LTE Downlink Scheduling: A True Bayesian Estimate Approach

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Abstract—Extended research has been made in exploring the possibilities of a better real-time oriented downlink scheduling. The reason for these possibilities is caused by a fast-paced growth demand for multimedia applications that are mainly developed for mobile devices, and requires a high-speed wireless transmission for its satisfaction. Repositioning mobile devices have been one of the challenges arising from that demand. Due to the growth of mobile device users, another challenge has also been found, which is the capability of wireless networks to handle multiple simultaneous users within a single cell network environment. Current downlink scheduling algorithm which can cope with this challenge, Most Largest Weighted Delay First (MLWDF), needs to be improvised to suits the demands. True Bayesian Estimate (TBE) is one of the Bayes Estimator models which is suitable for handling multivariate parameters. Three proposed TBE algorithms have been designed with each having a different key design and objective. TBE-Fair (TBE-F) has provided a fairer and less delay scheduling as compared to MLWDF while TBE-Delay (TBE-D) manages to have a higher throughput rate. TBE-Flow Delay (TBE-FD) is an overall scheduler that manages multivariate QoS to perform better for real-time scheduling. All the TBE's algorithms have better performances than MLWDF in real-time traffic due to its main key design of real-time oriented scheduling which focuses more on video and VoIP flows.

Index Terms—LTE, estimate-based, Bayesian, true Bayesian estimate, downlink scheduling

I. Introduction

Multimedia applications have been growing at a fast pace along with the growth of mobile devices. To cope with the demand of every multimedia application, a stable and high-speed network is needed to handle the wireless transmission needed. Video streaming and calling have been famous in the mobile multimedia application and the process of transmitting it to multiple users at a simultaneous time has been a new challenge for wireless communication. Along with video streaming, Voice over Internet Protocol (VoIP) demand has also increased which leads to a challenge of stability in a voice calling application where less delay, jitter and packet loss are expected. Long Term Evolution (LTE) has taken the responsibility to handle all these challenges as it is the current wireless network technology used globally. Its performance has been proven but its sustainability to cope with the never-ending demand of multimedia applications has opened an opportunity for improvement.

The rapid growth increment of mobile device users has raised another concern in networking where mobility will affect the stability of network transmission. Due to moving users requesting and receiving transmission within a network cell, the bandwidth distribution will need to cope with the repositioning factor, which will disturb the scheduling process thus decreasing the network performance. Aside from the repositioning factor, fairness of bandwidth allocation would be another issue that needs to be considered, with multiple simultaneous users using the multimedia application, there would be multiple requests for downlink and uplink transmission and to be able to cope while providing the necessary fairness is another challenge.

On top of every factor mentioned, the main objective of handling real-time traffic is to optimize the performance of the multimedia application. This involves lesser delay and packet loss which can only be achieved if the fairness and bandwidth allocation perform well. Yet, the basic idea is to increase the throughput of the traffic which will result in lesser delay and packet loss and this can be achieved by optimizing the bandwidth allocation. Hence, optimizing the bandwidth allocation while coping with repositioning issues along with the fairness factor will be the ultimate challenge in satisfying the Quality of Service (QoS) of the real-time traffic. By fulfilling the requirement of QoS, the network performance will be in a better state thus satisfying the demand made by the multimedia applications.

To solve the ultimate challenge, a new scheduler needs to be designed with the entire key factor mentioned. Since most multimedia applications use more downlink requests than uplink, the solution is presented as a downlink scheduler algorithm that can prioritize real-time traffic. The objective is to reduce the delay and packet loss for real-time traffic experienced by the network. With the main focus on optimizing both video and VoIP traffic, the downlink scheduler algorithm had the capability of

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handling multiple simultaneous users. To achieve the satisfaction of the QoS, the proposed downlink scheduler algorithm is designed with incorporating True Bayesian Estimate (TBE) for provisioning and adaptive mechanism to cope with various uncertainty caused by the network through mobile usage. Bayes Estimator is one of the Bayesian families which often applied to calculate multivariate normal mean, empirical study of a sample and also used for fair rating system. TBE is one of the Bayes Estimator elements which widely used for the fair rating system application.

The problem with the existing scheduler such as MLWDF, and EXP/PF is they are not real-time oriented which means no priority will be given to a real-time transmission. Other than MLWDF, the existing scheduler does not support multiple simultaneous user transmissions which will lead to network congestion. To effectively deal with the uncertainty of real-time multiple simultaneous user transmission factors, TBE is used as the cost function which is prioritizing the real-time transmission factor and the user factor. Yet, no research that uses TBE in the wireless network has been carried out although TBE has shown its capability in obtaining outstanding performance in the computer industry. To our observation and findings, TBE implementation has a significant output in improving approximately 20% in video delay and 90% in VoIP delay as compared to MLWDF, where MLWDF is claimed as the most prominent scheduling algorithm for real-time traffic in a wireless network. For video and VoIP packet loss ratio, TBE has a significant difference of 10% and 64% respectively compared to MLWDF. TBE also has a higher fairness index of 8% in the video, compared to MLWDF. As for throughput, due to prioritizing real-time transmission, has around 103% improvement compared to other existing schedulers tested in the simulation.

This paper focuses on the downlink scheduling schemes for real-time applications in the LTE network. It is observed that the most common existing scheduler such as MLWDF and EXP/PF are not well-performing in scheduling real-time applications in LTE networks. Moreover, the proposed algorithm can improve the network performance (delay, packet loss ratio, fairness index, throughput and spectral efficiency) in the LTE network. Particularly, the contributions of this work can be summarized as follows:

1. This work discovers that network performance could be further improved by using the True Bayesian Estimate (TBE) approach in distributing the bandwidth fairly among all the downlink requests.

2. This work proposes and verifies the proposed TBE schemes with each having a different key design and objective. The three proposed TBE algorithms; TBE-Fair (TBE-F) provides a fairer and less delay scheduling, TBE-Delay (TBE-D) produces a higher throughput rate and the TBE-Flow Delay (TBE-FD) manages multivariate QoS well, for the point-to-multipoint (PMP) operation mode of the LTE network. The proposed schemes take into consideration all of the factors that could affect real-time applications and apply the TBE approach in the downlink scheduler. Furthermore, the proposed scheme had been tested and its efficiency verified in different network scenarios.

Further background study will be elaborated in Section II, which contains LTE architecture and downlink scheduler algorithm studies. In Section III, a further explanation of TBE and its practical example is presented. Section IV describes the proposed solution design and discussion. The results with discussion are depicted in Section V, where an explanation of the simulation parameter is included. Section VI contains the conclusion of this paper with the future works.

II. Background Study

In this modern world where wireless transmission is one of the main networking mediums, LTE has been handling various transmissions especially from mobile devices. With the current deployment of LTE, most users' applications experienced better connectivity especially in multimedia transmission. LTE is designed to support only packet-switched services [1], to provide seamless Internet Protocol (IP) connectivity among mobile device users. The main evolvement that LTE achieved is the transformation of the previous network radio access, Universal Mobile Telecommunication System (UMTS) to LTE standard radio access, Evolved UTRAN (E-UTRAN) [2] which is also accompanied by the evolution of the packet network access, Evolved Packet Core (EPC) [3], [4]. This architecture is the main foundation for LTE deployment which enables high-speed packet transmission over wireless transmission. Among both main architectures mentioned, E-UTRAN is the main processing transmission that has the responsibility of scheduling the traffic, load balancing and distributing the bandwidth [5]. Fig. 1 contains an illustration of the E-UTRAN architecture framework.

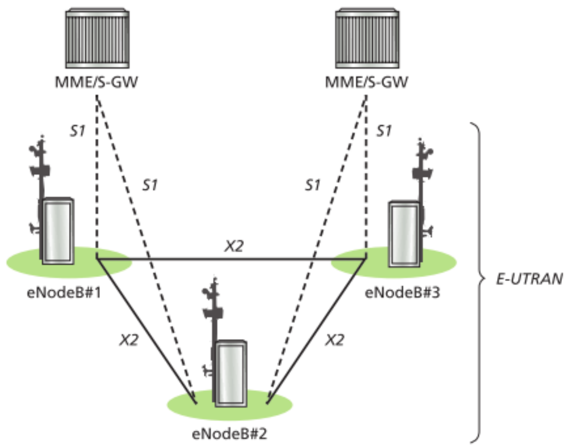

Fig. 1. E-UTRAN architecture [1]

E-UTRAN architecture consists of eNodeB (eNB) networks that interconnect with each other, as seen in Fig 1. E-UTRAN is the access network that coordinates the LTE transmission to the gateway, while the MME/S-GW gateway is the bridge from E-UTRAN to access EPC [1]. Most of the traffic schedulers are done in eNB, where every traffic will be modulated separately according to the type of transmission, both uplink and downlink. For every traffic that is running, the eNB will request a resource block to fulfill the traffic requirement. Downlink traffic has a better QoS mechanism since most of the QoS variables are retrievable for scheduling purposes. Hence, a better scheduler can be designed with various QoS key factors incorporated. The study is focused more on downlink scheduling which utilizes all the QoS key factors mentioned. The external key design is multiple simultaneous users and user repositioning is also included in the study.

A. LTE Downlink Scheduling Study

QoS satisfaction had always been the main objective of the scheduling algorithm. QoS satisfaction act as a requirement needed for the data flow of the traffic [6]. Most of the scheduling algorithms that existed in LTE downlink had always incorporated fairness as one of the key designs, one of QoS satisfaction requires fair bandwidth distribution. Another main reason fairness is incorporated is to provide a better load balancing of network transmission [7], [8]. In multimedia transmission which requires real-time flow and Guaranteed Bit Rate (GBR) are used in the scheduling to provide a stable transmission [9]. It is one of the key designs in LTE implementation where its responsibility is to handle real-time services such as conversational voice, video and gaming. Allocation of GBR is based on the downlink scheduling algorithm which allocates the metrics of priority for a data flow.

Another key design of LTE included is Non-Guaranteed Bit Rate (NGBR) which handles efficiently by a scheduler algorithm called Proportional Fairness (PF) [9]-[11], PF provides fair scheduling for NGBR which is used by the best effort data flow. PF has the capability of providing fair metrics for every user within the transmission period by utilizing the availability of the bandwidth and the average bandwidth used. Due to its fairness properties, most of the widely used downlink scheduling algorithms have applied PF as their base to maintain the fairness properties in their scheduling. Some of the known schedulers that implemented that method would be Modified Largest Weighted Delay First (MLWDF) [12]-[14], Exponential Proportional Fairness (EXP/PF) [13], [15], Opportunistic Packet Loss Fair (OPLF) [16], and Criterion-Based Modified Largest Weighted Delay First (C-BMLWDF) [17]. Most the downlink scheduling algorithms have less complexity with optimal performance. For this reason, the key design of downlink scheduling has been identified and a brief study of it based on some scheduling algorithms is presented. A new downlink-scheduling algorithm for serving video applications through LTE culler networks, and also accounts for QoS needs and channel conditions [18]. The author in [19] proposed two metrics, which increase the QoS-fairness and overall throughput of the edge users without causing a significant degradation in overall system throughput.

B. Modified Largest Weighted Delay First

MLWDF has been proven to be able to handle multiple simultaneous users thus providing a better Quality of Experience (QoE) to the users [12]. MLWDF has also been claimed to be one of the best downlink scheduler algorithms for managing real-time services [13]. One of the MLWDF traits that make the claim been made is its capability in handling up to more than 50 users at a time in a single cell network transmission. Besides that, MLWDF implemented a simple level of complexity which suits its implementation in most of the E-UTRAN architecture. Both supporting multiple simultaneous users and simple complexity are the key designs in MLWDF.

In QoS satisfaction, MLWDF performance has achieved a higher network throughput, fair distribution, less delay and less packet loss, which has proven to handle the video streaming service better [20]. MLWDF is a channel-aware algorithm that considers various factors such as Channel Quality Indicator (CQI) and also the data flow QoS [17]. Its multi-user diversity had also been tested in certain research and proven to be able to perform at optimal with 45 video streams and 175 VoIP at a simultaneous time [21]. Due to its achievement in many tests, MLWDF is considered one of the versatile scheduler algorithms that can meet the demand for real-time services.

MLWDF uses the Head-of-line (HOL) delay information received from the data flow requested by the UE and the drop probability set by the QoS to calculate the metrics of the scheduling. Incorporating PF as its numerous base operations, MLWDF also provides fairness for its real-time flow scheduling. Whereas for best-effort flow, MLWDF uses PF for its scheduler because best-effort traffic does not contain any specific requirement. This mechanism ensures MLWDF always prioritizes delay of real-time flow in scheduling thus providing effective scheduling even if the user increases in simultaneous time.

Due to the effectiveness of MLWDF, there is a lot of research that utilizes MLWDF as its foundation. Adaptive Modified Largest Weighted Delay First (AMLWDF) is one of the improvised versions of MLWDF where the adaptive mechanism is designed using Greedy Dynamic Resource Allocation (GDRA) which enables multiple RBs assigned to multiple channels [22]. Other research incorporates MWLDF with a virtual token concept to ensure the QoS value load balancing, called Virtual Modified Largest Weighted Delay First (VT-MLWDF) [7].

C. Criterion-Based Modified Largest Weighted Delay First

Criterion-Based (C-B) key design is based on one of the Bayesian equations called Bayesian Information Criterion (BIC), [23], [24]. BIC uses a criterion approach for model selection among a finite set of parameters, with the adaptation of likelihood function from the Akaike Information Criterion (AIC) [25]. With the incorporation of the Bayes Factor [26] into the function, BIC penalized the free parameter more than AIC which resulted in a better approximation outcome. Mathematically, BIC also incorporates Maximum Likelihood Estimation (MLE) [25] in some of its practical applications.

C-B utilizes the BIC normal assumption formula to provide a profit cost function to downlink scheduling by penalizing the delay as the observed data which resulted in higher metric output from a data flow that contains more delay. Since C-B implementation focuses heavily on penalization of delay, a better approach had been made to incorporate fairness into the algorithm. C-BMLWDF [17] incorporated MLWDF core design which utilizes the HOL delay and QoS drop probability as the free parameter and overall delay as the observed data which is then neutralized by PF as its numerous base operation, the same incorporation made by MLWDF. By adapting MLWDF core design, C-BMLWDF has the same traits as MLWDF, simple complexity along with the capability of handling multiple simultaneous users.

The main improvement and also difference between C-BMLWDF and MLWDF is that C-BMLWDF focuses more on real-time services where it only observes the real-time services delay, and it schedules based on the real-time services environment whereas MLWDF tends to take every delay into account. C-BMLWDF is a real-time flow-oriented downlink scheduler that utilizes and allocates more bandwidth to the real-time flow. It has better throughput, lesser delay and lesser packet loss compared to MLWDF [17] for video and VoIP flow. Similar to MLWDF, C-BMLWDF uses PF for its best-effort traffic scheduling.

III. Bayes Estimator

The Bayesian method has been used in various fields. Its capability of dealing with uncertainty factors makes it flexible to handle various key factors. For that reason, it is often incorporated into the artificial intelligence field to deal with constraints found in learning and adapting the technology. In the research expert system field, Bayesian Belief Network (BBN) is incorporated with the fuzzy network to produce a better root cause analysis application which has better learning in constructing a knowledge model [27]. BBN consists of two stages of learning, structure learning and parameter learning which are needed for a BBN to produce a Conditional Probability Table (CPT). A CPT will provide backward and forward inference capability mechanisms which are then used for the diagnosis and prognosis of a root cause [27]. Bayesian contains other statistical inference methods which are useful in various fields.

Bayesian is also not a stranger in the networking field where there are many types of research and applications that have been designed using one of the Bayesian methods as its core design. In research conducted for determining the congestion level that is experienced by a network, BIC has been used to produce the probability of the congestion level based on the information used in BIC operation [28]. Based on the determination made by BIC on the analysis level, the network then controls the cognitive multi-hope appropriately without reaching a higher congestion level. In research closely related to LTE downlink scheduling, C-B [17] has been used as the weighting function for the proposed scheduling algorithm to produce the scheduler metrics as mentioned in Section II. While in another field of networking, Bayesian Game is used as the method to distribute resource allocation in a wireless network under the condition of uncertainty variable present [29]. By utilizing the Bayesian Nash Equilibrium [30], [31], the researchers manage to solve the bandwidth distribution sharing problem caused by multiple mobile nodes from a single wireless access point.

In this research, Bayes Estimator would be incorporated into the proposed downlink scheduler algorithm. Bayes Estimator is often used to calculate the multivariate normal mean of a value or to construct an empirical study of a data set [32]. By incorporating the Bayes Estimator element in an anomaly detection system called Audit Data Analysis and Mining (ADAM) [33], the application has improved in detecting new novel attacks and helping reduce false alarm detection. The concept is to use Bayes Estimator as the filter to detect new attack properties that are different and distinguishable from the normal instance of training data [34]. The capability of the Bayes Estimator to estimate the likelihood of an uncertainty factor has been proven to be suitable in the networking field.

From a different area of research, Bayes Estimator has been used in a practical application of a rating system. The application uses Bayes Estimator and derives it to suit the rating system, calling it True Bayesian Estimate (TBE). TBE is derived from two average ratings; R represents the nearest neighbor's average rating and C represents the overall average rating [35]. In this application, TBE is used to weigh fairly the movie's rating on the top 250 lists which include old and latest movies. TBE manages to keep the high-ranked old movie rating along with a high-ranked new movie together with its variation of likelihood estimation. TBE utilizes the multivariate normal mean method in Bayes Estimator as shown in (1).

$$W = \frac{v}{v+m}R + \frac{m}{v+m}C \tag{1}$$

where:

W = weighted rating.
R = average rating of observed data.
v = number of votes for the observed data.
m = minimum weight given to the prior estimation.
C = the mean vote across the whole pool.

Equation 1 shows that W is obtained based on the variance that is given to the calculation. R represents the average rating of the particular observed data which varies depending on the variance. Another variable parameter would be v, which is subjective based on the observed data quantity. The modulator of the equation would be m, the minimum requirement for every observed data parameter, whereas C is the mean for the entire variance consisting multivariate value of the same observed data. TBE can deal with a multivariate parameter of the same observed data category due to the modulator and the total mean provided in the calculation and the incorporation of both provides a fair weighting mechanism estimation output. As the volume of m is exceeded, the confidence of the average rating surpasses the confidence of the prior knowledge, where the value of W would be straight average, R. Meanwhile, the closer v gets to the value zero, the closer W would get to C. This occurred as W acts as the weighted rating which is equivalent to the Bayesian posterior mean.

Based on TBE's unique traits of managing multivariate value, it succeeds in effectively weighting any of its applications by maximizing the posterior expectation of the observed parameter. Thus, TBE is suited to be the core foundation of the proposed downlink scheduling algorithm which needed a weighting function that uses various QoS parameters. The adaptive of TBE in handling uncertainty is another key design in handling the unexpected behavior in a downlink transmission which includes the variance of uncertainty, such as signal interferences and noises. The foundation of TBE will be built along with the proposed downlink scheduling algorithm as shown in Section IV.

IV. Proposed Solution

Following the previous research done in designing a proposed downlink scheduling algorithm that uses TBE, TBE-1 and TBE-2, they are yet to meet the minimum key design requirement [36]. Taking that into consideration of several key designs, PF [11] has been made as to the numerous base of the proposed solution to provide the fairness factor. Three solutions have been derived based on TBE-1 and TBE-2, with each of them having a different key design objective and a different set of parameters. The main objective is to design a more real-time oriented downlink scheduler that is tested in different QoS-focused parameters. TBE is flexible as it estimates based on the observed data and the variance provided which allow the solution to tackle different key design objective. At the same time, it is purposeful for getting an optimal performance scheduler. PF algorithm is shown in (2) and the proposed downlink scheduler algorithm can be seen in (3).

$$PF = \frac{Available\ Bandwidth}{Average\ Bandwidth\ used} \tag{2}$$

TBE-based algorithm:

$$W = \frac{Rv + Cm}{v + m} \times PF \tag{3}$$

Every proposed solution uses a different set of variances and has different observed QoS parameters. Hence, every solution has a different objective in its key design. The three solutions proposed are named TBE-Delay (TBE-D), TBE-Fairness (TBE-F) and TBE-Flow Delay (TBE-FD). Following would be the description of the proposed solutions.

A. *True Bayesian Estimate Delay (TBE-D)*

TBE-D focuses more on delay variance where there is a multivariate parameter regarding the delay. The main objective is to design a real-time downlink scheduler that schedules based on delay variance. The concept of focusing on delay derives from a lot of other famous downlink schedulers such as MLWDF [9] and Exaggerated Earliest Deadline First (E-EDF) [37]. Lesser delay will result in lesser packet loss; thus, improving the performance of real-time services scheduling. The idea of utilizing delay also affects one of the external key factors of multiple simultaneous user handling. This is because the prioritizing delay will enable the scheduler to reduce the current network delay as the user increases. There is another theory such as having lesser delay which would provide a more stable connection to multiple users which repositioning in a single cell network. The parameter assigned is shown in Fig. 2.

```
Algorithm: True Bayesian Estimate Delay scheduling for LTE Downlink
  {Initialization}
  Initialise Queue[] = 0
  Initialise i = 0
  Initialise metric = 0
  Initialise temp = 0
  R ← average bandwidth distribution based on number of flow
  v ← total real-time delay
  m ← QoS maximum delay requirement
  C ← mean of real-time delay
  RB ← resource block
  fd ← delay per flow
  i ← number of real-time flow
  {Main Iteration}
  for each i do
     {Inputs}
     Retrieve R, m, and fd
     v is equal to the sum of fd for every real-time flow
     C is equal to the mean of v
     {Main Process}
     Compute temp using Equation 3
     if temp is greater than metric, then
         metric = temp
         Put as lead in Queue[i]
         Assign RB's to respective flow in Queue[i]
     else
         Insert temp into Queue[i]
         Sort Queue[i]
         Queuing Queue[i] for RB's allocation
     end if
  end for
```

Fig. 2. Bayesian estimate delay algorithm pseudocode

In TBE-D, R represents the average bandwidth distribution based on the number of flows as its observed data. The observation of the current transmission bandwidth availability by determining the average bandwidth needed based on the number of the flow is a key design in measuring the tolerance of bandwidth distribution over the delay. This resonates with other multivariate delay parameters such as v which is representing the total real-time delay. v is crucial in TBE-D design as total real-time delay is varied from time to time, based on the transmission and user's involvement; higher users with a higher video or VoIP requests will result in a higher total real-time delay. Whereas m acts as the modulator of the algorithm, determined by the QoS maximum delay which is fixed for that certain eNB. As m represents the minimum weight given the prior estimation, QoS maximum delay suits the criteria as any transmission above the maximum delay would be discarded. C represents the mean of real-time delay for every stage in scheduling. The mean of real-time delay is one of the variances needed for better metric scheduling produced by TBE-D.

B. True Bayesian Estimate Fairness (TBE-F)

Algorithm: True Bayesian Estimate Fairness scheduling for LTE Downlink

```
{Initialization}
Initialise Queue[] = 0
Initialise i = 0
Initialise metric = 0
Initialise temp = 0
R ← ave bandwidth distribution based on queue size
v ← fairness value
m ← QoS maximum delay requirement
C ← mean of real-time delay
RB ← resource block
fd ← delay per flow
i ← number of real-time flow
{Main Iteration}
for each i do
    {Inputs}
    Retrieve available bandwidth, average transmission
    rate, R, m, and fd
    v is equal to available bandwidth divided by average
    transmission rate
    C is equal to the mean of fd for every real-time
    flow
    {Main Process}
    Compute temp using Equation 3
    if temp is greater than metric, then
        metric = temp
        Put as lead in Queue[i]
        Assign RB's to respective flow in Queue[i]
    else
        Insert temp into Queue[i]
        Sort Queue[i]
        Queuing Queue[i] for RB's allocation
    end if
end for
```

Fig. 3. True Bayesian estimate fairness algorithm pseudocode

As TBE-1 and TBE-2 [36] incorporated only PF for its fairness perspective, TBE-F's main objective is to focus on the fairness of the real-time scheduling. Fair scheduling provides better bandwidth distribution; thus it utilizes spectral efficiency. Some of the algorithms have prioritized fairness such as the one used in EXP/PF [15]. The concept of utilizing fairness is to handle multiple user transmissions throughout the user's growth. As the user increases, it will be hard to maintain the fairness of bandwidth allocation; thus, resulting in lower throughput that will lead to more delay and packet loss. Without fairness, the network performance will be easily decreased once it hits the user cap; limit of transmission in a network. Refer to Fig. 3 for pseudo code.

The average bandwidth distribution is chosen for R, with queue size as its measurement compared to the number of flows used in TBE-D. TBE-F focuses more on the bandwidth distribution based on the queue size since prioritizing the bandwidth needed by the queue size can increase the fairness of the transmission. Corresponding to that, v again uses PF to gain the fair value of the current transmission. PF which utilizes the value of the mean of bandwidth is a good fairness variance that changes over the transmission as the user increases. Another close factor related to fairness would be a delay as a higher delay indicates lower fairness, so m is the same as TBE-D, which uses the QoS maximum delay as its fixed variance. The mean of real-time delay, C, is used to measure the current network delay condition and a proper metric estimation can be done by TBE-F. TBE-F focuses more on utilizing the bandwidth fairness distribution.

C. True Bayesian Estimate Flow Delay (TBE-FD)

The main objective of TBE-FD is to focus more on the flow delay that is transmitted throughout the network. The concept of TBE-FD derives from the concept of every flow comes with a delay, by prioritizing delay that is higher for every flow which then leads to a better delay-focused scheduler being designed. The key design is to observe the delay based on the real-time flow of the current transmission. An algorithm such as VT-MLWDF [7], focuses on adapting the largest weighted delay first for every flow by assigning a virtual token as the indicator. The increase in flow will increase the delay; thus, incorporating flow and multivariate delay parameters is the TBE-FD key design. Refer to Fig. 4 for pseudo code.

For TBE-FD, the observed data in R would be the average real-time delay as the variance identify the average delay experienced by every real-time flow. With the total real-time delay as v, TBE-FD implemented the delay-focus flow observation throughout the transmission. The HOL packet size in m is the minimum indicator of the packet size carried, either video or VoIP traffic, while m is fixed based on real-time traffic type. The separation between video and VoIP flow can make the scheduler achieve fairer scheduling as the metrics are calculated based on flow and its delay. While C represents the real-time flow, separating the flow of video and VoIP, the real-time flow is always representing its mean. The key design is segregating the TBE-FD scheduling based on the flow and its delay.

Algorithm: True Bayesian Estimate Flow Delay scheduling for LTE Downlink

```
{Initialization}
Initialise Queue[] = 0
Initialise i = 0
Initialise metric = 0
Initialise temp = 0
R ← average real-time delay
v ← total real-time delay
m ← head-of-line packet size
C ← number of real-time flow
RB ← resource block
fd ← delay per flow
i ← number of real-time flow
{Main Iteration}
for each i do
    {Inputs}
    Retrieve m, C and fd
    v is equal to the sum of fd for every real-time flow
    R is equal to the v divided by C
    {Main Process}
    Compute temp using Equation 5.3
    if temp is greater than metric, then
        metric = temp
        Put as lead in Queue[i]
        Assign RB's to respective flow in Queue[i]
    else
        Insert temp into Queue[i]
        Sort Queue[i]
        Queuing Queue[i] for RB's allocation
    end if
end for
```

Fig. 4. True Bayesian estimate flow delay algorithm pseudocode

V. Results and Discussion

LTE-Sim [38] has been chosen as the simulator for the proposed downlink scheduling simulator. LTE-Sim contains most of the famous downlink schedulers such as PF, MLWDF and EXP/PF, which have a designated platform exclusively for the LTE network. The previous research of C-BMLWDF [17] is also conducted in the same simulator thus simulating the scheduler in LTE-Sim would be preferable. Based on the previous research conducted [6], [17], [36], the same simulation parameter is chosen to standardize the results standard. Table I presents the simulation parameter used.

Table I: Simulation Parameter for TBE'S algorithm Simulation

Parameters	Values
PHY	OFDMA
Bandwidth/Frame Length	5Mhz/ 10ms
Frame Structure	TDD
Modulation	QAM, 4-QAM, 16-QAM
Simulation Duration	60s
Number of Simulation	5
Traffic Models	Real-time: Video and VOIP; non real-time: best-effort
Mobility	eNodeB: Constant Position; UE: Random Direction
Speed	3km/h
Number of UE's	10-90
UE's Interval	10
Downlink Scheduling Algorithm	TBE-D, TBE-F, TBE-FD, C-BMLWDF, MLWDF

Some configurations have been made to suit the key design challenges; random positioning of UE's is for simulating the repositioning factor. Increment of 10 users per interval is to cope with the key design requirement which is multiple simultaneous user environments. A fixed video rate of 242kb is used for video traffic and 20 bytes per second transmission for VoIP [17]. The random repositioning is set to the limit of within 1km. The simulation is conducted in a single cell with interference to simulate a real-world single network environment. The results are obtained based on the QoS variable and discussed as follows.

A. Delay

Referring to Fig. 5, both TBE-D and TBE-F start with a lower delay value compared to other algorithms, having 0.00675s and 0.00745s respectively. Although TBE-FD has a higher starting value with 0.01629s, it is still lower compared to C-BMLWDF and MLWDF. At 20 users, TBE-D was shown to have a slightly lesser delay compared to other TBE algorithms, with 1% difference compared to TBE-F. TBE-FD is experiencing higher delay compared to its predecessor as it only has 13% to 20% difference compared to C-BMLWDF and MLWDF at 30 and 40 users. As the delay gradually increases, TBE-F is shown to have a lesser delay at 50 users. It is 35% different compared to C-BMLWDF and 39% for MLWDF. With an average different percentage of 15% to 20% better compared to TBE-D and TBE-FD, TBE-F has a good performance in handling video delay. The results concluded that TBE-F is the best for video delay with a difference of 24.2% between TBE-F and MLWDF at 90 users, while TBE-D has a lower delay compared to C-BMLWDF and MLWDF with 10% to 14% percentage difference. TBE-FD is still ranked higher than C-BMLWDF with a lesser delay value of 1.6% compared to C-BMLWDF.

By focusing more on fairness, TBE-F manage to have a lesser delay due to the effectiveness of prioritizing video flow to have more bandwidth allocation, which resulted in faster transmission and lesser packets in the queue. While TBE-D which focuses on the same key design as C-BMLWDF and MLWDF are having a slight improvement in video delay as it has an extra parameter observed for the weight calculation, which is the mean of real-time delay. TBE-D observes the mean of real-time delay prioritizes more for video flow as the mean will grow as the number of users increases. While TBE-FD which combined two key designs in its algorithm, focuses both on fairness and delay of the real-time flow, has a better result compared to C-BMLWDF and MLWDF. By focusing on the average real-time delay along with the real-time flow count, the scheduling would be based on the fairness needed to handle the delay available.

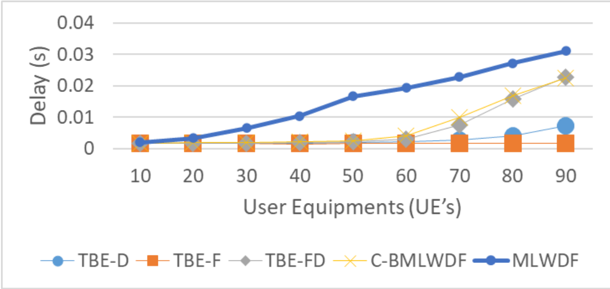

Fig. 5. VOIP delay

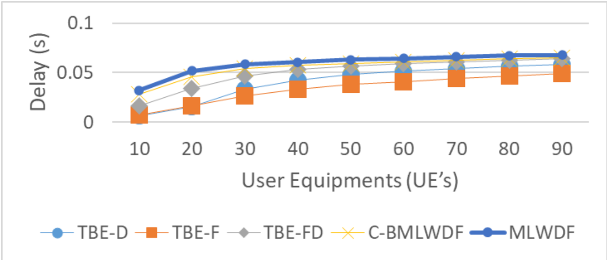

Fig. 6. Video delay

VoIP delay has a lower delay value for all algorithms. At the initial stage of 10 users, almost every algorithm started with a similar value of 0.0017 until 0.002 seconds. As the transmission started to increase to 20 users interval, it can be seen that MWLDF has a slightly higher delay of 0.003 seconds compared to other algorithms which remain below 0.002 seconds. Fig. 6 shows that MLWDF starts to increase rapidly in delay from 30 users with an average difference of 55% to 75% compared to other algorithms. TBE-F has shown a promising improvement in VoIP delay as it manages to get 85% different compared to MLWDF at 40 users. TBE-D started to have a better delay result at 50 users with 23% different compared to C-BMLWDF, 88% different compared to MLWDF, with the value of 0.0019 seconds delay. At 70 users, both TBE-FD and C-BMLWDF algorithms start to increase higher compared to the gradual increase of TBE-D and TBE-F. TBE-F and TBE-D have a small difference in delay with an average of 25% at the end of transmission which stable increment as shown in Fig 3. Again, TBE-F has proven to handle delay better with 90% difference percentage compared to MLWDF at 50 users and gradually performed better and reach 94% difference at the end of transmission. TBE-F also has 92% difference compared to C-BMLWDF at the end of the transmission.

TBE-D performs second-best by having 67% and 76% differences compared to C-BMLWDF and MLWDF at 90 users. TBE-F performance is slightly better than C-BMLWDF with an average difference percentage of 10% to 20% throughout the transmission. TBE-F performs better in VoIP delay as the fairness properties ensure better scheduling for VoIP flow which are having lesser delay compared to video flow. So instead of prioritizing the larger delay flow, VoIP flow is scheduled based on the fair value, which is the bandwidth needed for distribution gained from the average transmission rate and available bandwidth information. While TBE-D, which its key design is to focus on the real-time delay, had managed to get an improvement in VoIP delay handling compared to an older algorithm such as C-BMLWDF and MLWDF. By utilizing the mean of real-time delay and the total real-time delay, TBE-D managed to lower the delay experienced by the VoIP flow. TBE-FD which are having a higher delay at 90 users with a slightly different of 1% compared to C-BMLWDF, still outperforms C-BMLWDF in the earlier transmission with an average of 10% to 25% different. TBE-FD focuses on the balance of fairness and delay, which resulted in a fair delay prioritization for the flow but are lesser in value compared to its predecessor, TBE-D and TBE-F. Still, TBE-FD manage to achieve its objective of reducing the delay as compared to C-BMLWDF and MLWDF.

B. Packet Loss Ratio

PLR has always been influenced by other variables which are throughput, delay and fairness. Fig 7 illustrates a linear line for every algorithm with every algorithm throughout the simulation. By having a lower value at the start point with a value of 0.005%, both TBE-D and TBE-F gradually increase to 0.03% and 0.01% at 20 users. While TBE-FD are having 0.3% packet loss ratio at 30 users with different percentages of 9% and 26% compared to C-BMLWDF and MLWDF. As the transmission grows to 40 users, it can be seen that both TBE-F and TBE-D are experiencing a lesser packet loss ratio compared to other algorithms with a difference of 35% and 27% compared to MLWDF. All TBE algorithms started to have a higher packet loss ratio starting at 50 users and decreased its difference with C-BMLWDF and MLWDF with TBE-F which shave the lowest packet loss ratio, having 0.5%, 0.1% different compared to other algorithms at this user interval. Throughout the transmission, both TBE-D and TBE-F are having average difference percentage of 3% to 10% compared to other algorithms starting from 70 users point. The simulation concluded that TBE-D has a lesser packet loss ratio with 2% to 8% difference compared to another algorithm. TBE-F followed TBE-D with 6% difference compared to MLWDF and 2% rate behind TBE-D. TBE-FD has a slightly better performance compared to C-BMLWDF with 0.6% difference at the end of the transmission followed by C-BMLWDF which has 3% lower rate compared to MLWDF.

By focusing on delay, more video packets that experience more delay compared to other packets that arrive at the same time, are prioritized for scheduling. This ensures lesser video packets experience delay; thus increasing the chances of packet loss, which are shown in Fig. 7. As for TBE-F, it manages to have a low rate of packet loss due to its fair scheduling of video packets. As the user grows, more video packets would arrive simultaneously, which is affecting the queue size, so based on the available bandwidth of the current transmission, the scheduler calculated the needed bandwidth to be distributed based on the queue size thus prioritizing more video packet to be scheduled compared to other packets.

While TBE-FD which is the equilibrium of both its predecessor, TBE-D and TBE-F, takes into account almost every aspect of real-time transmission, considering both parties of video and VoIP. By monitoring the real-time flow and average real-time delay, with the key factor of the head of line packet size, every real-time flow is scheduled accordingly to its priority which resulted in lesser packet loss experience compared to an older algorithm such as C-BMLWDF and MLWDF.

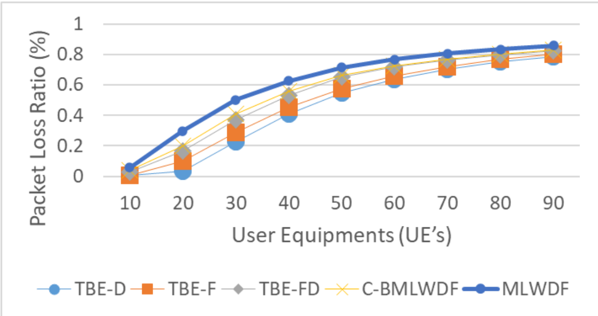

Fig. 7. Video packet loss

Fig. 8 shows that most TBE algorithms have a higher packet loss rate of up to 20 users. This is caused by the fact of VoIP has a higher drop probability due to its continuous stream of transmission in the simulation, every 20 bytes is transmitted for every 20 seconds resulting in higher packet loss and lesser delay. Most TBE multivariate is focused on real-time delay and having lesser delay due to the higher drop probability and small queue size, TBE is unable to perform well when the user is little. After 30 users, TBE-F starts to have a lesser packet loss rate compared to others with 28% difference compared to C-BMLWDF and 64% difference compared to MLWDF. TBE-D which is fully focused on delay can only be managed to surpass C-BMLWDF at 80 users, with 20% lesser packet loss rate and 44% at the end of transmission. TBE-D is experiencing a higher packet loss ratio with the maximum value of 0.004% at the initial of 10 users, with an average difference of 0.002% compared to other algorithms at the same user interval.

TBE-FD which utilizes the delay of flow has a better performance compared to TBE-F and TBE-D with an average of 50% difference compared to MLWDF throughout the simulation. C-BMLWDF has a similar value to TBE-FD when the transmission reaches 70 users and above. At the end of the transmission, it can be seen that MLWDF are having the highest packet loss ratio of 0.03% with a difference of 77% compared to TBE-D, 92% compared to TBE-F, and 54% compared to TBE-FD. TBE-D failed to surpass the VoIP packet loss ratio expectation at the earlier transmission due to its scheduler key design, VoIP flow is having a lesser delay compared to video, which makes it less prioritized and scheduled. Whereas the key design of fairness founded in TBE-F manages to overcome the problem as the user grows, where the more users with VoIP flow transmitting at the simultaneous time, the better it can prioritize due to its fairness properties. When the user increases, the average transmission rate increase, as well as queue size, which enable TBE-F to prioritize more VoIP flow for transmission. As for TBE-FD, it focused more on the balance of both real-time flow, VoIP and video, and prioritized according to the current network situation, so as the user increased, TBE-FD started to prioritize more VoIP flow for scheduling and manage to get the expected result. TBE-F and TBE-FD both manage to decrease their packet loss ratio starting at 30 users and maintain it to the end of transmission compared to C-BMLWDF and MLWDF.

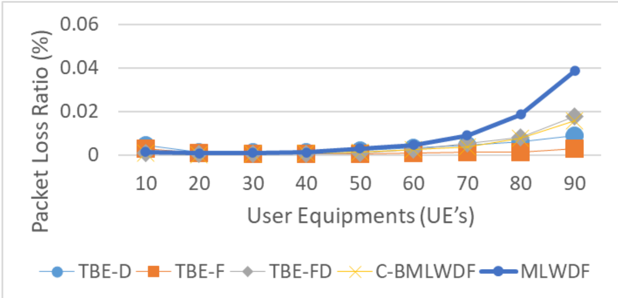

Fig. 8. VOIP packet loss

C. *Fairness Index*

The fairness index is the percentage to indicate the fairness of the scheduler which is affected by the fairness properties of every algorithm key design. MLWDF has better fairness compared to C-BMLWDF and TBE-D at the end of the transmission with 1.5% and 5% respectively. Due to the key design in TBE-F, the fairness index for TBE-F is the highest among other algorithms. TBE-F has 36% improvement at 40 users compared to MLWDF, it also has 8% better index at 90 users compared to C-BMLWDF. Other than TBE-F, TBE-FD also performed well by having a constant 1% to 5% difference compared to C-BMLWDF throughout the transmission. TBE-FD also outperforms MWLDF by having an average of 8% in most of the points and having 0.19% improvement at the end of the transmission. TBE-FD performs well in the fairness index as one of its key designs is determined by the variance of real-time flow every flow delay allows TBE-FD to be scheduled fairly. At the start of the transmission, every algorithm started with a value of 0.29% to 0.3% fairness index rate. As the user's interval increases to 20 users, TBE-F started to have a better fairness index value compared to C-BMLWDF and MLWDF with a difference of 3% and 11% respectively.

At the interval of 30 users, it can be seen that both TBE-F and TBE-D share a similar fairness index value of 0.34%, which is 15% different compared to C-BMLWDF and 29% compared to MLWDF. Starting from 40 users, TBE-F started to have the highest fairness index value compared to other TBE algorithms with 0.24%, while TBE-D is having 0.23%, and TBE-FD at 0.21%. As the user's interval grows, TBE-F started to show its effectiveness in scheduling fairly every multiple simultaneous flow which are consist of video, VoIP and BE traffic. TBE-D decreases dramatically at 60 users, losing almost half of its value at 20 users, with the fairness index value of 0.16%. TBE-D continue to decrease dramatically at 70 users with only 0.7% different percentage compared to C-BMLWDF and finally surpassed by C-BMLWDF at 80 users with a difference of 0.0016. As TBE-D focuses on delay, it failed to maintain the fairness properties of the flow as it prioritizes more delay packet for scheduling, which disrupt the balance of fairness index ratio of the flow. For video, the delay experienced is considered higher than other flow, so when multiple users are using the video traffic, the scheduler will choose the highest delay to be prioritized, making other videos flow at that particular time, being less prioritized, which leads to unfair scheduling. While TBE-FD, it maintains a constant fairness index which is better compared to C-BMLWDF with an average of 1% until 4% different from 30 users until the end of the transmission. Refer to Fig. 9 for results illustration.

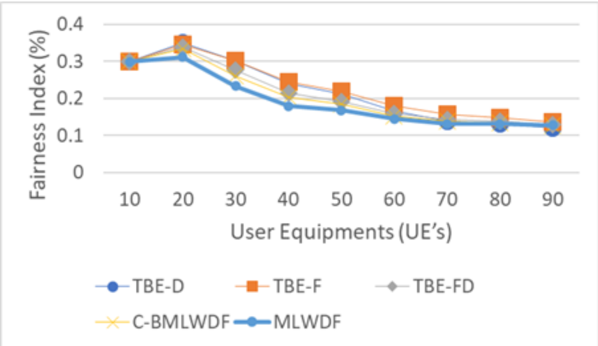

Fig. 9. Video fairness index

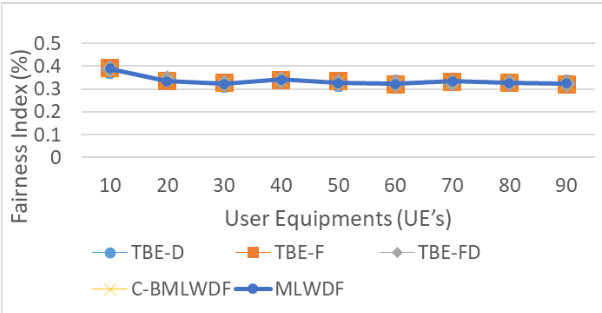

Fig. 10. VOIP Fairness Index

In Fig. 10, the fairness index of every algorithm is having a stable trend throughout the transmission. VoIP uses less bandwidth due to its small packet size and a continuous stream of transmission; thus enabling every algorithm to have a better fairness index. Every algorithm started with a similar value of 0.3% index, the index is maintained throughout the transmission with every algorithm having the same 0.3% value at the end of the transmission. Some fluctuations in the fairness graph shown in Fig. 7 are most probably caused by noise and interference that is a part of the simulation environment. Still, TBE-FD is shown to have a 0.5% average difference compare to C-BMLWDF and MLWDF at some point of the transmission. TBE-D and TBE-F have shown a decrease and increase in fairness index throughout the simulation which indicates instability of the scheduling that may be caused by the interference. TBE-FD would be the best scheduler that can maintain the performance until the end of the transmission with 0.04% difference compared to C-BMLWDF.

D. *Throughput*

As shown in Fig. 11, all the algorithms are having higher throughput compared to MLWDF. Due to the real-time oriented key design, the algorithms are having higher bandwidth allocation which resulted in a higher throughput rate. Among these algorithms, TBE-D is proven to have the highest throughput through its key design of observing the average bandwidth used for every flow. TBE-D has a stable increase from 31% at 20 users to 48% at 90 users compared to C-BMLWDF. TBE-D is also having an average of 90% to 103% improvement compared to MLWDF. TBE-F which focuses on fairness key design has a higher throughput compared to TBE-FD, C-BMLWDF and MLWDF with a peak improvement of 76% difference at 50 users. TBE-FD is slightly better than C-BMLWDF with 5% to 10% difference percentage throughout the simulation. Most of the algorithms have at least 30% improvement as compared to MLWDF. At the initialization of the simulation of 10 users, TBE algorithms started with 80 Mbps average. While both C-BMLWDF and MLWDF are having 7.8 and 7.5 Mbps respectively. As the transmission grow to 20 users, TBE-D started to have the highest throughput value of 157 Mbps, 10 Mbps different compared to TBE-F, the second-highest throughput scheduler.

At 20 and 30 user intervals, TBE-FD is shown to have a similar value of throughput of 120 Mbps average, 10 Mbps higher, compared to C-BMLWDF. While MLWDF is having its highest throughput value at 20 users, with 97 Mbps rate, which decreases even further throughout the transmission. At 40 users, TBE-D finally gain 103% different compared to MLWDF followed by 102% different at 50 users. C-BMLWDF, the variation of the MLWDF algorithm, manage to close the gap but still are having lower throughput compared to every TBE algorithm during the simulation. Although TBE-FD has lower throughput compared to its predecessor, it has balance properties that enable it to schedule based on the flow and the delay in the real-time, instead of focusing only on one matter. The importance of balance is to ensure that video and VoIP flow are scheduled without any of the flow having higher priority compared to another. If the delay is the main key design, the scheduling will be upsetting for VoIP as video flow would always have more delay.

While for fairness as a key design, the VoIP flow that uses a lesser average transmission rate would be prioritized after the larger flow than has a higher average transmission rate is scheduled. So, to provide an equilibrium between these two factors, TBE-FD focuses on the delay for each flow while considering the size of each packet. This gives better fair scheduling while still prioritizing delay of real-time for each flow, video and VoIP. As seen in MLWDF throughput results, when the number of users increases, with multiple simultaneous real-time demands, the throughput decreases much faster compared to another algorithm. Due to it didn't focus on real-time traffic mainly, other algorithms such as C-BMLWDF and TBE algorithms managed to improve the throughput rate compared to MLWDF. The real-time schedulers focused more on improvising the real-time bandwidth allocation and throughput rate is the measurement of a better bandwidth allocation.

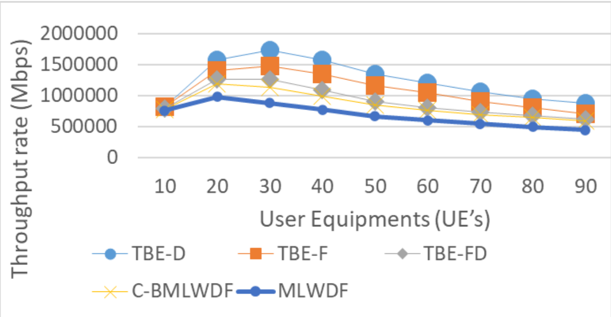

Fig. 11. Video throughput rate

Fig. 12 illustrates the linear line for every algorithm with similarity value among them. Due to the VoIP environment, it is clear that VoIP does not require a lot of throughputs for its processing; thus, most bandwidth allocation is done accordingly to VoIP needs. Although the differences are small, TBE-F and TBE-FD are proven to have a higher throughput rate for VoIP. TBE-F has outperformed every algorithm with the 3% average improvement for throughput rate followed by TBE-FD which have 2.6% higher throughput compared to MWLDF at the end of the transmission. TBE-D doesn't perform well compared to other TBE algorithm, but still have a slight improvement compared to MLWDF with 1.8% difference at the end of the transmission. C-BMLWDF outperforms TBE-D with a slight improvement of 1% starting 70 users. All algorithms started with an average of 3 Mbps at 10 users, with TBE-F having a slightly higher different percentage of 2% compared to the second-highest, MLWDF. TBE-F is again having the highest throughput at 20 users with 6 Mbps, 0.1 Mbps different compared to another algorithm. At 30 users, both TBE-F and C-BMLWDF share a similar throughput value of 9 Mbps, with MLWDF having a slightly higher value of 9.1 Mbps.

TBE-D has a better throughput at 40 users with a different percentage of 1% compared to MLWDF. At 50 users, TBE-D is having 2.9% difference percentage compared to both C-BMLWDF and MLWDF. While every TBE algorithm has lesser throughput compared to C-BMLWDF at 60 users, TBE-FD has 1% difference compared to C-BMLWDF with a value of 1.78 Mbps, 0.02 Mbps higher. As TBE-D seems to have lesser value throughout the transmission, the result is expected as the key design of TBE-D is revolving around the real-time delay. The larger the delay encountered by the flow, the more it will be prioritized in the scheduling. As VoIP always has a lesser delay due to its small packet size compared to video flow, it is understandable that it will be less prioritized in the scheduling; thus the lack of bandwidth allocated for it. While TBE-F on the other hand, focuses more on the fairness of the real-time, which ensures its scheduling prioritizes both real-time flows contained in the transmission. TBE-FD which focuses on the flow delay of the real-time traffic, manages to maintain its bandwidth distribution for VoIP throughput rate due to the scheduler prioritizes the real-time flow with the head of line packet size as its variant.

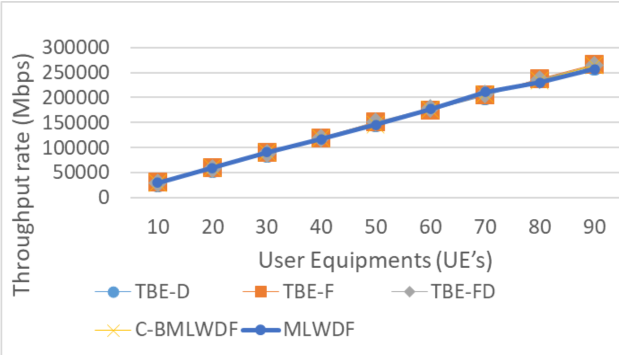

Fig. 12. VOUP throughput rate

E. *Spectral Efficiency*

Due to the key design of real-time oriented downlink scheduling, MLWDF still has the best spectral rate compared to other algorithms as shown in Fig 13. In addition, MLWDF focuses on the delay of every transmission which even covers the best-effort traffic delay. This in turn, enables MLWDF to effectively utilize the spectral efficiency of the whole network. Whereas C-BMLWDF [17] and TBE's algorithm are designed to focus more on the real-time traffic which resulted in a poor bandwidth allocation to best-effort traffic. Because best-effort traffic does not use much bandwidth and does not have the delivery priority, TBE's algorithms are designed to focus and utilized more priority scheduling to the real-time traffic which in this simulation environment consists of video and VoIP. However, the difference in spectral rate is not that big as TBE-FD has only 8% difference, compared to MLWDF with the benefit of having overall better performance compared to MLWDF in scheduling real-time traffic.

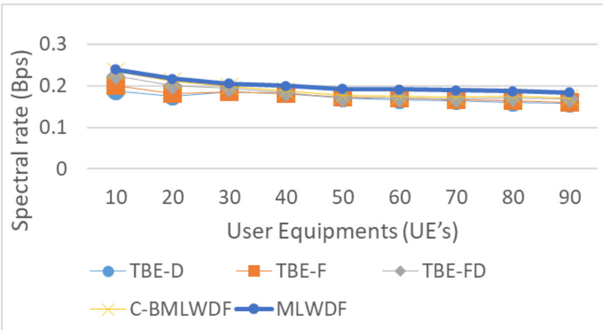

Fig. 13. Spectral efficiency rate

VI. Conclusion and Future Study

TBE's algorithms have proven to be an applicable solution for LTE downlink scheduling. Based on the simulation conducted, all TBE's algorithms outperform C-BMLWDF and MLDF in real-time QoS parameters. TBE-F is having lesser video and VoIP delay, lesser VoIP packet loss ratio and a higher fairness index compare to other algorithms. TBE-F's objective of utilizing real-time fairness has been achieved. TBE-D which has a key design of observing multivariate delay achieved its key design of having higher throughput, and lesser video packet loss ratio; but it did not achieve its objective of having lesser delay. TBE-FD is having an overall performance where it exceeds in having a higher VoIP fairness index and higher VoIP throughput, while TBE-D and TBE-F failed to surpass its predecessor, C-BMLWDF and MLWDF. Still, TBE-FD has lower performance in other QoS parameters compared to TBE-D and TBE-F, while achieving its objective of focusing on real-time flow, which resulted in a better VoIP parameter performance.

Conflict of Interest

The authors declare no conflict of interest.

Author Contributions

Khairul Anwar Bin Kamarul Hatta is the main contributor of this work. Wee KuokKwee and Cheah Wooi Ping supervise and guide in the entire research and simulation. Meanwhile, Wee Yityin shares and prepares some Bayesian knowdelge and algorithms.

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