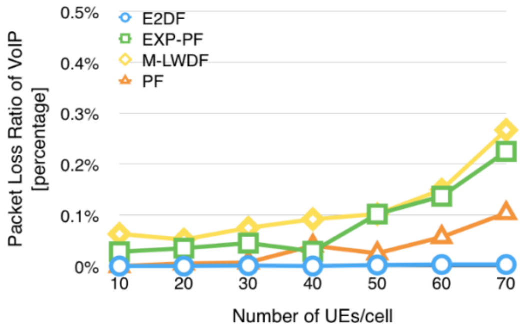


Fig. 3. Packet Loss Ratio of VoIP

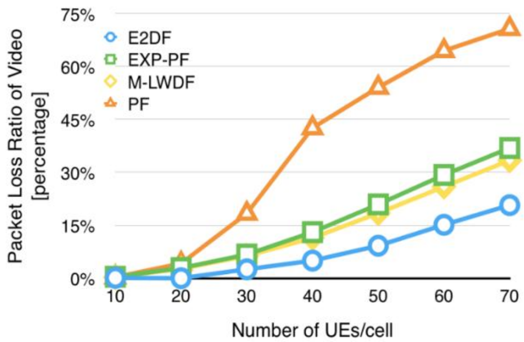


Fig. 4. Packet Lost Ratio of Video

Fig. 3 and Fig. 4 show the PLR for video and VoIP flows respectively. For VoIP flows, there are no significant changes in PLR with the changes in a number of UEs and scheduling algorithms. However, PLR for E2DF remains zero even regardless the changes in the number of UEs. PLR for other algorithms reaches up to 0.3% when number of UEs reaches 70. In contrast, the video flows have significantly lower PLR in E2DF compared to all the other algorithms as observed. By comparing E2DF and PF, there is 241% decrease in PLR for E2DF, and respectively 78% and 61% decrease as compared to EXP-PF and M-LDWF at 70 UEs. In other words, E2DF performs better in terms of PLR among the other 3 algorithms that have been tested.

Fig. 5 and Fig. 6 show the average packet delays for VoIP and video flows respectively. Like the PLR performance, VoIP delays for all scheduling algorithms are almost similar. The delays are always less than 10 ms. However, the delay for M-LWDF and E2DF are observed to increase slightly when the number of the UEs increases. On the video flows, all algorithms have an average delay increment as the number of UEs increase. E2DF has 64% increase in average delay compare to the best average delay by EXP-PF and 594% decrease in average delay as compared to the worst average delay by PF at 70 UEs. Overall, E2DF maintains a reasonable average delay in both video and VoIP flows.

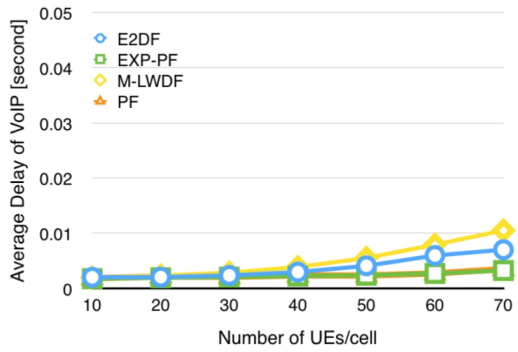


Fig. 5. Average delay of VoIP

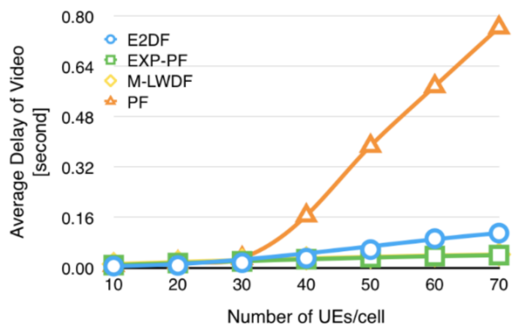


Fig. 6. Average delay of video

Fig. 7, Fig. 8 and Fig. 9 show the average goodput for VoIP, video and best effort flows. For VoIP flows, there is a constant increment in average goodput for all scheduling algorithms when the number of UEs increased. As observed in Fig. 8, at 30 UEs or more, average goodput tremendously decreases for the PF scheduling algorithm. When number of UEs more than 50, average goodput is getting worst in both M-LWDF and EXP-PF, but not E2DF. E2DF is maintaining robust average goodput among all the algorithms, with achieving 71% more as compared to PF and it is respectively 39%, 43% more if compared to M-LWDF and EXPPF. In fact, even when the number of UE reaches 70, average goodput in E2DF is still increasing, and there is no sign of it getting weak. As observed from Fig. 8, there are a lot of improvements on average goodput in E2DF, and this is achieved by giving high priority to the packet closest to meet its deadline. Also, as seen in Fig. 9, PF, M-LWDF and EXP-PF perform better as compared to E2DF. However, in the contrast to the average goodput for video and BE flows, fairness of E2DF between different services is better. For instance, at 60 UEs, E2DF allocates an almost similar amount of goodput for both video and BE flows.

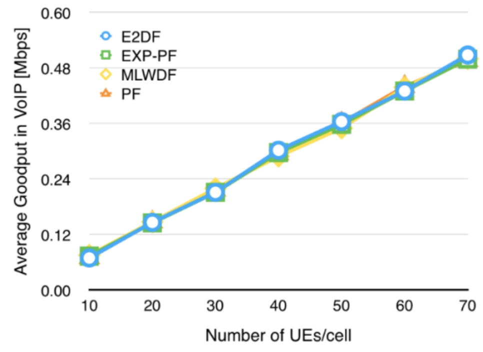


Fig. 7. Average goodput in VoIP

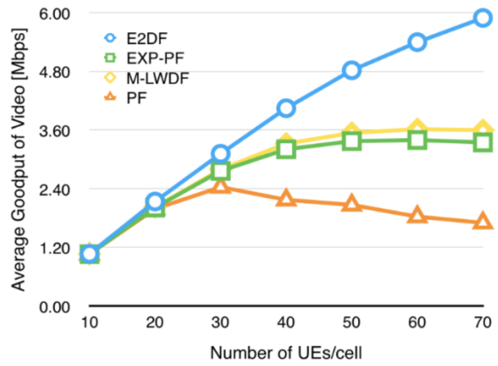


Fig. 8. Average goodput of video

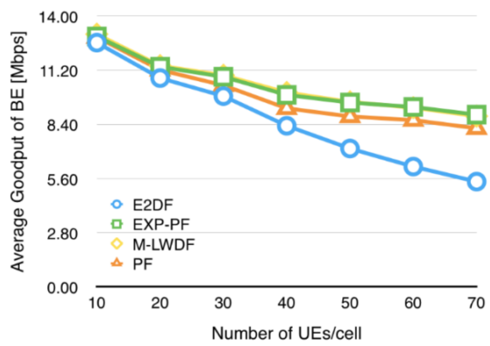


Fig. 9. Average goodput of BE

## VI. CONCLUSIONS

Based on the proposed downlink scheduler, E2DF, takes into consideration on both the channel condition as well as the QoS parameters, such as the packet's delay. Apart from that, the results obtained have shown an improvement in the real-time traffic performance of which includes the VOIP and video transmissions. Overall, E2DF performs better in terms of having a low PLR in both video and VOIP transmissions. Moreover, in VOIP, the PLR was managed to be reduced to zero as compared to the PF, M-LWDF and EXP/PF. As for the delay, E2DF maintains a reasonably low percentage of increment. E2DF provides a higher throughput value in video transmission. In short, E2DF is able to perform better compared to EXP/PF, PF, as well as M-LWDF in terms of bandwidth allocation, delay, goodput, and packet loss rate. The QoS for real-time application in LTE network is further assured by our proposed scheduling algorithm.

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