A RESOURCE ALLOCATION METRIC FOR MULTIMEDIA CONTENT IN LTE NETWORK

ABSTRACT

New Generations of wireless networks such as LTE, should support bandwidth-consuming services like VoIP, Video, and multimedia streaming. Adding multimedia services to wireless communication networks leads to a lot of challenging issues in the radio resource allocation process. One of the fundamental resource management schemes in LTE networks is flow scheduling based on pre-defined metrics. Scheduling metrics definition, in particular, plays a fundamental role in selecting and distributing radio resources among different stations with fine time and frequency resolutions taking into account, channel conditions and QoS requirements. In order to reach acceptable performance, both frequency and time domain allocations for LTE resource blocks should be considered in the metric definition. One of the well-known LTE scheduling metrics, is EXP-Rule which controls real-time multimedia traffic delay. However, as a drawback, EXP-Rule does not consider the differences between flow characteristics like VoIP head of line delay compared to the video. In this paper, an exponential metric is proposed based on some modifications in the concept of resource allocation of EXP-Rule in order to differentiate between flow types with different heuristic queue sizes. This flow type differentiation, improves the metric fairness between flow types like VoIP and Video compared to EXP-Rule. Simulation results show a balance between VoIP delay and video packet loss rate. For the number of users equal to 70, in a single cell with interference, the proposed CB-EXP improves the video packet loss rate by 7.9%, 35%, 54.7%, 50.7%, and 56.6%, compared to EXP-MT, EXP, mEXP, sEXP and E2M respectively, also improves VoIP delay by 57%, 66%, 75% and 114% compared to mEXP, sEXP, EXP and EXP-MT respectively. In single cell without interference, for 70 users, the proposed CB-EXP metric improves video packet loss rate by 11.3%, 17% and 30% compared to LOG-RULE, EXP / PF and PF respectively and for VoIP packet delay, 11%, 15%, 103% and 32% compared to MLWDF, LOG_Rule, EXP / PF and EXP_Rule, respectively. On the other hand, applying the maximum throughput metric in the proposed formula, leads to better throughput results, especially in an interference including single-cell channels.

Keywords: Resource allocation, LTE, QoS, Video, VoIP.

INTRODUCTION

Packet delay and loss ratio are the main parameters that affect QoS in LTE access networks. Therefore, resource allocation policies including scheduling play important role in network performance. Scheduling is one of the main aspects of the LTE network to distribute available time-bandwidth resources efficiently among active users. Different QoS and QoE-related parameters like SINR, flow resource allocation history, queue status, and buffer status should be considered in resource allocation among user equipment (UEs) [1, 2]. Radio access in the LTE system uses Orthogonal Frequency Division Multiple Access (OFDM) technology that supports both frequency division multiplexing (FDD) and time-division multiplexing (TDD) [3, 4]. In the frequency domain, the total available bandwidth is divided into 180 kHz sub-channels where each sub-channel consists of 12 subcarriers. Resources are framed in the time domain with 10 ms duration where each frame consists of 10 Time Transmission Interval (TTIs). Each TTI, consists of two 0.5 ms slots each corresponding to 6 or 7 OFDM symbols. Each time-frequency segment called a resource block (RB), consists of a frequency domain sub-channel and a time slot with length of 0.5TTI in the time domain. *Fig. 1* shows an LTE frame structure [1, 4]. LTE resource allocation is a process based on which a scheduling metric is calculated for all users and for each resource block, in order to determine priority for each user to be scheduled. The most appropriate modulation and channel coding, is applied for each RB allocated in scheduling process [5]. It should be noted that the scheduling process is performed after sending channel quality indicator (CQI) (the instantaneous Signal to Interference plus Noise Ratio, SINR) by users to the related Evolved Node B (eNB). The scheduling module allocates

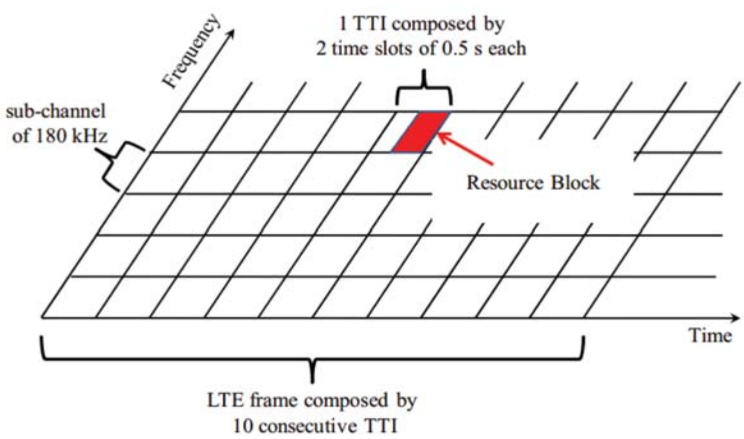

Fig. 1 LTE frame structure [1]

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resource blocks between active users based on CQI and other QoS parameters [1, 5]. First Input First Output (FIFO) and Round Robin (RR) are well-known algorithms generally applicable to wireless networks and fall into the category of channel unaware and QoS unaware algorithms.

Earliest Deadline First (EDF) and Largest Weighted Delay First (LWDF) are two metrics defined for real-time traffic in wireless networks for sending packets prior to expiration [6, 7] by applying D_{HoL} in their corresponding formula. The EDF and LWDF metrics are only aware of the QoS requirements and are not able to monitor instantaneous channel conditions. In [8, 9] the Proportional Fair scheduling metric for multi-carrier data transmission systems is introduced. This metric allocates resource blocks to users at each TTI, by getting feedback from the user's channel condition and calculating the capacity using the Shannon theorem as well as the history of resources allocated to each user. The metric is according to (4) and (5) indicates the Shannon theorem.

$$m_{i,k}^{PF} = d_k^i(t) / \overline{R^i}(t-1) \tag{4}$$

$$d_k^i(t) = \log\left(1 + SINR_k^i(t)\right) \tag{5}$$

Where $d_k^i(t)$, is the expected data rate of ***i-th*** user on the ***k-th*** resource block at time ***t*** and is calculated by the Shannon theorem. On the other hand, $\overline{R^i}(t-1)$ is the average received throughput by the ***i-th*** user before time ***t*** and $SINR_k^i(t)$ is the signal to interference plus noise ratio of the ***i-th*** user in the ***k-th*** resource block. The Proportional Fair scheduling algorithm falls into the category of scheduling methods that are channel condition aware with the drawback of not supporting QoS.

Another channel aware and QoS unaware scheduling algorithm is maximum throughput (MT) metric that is provided in equation (5) [10].

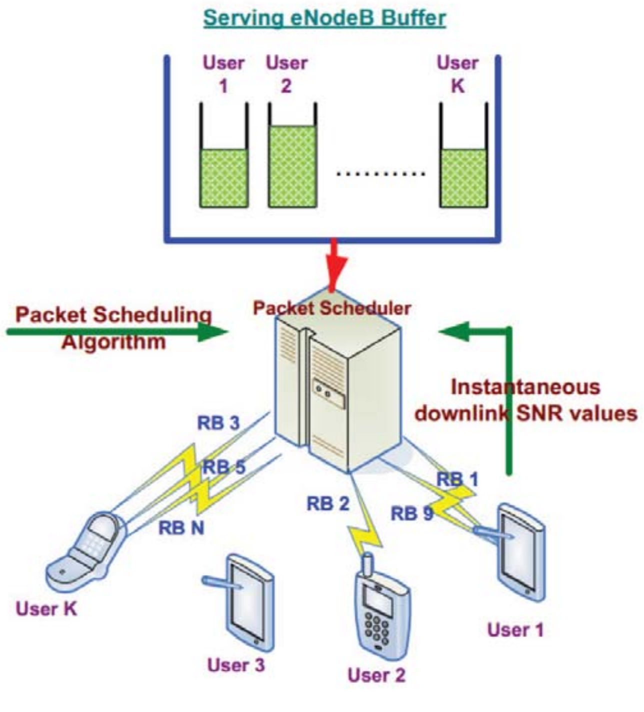

Fig. 2 General model of multi-carrier wireless scheduling [9]

Other important LTE scheduling metrics falling in the category of channel and QoS aware algorithms, are discussed briefly as follows.

In [8], M.Andrius et.al proposed Modified LWDF (MLWDF) as a modified version of LWDF considering QoS requirements [11] as shown in (6):

$$m_{i,k}^{MLWDF} = \alpha_i . D_{HOL,i} m_{i,k}^{PF} \tag{6}$$

Where α_i is computed by $\alpha_i = \dfrac{\log \delta_i}{\tau_i}$ and its value is selected based on legitimate requirements of acceptable PLR and packet expiry deadlines of two flows with the same HOL packet delay [1, 12]. The considering terms in MLWDF is as follows:

1. $Pr\left\{D_{HOL,i} > \tau_i\right\} \le \delta_i$
2. The average provided throughput to the ***i-th*** user must be greater or equal to a predefined limit.

Thus, by applying quality parameters, the MLWDF metric can differentiate real time and non-real time traffics [11, 13]. The performance of this algorithm degrades with increasing number of users [14]. Authors in [14] have defined the Exponential / PF (EXP/PF) metric as shown in (7):

$$m_{i,k}^{EXP/PF} = \exp\left(\frac{\alpha_i D_{HOL,i} - \chi}{1 + \sqrt{\chi}}\right) * m_{i,k}^{PF}$$

$$\chi = \frac{1}{N_{rt}} \sum_{i=1}^{N_{rt}} \alpha_i D_{HOL,i} \tag{7}$$

$$\alpha_i \in [\frac{5}{0.99\tau_i}, \frac{10}{0.99\tau_i}]$$

In [15], the LOG-RULE metric is presented to support real time and non-real time services. The scheduling metric is as follows:

$$m_{i,k}^{LOG\text{-}RULE} = b_i \log(c + a_i D_{HOL,i}).\Gamma_k^i \tag{8}$$

Where $c = 1.1\,;\, b_i = \dfrac{1}{E\left[\Gamma^i\right]}$ and $\alpha_i \in \left(\dfrac{5}{0.99\tau_i}, \dfrac{10}{0.99\tau_i}\right)$ and Γ_k^i is the spectral efficiency of the ***i-th*** user for the ***k-th*** resource block. In [15-17] the EXP-RULE is introduced for real time traffics. This metric is an improved version of EXP / PF based on equation (9) in which one user queue delay is normalized by the total experienced delay by other active users [1]. In overall conclusion, it also performs better than MLWDF and LOG-RULE [18].

$$m_{i,k}^{EXP\text{-}RULE} = b_i \exp\left(\frac{\alpha_i D_{HOL,i}}{c + \sqrt{\left(\frac{1}{N_{rt}}\right)\sum_{j=1}^{n} D_{HOL,j}}}\right)\Gamma_k^i \tag{9}$$

Where $m_{i,k}^{EXP\text{-}RULE} = b_i \exp\left(\frac{\alpha_i D_{HOL,i}}{c + \sqrt{\left(\frac{1}{N_{rt}}\right)\sum_{j=1}^{n} D_{HOL,j}}}\right)\Gamma_k^i$ and ***c = 1***.

PROBLEM STATEMENT, MODEL, AND ASSUMPTIONS

With evolution of LTE system architecture, improving network performance encounters new challenges for radio resource management to offer fairness while providing bandwidth efficiency and users' QoS requirements [19].

Based on the existing results of scheduling metrics, the objective of this paper is to propose a metric to improve EXP_Rule metric while attempting to maintain the VoIP delay and video packet loss in an acceptable range as well as increasing the number of concurrent active users.

Consider the following system characteristics:

- N user input streams ($N = \{1, 2, ..., N\}$)
- One physical channel
- M CQIs at any time interval $[t, t+1]$ and $t = 0, 1, 2, \ldots$.
- A service data rate vector $\left(\mu_1^m, \mu_2^m, \ldots, \mu_N^m\right)$ assigned for each $CQI=m$,
  - Where μ_i^m is average service data rate for ***i-th*** user for $CQI=m$.
- Statistical priority vector $\left(\sigma_1, \ldots, \sigma_N\right)$ represents the random priority allocated to each user [16, 17, 20] where $\forall i\ \Sigma\sigma_i = 1$.
- The channel condition random process is assumed to be an irreducible discrete Markov chain with finite state space M.
- The number of users with the stream type i entering in the time slot t is represented by $A_i(t)$.
- Also, the average input data rate of i, which means the average of users of type i entering in the slot t, is represented by λ_i and $i = 1, ..., N$. It is assumed that all $A_i(t)$ are Independent and Identically Distributed (i.i.d) random variables with constrained exponential torque and shown as follows.

$$A = \{A_1(t), \ldots, A_N(t), t = 1, 2, \ldots, N\} \tag{14}$$

In addition, it is assumed that the average stream data rates of the above definition converge exponentially to their average data rates, meaning that for any $i \in N$ and $v > 0$, a fixed number $a = a(v) > 0$ exists so that equation (15) could be as an estimation relatively large n's [7, 21, 22]. The behavior of the whole system with the random process S is described as (16).

$$Pr\left\{\left|\frac{1}{n}\sum_{i=1}^{n} A_i(t) - \lambda_i \geq v\right|\right\} < e^{-an} \tag{15}$$

$$S(t) = \{(U_{i1}(t), \ldots, U_{iQ_i(t)}(t), \quad i = 1, 2, \ldots N;\ m(t)\} \tag{16}$$

Where $Q_i(t)$ length of the queue of type i at time t, according to (17). $m(t)$, is the instantaneous channel condition and $U_{ik}(t)$ is delay of the ***k-th*** user with ***i-th*** stream at time t. Users are numbered by order of entering. Hence, the number of each user will be between 1 and $Q_i(t)$. Also, if we show the number of users with ***i-th*** service served in the time slot t by $D_i(t)$, for each time slot t and for each ***i-th*** we have:

$$Q_i(t+1) = Q_i(t) - D_i(t) + A_i(t) \tag{17}$$

1.1 THE EXP_RULE METRIC

The mapping H that takes the system with state S(t) in the time slot t into the probability distribution of H(S(t)) fixed on the vector cluster, called the scheduling rule or queue order. In Section 2, the overall system model, the underlying assumptions, and the overall scheduling operation corresponding to a channel and QoS aware approach are discussed. The important result of the discussion is to derive a scheduling rule based on the model and general assumptions for the system. This scheduling rule can take into account the QoS requirements that apply to the metric function with its corresponding variable [22]. In the following, after introducing the scheduling EXP_Rule in general, a new metric based on EXP_Rule is proposed.

The following equation shows the general form of EXP_Rule with positive and constant parameters $\gamma_1, \ldots, \gamma_N, a_1, \ldots, a_N, \beta$ and $\eta \in (0,1)$.

$$i \in i(S(t)) = \arg\max\ \gamma_i\, \mu_i(t) \exp\left(\frac{a_i Q_i(t)}{\beta + \left[\overline{Q(t)}\right]^{\eta}}\right) \tag{18}$$

و

$$\mu_i(t) = \mu_i^{m(t)}$$

Where a_i, are positive values and represent a function of QoS requirements and γ_i, are positive arbitrary values that usually considered to be 1 [22]. Also $\mu_i(t)$ is the weight that is multiplied by the exponential expression and can be the output of any channel aware algorithm such as PF or MT.

1.1.1 SCHEDULING PROCEDURES AND GENERAL FLOWCHART OF EXP_RULE

The step-by-step scheduling procedure based on the EXP_Rule metric is expressed as follows:

1) Classify and prioritize packets based on their calculated metric;
2) Calculate HOL packet delay based on the arrival time of packets and total delay that each packet has experienced;
3) Compare Total Delay with the tolerable delay threshold value of each packet;
4) Removing the packet if it exceeds the acceptable delay threshold;
5) Calculate a_i using the equation defined for it; and
6) Calculate the service data rate vector $\mu_i(t)$.

The general flowchart for a TTI is shown in *Fig. 3*.

PROPOSED SCHEDULING METRIC

Based on general idea of exponential metrics, a new metric is proposed to improve VoIP delay and video packet loss compared to EXP. Since the main idea of proposed metric is applying the effect of content type in formula, the authors, name it "content based EXP" or CB_EXP with the below formula:

$$m_{i,k}^{CB\_EXP} = \exp\left(10 * \exp\left(D_{HOL,i}\right)\right) m_{i,k}^{MT} \tag{19}$$

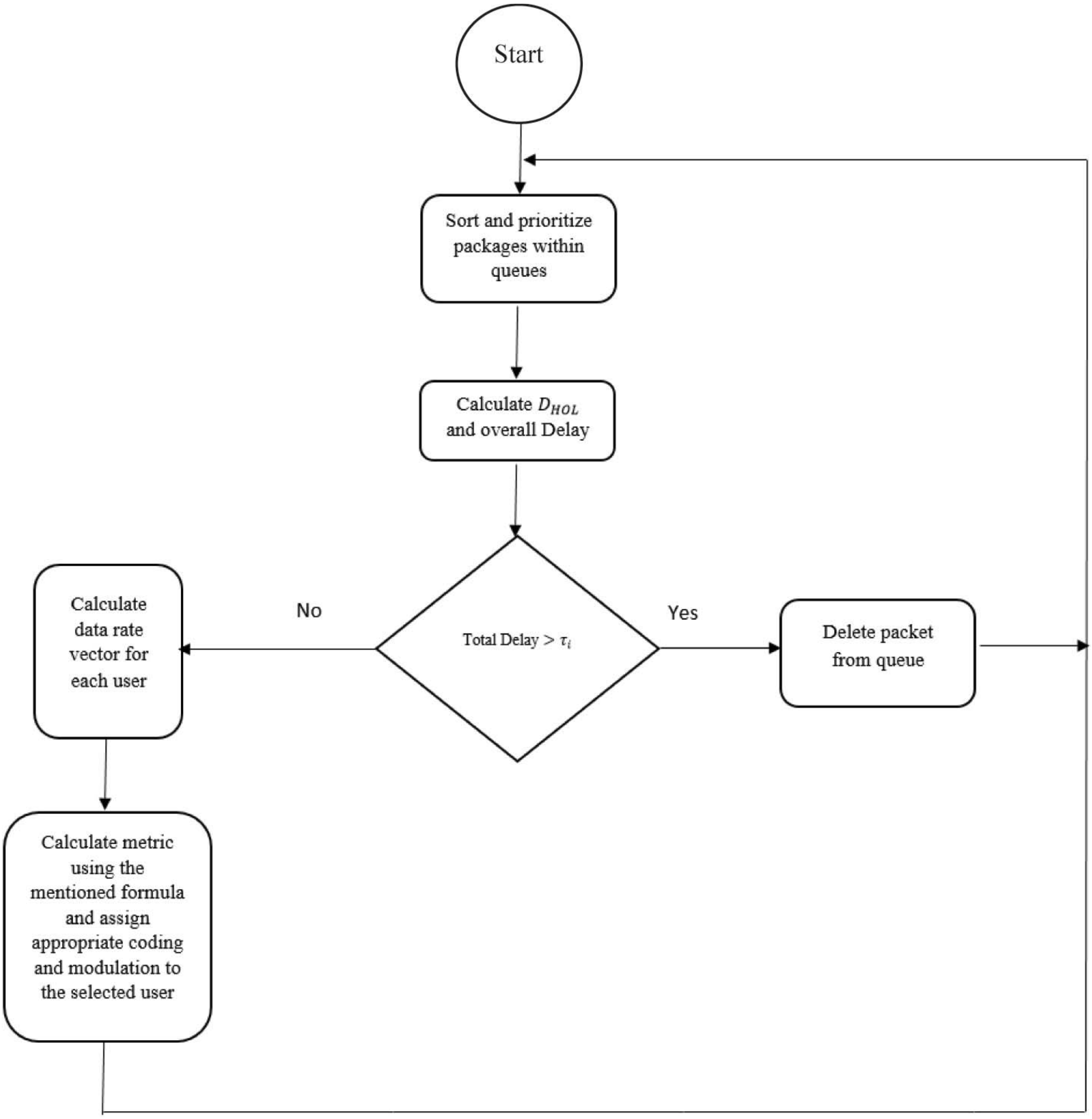

Fig. 3 General flowchart of EXP_Rule scheduling

Where:

- $D_{HOL,i}$, represents the head of line packet delay for i-th user.
- $m_{i,k}^{MT}$ is MT metric for i-th user in k-th resource block.

Applying MT factor in the proposed channel and QoS-aware metric, improves performance, especially in cells with interference.

Due to the special mobile channel conditions in the interference scenario, to prevent retransmission of the packet, which results in waste of resources, the weight MT is considered to give less priority to resource allocation for states with low SINR. There is also a good and acceptable tradeoff between delays and packet loss. It is necessary to explain how the system works. One reason for using the exponential function is to give higher priority to video streams rather than VoIP streams. The reason for this is as follows:

Since the average D_{HOL} for video packets usually is greater than for VoIP ones, its exponential effect will make the priority for video streams noticeably higher than that of VoIP streams. In other words, since packet delay and packet loss are usually inversely correlated, the scheduling metric needs to operate in such a way as to provide a balance between the two parameters and maintained at an acceptable point of QoS value. Experimental results show that the average overall delay as well as the D_{HOL} of video are higher than VoIP and this factor may increase the probability of the video packets expiring and being eliminated from the queue. So the metric needs to act in a way that takes this factor into account. If packet entering time is not considered in the scheduling, the video packets could be removed from the queue and increase the overall packet loss rate of video traffic [23]. As a result, Since D_{HOL} for video packets are greater than VoIP ones [1], the exponential function in the metric causes selective treatment of video packets. On the other hand, by increasing the waiting time in the queue (D_{HOL}), the performance of the proposed metric is appropriate due to the use of exponential function within the metric argument.

In the proposed CB_EXP method, due to the mathematical conditions underlying the exponential function, a metric is proposed considering the QoS parameters, which provides a good tradeoff between delay and packet loss for video and VoIP traffics. The difference between this algorithm and the existing metrics is to balance the two opposite parameters, delay and packet loss. The proposed metric has acceptable performance on video packet loss and VoIP delay parameters, especially in the presence of interference.

PERFORMANCE EVALUATION

In this section, the proposed CB_EXP metric will be compared to the existing metrics, such as PF, MLWDF, EXP / PF, EXP-RULE. The simulation results are obtained using LTE-Sim simulator [18] in two scenarios. In the first scenario, the comparison is performed in a single cell with interference. In the second scenario, the proposed metric will be compared in a single cell without interference.

2.1 SCENARIO 1: SINGLE CELL WITH INTERFERENCE

In this scenario, the performance of proposed CB_EXP metrics are compared to the EXP family methods considering two parameters, packet loss and delay, for both video and VoIP traffics. The simulation conditions are as shown in **Table 1** [1, 24].

Table 1: Simulation conditions of single cell with interference

Parameters	Values
Bandwidth/Frame length	5MHz/10ms
Frame structure	TDD
Modulation	QAM, 4-QAM, 16 QAM
Simulation Duration	46s
Stream Length	30s
Traffic Model	Real time traffic: Video & VoIP
Mobility	Fixed eNB but The user has a completely random move With speed of 3 km/hour
Number of users	5:5:70
Video traffic type	128 kbps/ foreman_H264_
VoIP traffic type	G.729 codec with a fixed packet length of 20 bytes over a period, plus 12 bytes of RTP header, totally 32 bytes
Cell number	7
Cell radius	1 km

According to the results of [18, 25], EXP-Rule performance in terms of packet loss rate and delay is generally better than MLWDF, PF, MT and EXP/PF. Therefore, the proposed metric only compares to EXP-RULE and other EXP-RULE extensions. Due to the statistical network features and random output at each test step, the simulation is performed up to 5 times and results are averaged for more accuracy.

Fig. 4 shows the video packet loss rate of the proposed and other metrics in the first scenario. As can be seen, the proposed CB_EXP metric has lower packet loss rate compared to sEXP, mEXP, E2M and EXP_Rule metrics. Albeit the proposed metric has higher packet loss rate than EXP_MT for number of users below 50, but as the number of users increases over 50, the packet loss rate decreases remarkably.

This result can be explained as follows: As more packets enter the queue and service demands increase, naturally the packet waiting time in the queue increases (D_{HOL}), and this variable is more significant for the video packet. In this case, due to the exponential function of the proposed metric, with the increase of the delay value of the video packet (D_{HOL}), its priority becomes noticeably high and as a result, the video packets are scheduled with higher priorities.

As depicted in *Fig. 4*, three CB_EXP, EXP-RULE and EXP_MT scheduling metrics are suitable for supporting video traffics. The CB_EXP metric performs better than the others in the number of users over 50. The percentage of improvement over EXP, mEXP, sEXP and E2M for 70 users are 35%, 54.7%, 50.7% and 56.6%, respectively. On the other hand, the proposed CB-EXP metric decreases the packet loss rate 7.9% compared to EXP-MT for a number of users equal to 70. It should be noted that the packet loss rate is computed using the equation (20) [26, 27].

Due to the expanding use of packet switching network users, attention to VoIP traffic is urgently needed. *Fig. 5* shows the packet loss rate result of the proposed metric compared to existing ones. As depicted in this figure, increasing the number of users leads to increasing VoIP packet loss rate. Also the proposed metric has more packet loss rate than the other metrics except E2M.

This is due to the increased speed of sending and scheduling of VoIP packets and naturally, reducing the integrity of scheduled VoIP packets. For packet loss rate of VoIP traffic, EXP-RULE metric has the best performance and can be used for VoIP traffic. Subsequently, sEXP and mEXP also have acceptable performance. It is worth mentioning that the proposed algorithm has acceptable performance up to 35 users. On the other hand, each metric is not capable of fully supporting all QoS parameters. Also, by recalling the purpose of this research to establish a state of equilibrium between delay and packet loss rate with the limited capacity of the system, in terms of the overall result, the proposed metric performs better than all existing metrics. For example, EXP_MT, which performs better in terms of video packet loss rate, has the worst performance in VoIP packet delay.

$$Packet\ loss\ rate = \left(1 - \frac{recievd\ packets}{sent\ packets}\right) \quad (20)$$

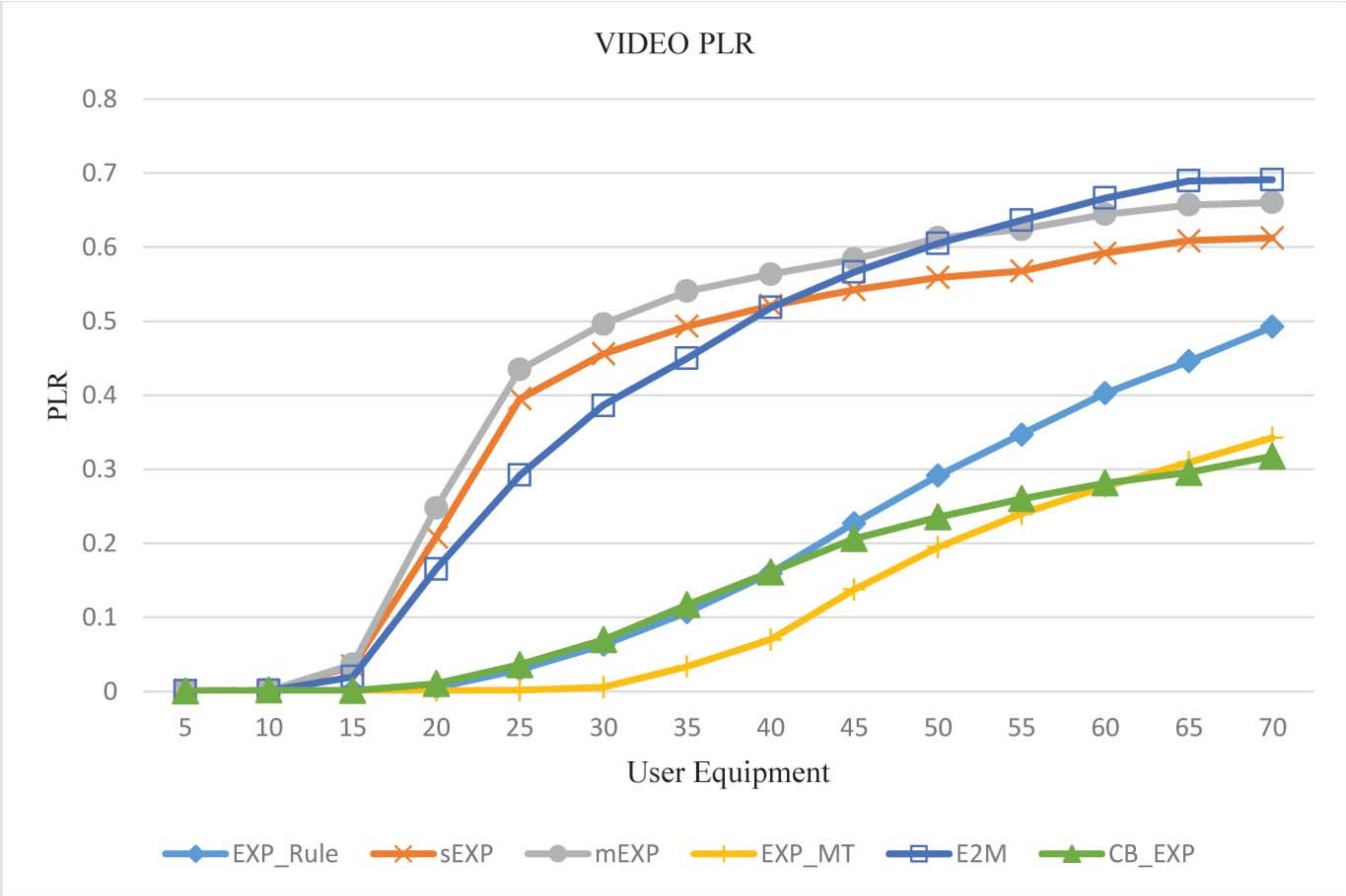

Fig. 4 Video packet loss rate in scenario1

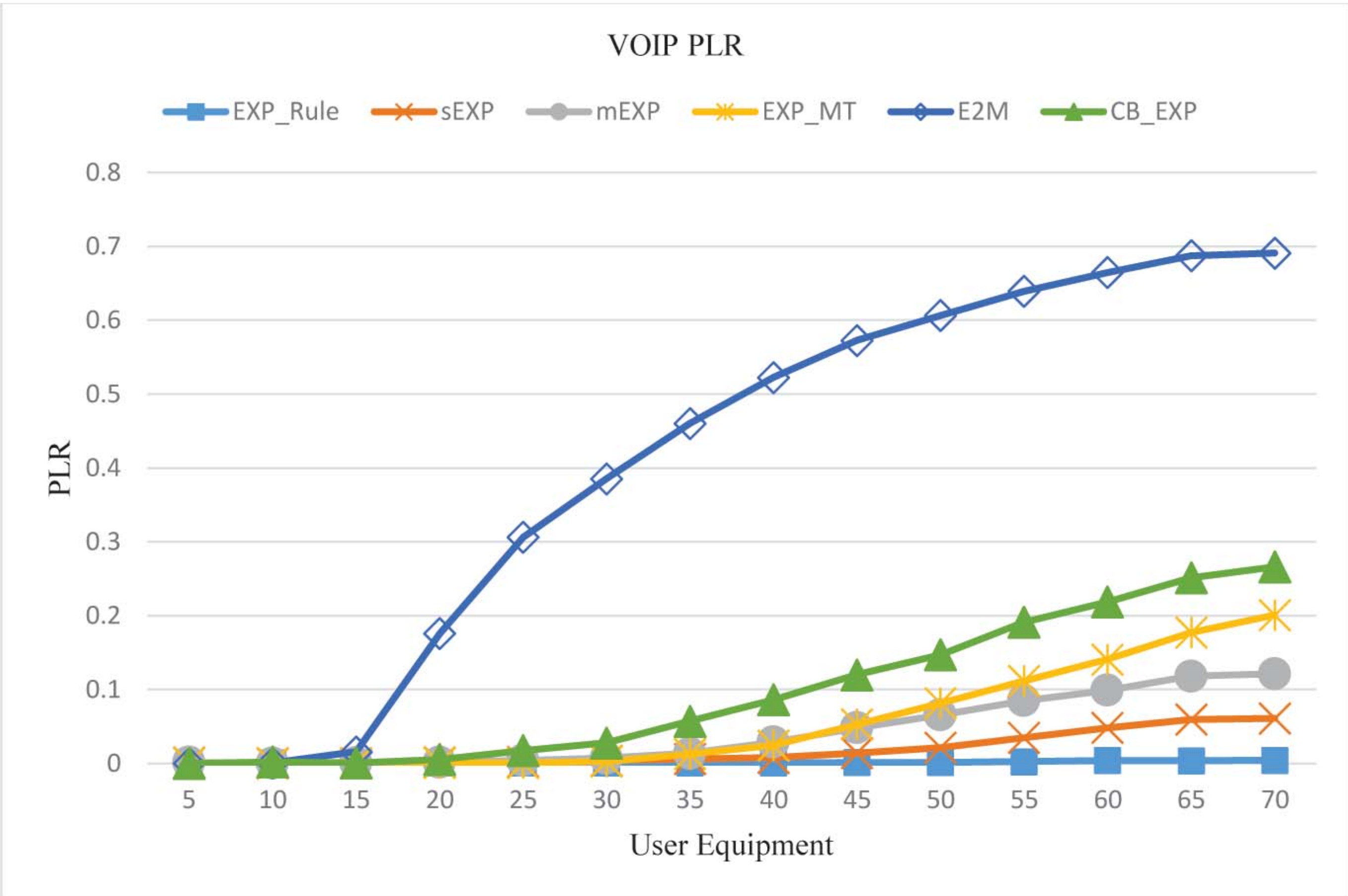

Fig. 5 VoIP packet loss rate in scenario1

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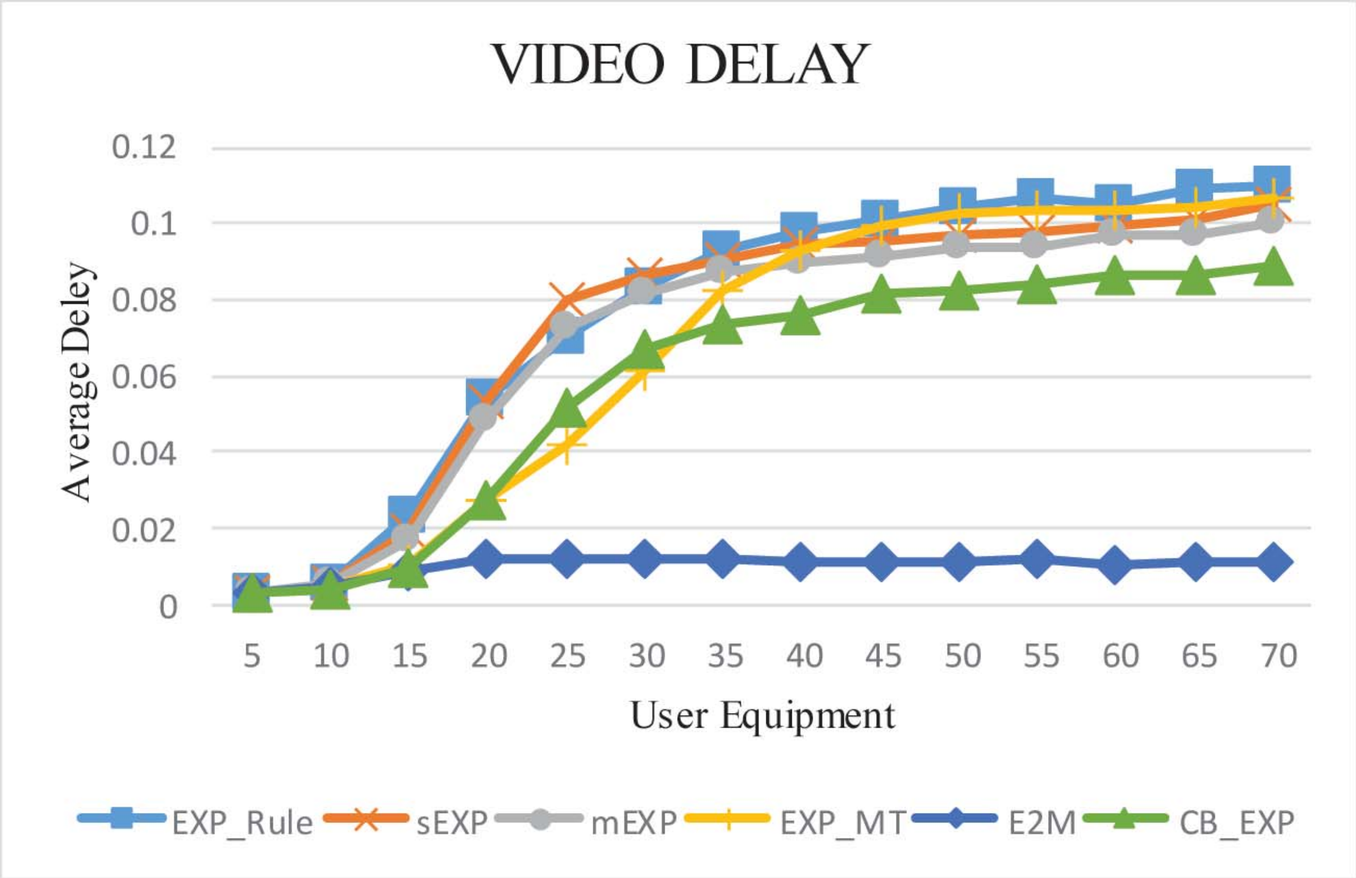


Fig. 6 Video packet delay in scenario1

Fig. 6 shows video packets delay by applying proposed CB_EXP metric compared to EXP, sEXP, mEXP, E2M and EXP_MT as a function of the number of users. As demonstrated, E2M metric has the least delay in scheduling video packets, but the overall performance of all metrics satisfies the threshold delay for real time video traffic [18].The CB_EXP metric has improved the video delay than other metrics except E2M. It should be noted that video and VoIP traffics delays are computed using the following equation [26]:

$$Delay = \frac{Sum\ of\ delays}{n_i} \tag{21}$$

Where, i indicate the type of traffic and n_i is the number of traffic with type i.

Fig. 7 shows the delay comparison of VoIP traffic. As can be seen, the EXP_MT metric enormously increased the delay for VoIP traffic. However, the CB_EXP metric has an acceptable delay for VoIP traffic [11, 28]. For the number of users equal to 70, the proposed CB_EXP metric improves VoIP delay by 57%, 66%, 75% and 114% than mEXP, sEXP, EXP_Rule and EXP-MT respectively.

The following general conclusion can be deduced in the single cell with an interference: the proposed CB_EXP metric competes with the EXP_MT and E2M metrics in two parameters, video packet loss rate and VoIP packet delay. The proposed CB_EXP metric improves packet loss rate for video traffic for the number of users more than 60. In other words, by using the proposed scheduling metric, the user number capacity for each cell in LTE system would be increased. Although EXP_MT decreases video packet loss rate remarkably, but excessively increases VoIP packet delay. E2M scheduler metric also works well in the delay parameter but does not improve video packet loss rate. The proposed CB_EXP metric not only decreases video packet loss rate, but also maintains video and VoIP traffic delays within an acceptable range.

2.2 SCENARIO 2: SINGLE CELL WITHOUT INTERFERENCE

In this scenario, CB_EXP metric performance is compared to PF, MLWDF, EXP_Rule, EXP / PF and LOG-RULE metric in terms of packet loss rate and delay for video and VoIP traffics. The simulation conditions are given in **Table 2**.

Table 2: Single cell without interference simulation conditions

Parameters	Values
Bandwidth/Frame length	5MHz/10ms
Frame structure	TDD
Modulation	QAM, 4-QAM, 16 QAM
Simulation Duration	190s
Stream Length	180s
Traffic Model	Real time traffic: Video & VoIP
Mobility	Fixed eNB but The user has a completely random move With speed of 3 km/hour
Number of users	5:5:70
Video traffic type	128 kbps/ foreman_H264_
VoIP traffic type	G.729 codec with a fixed packet length of 20 bytes over a period, plus 12 bytes of RTP header, totally 32 bytes

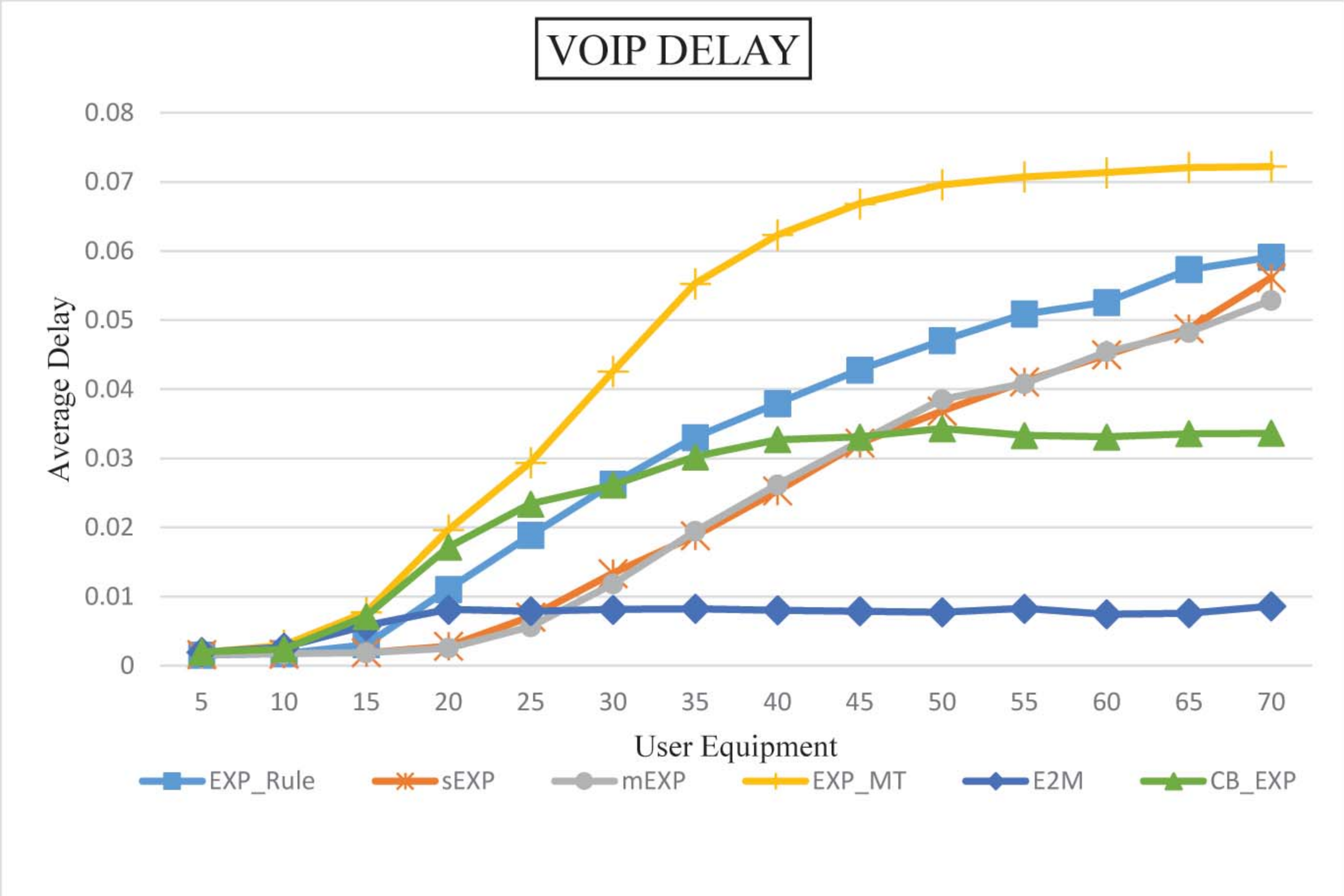


Fig. 7 Comparison of VoIP traffic transferred packets delay

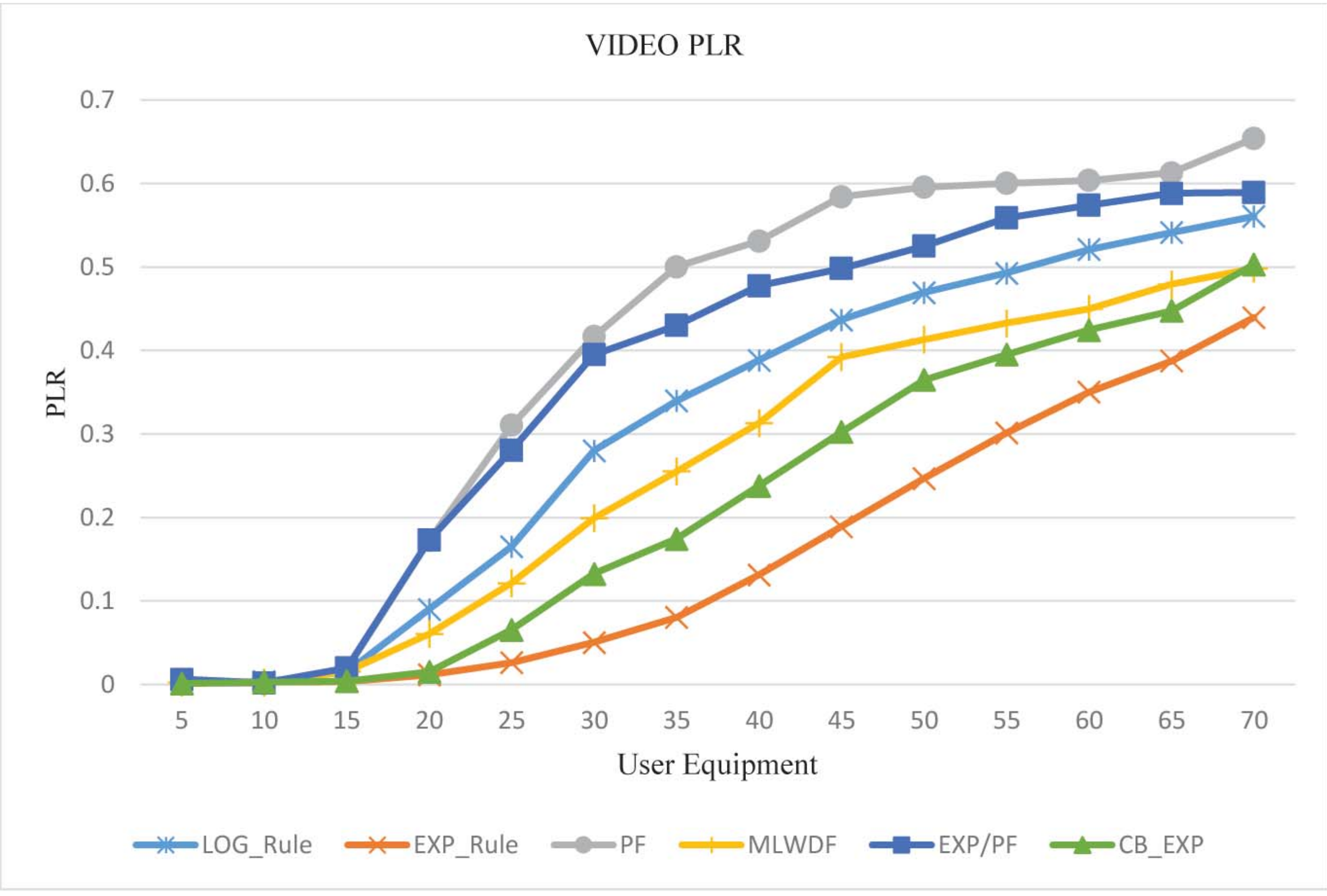


Fig. 8 Comparison of video packet loss rate

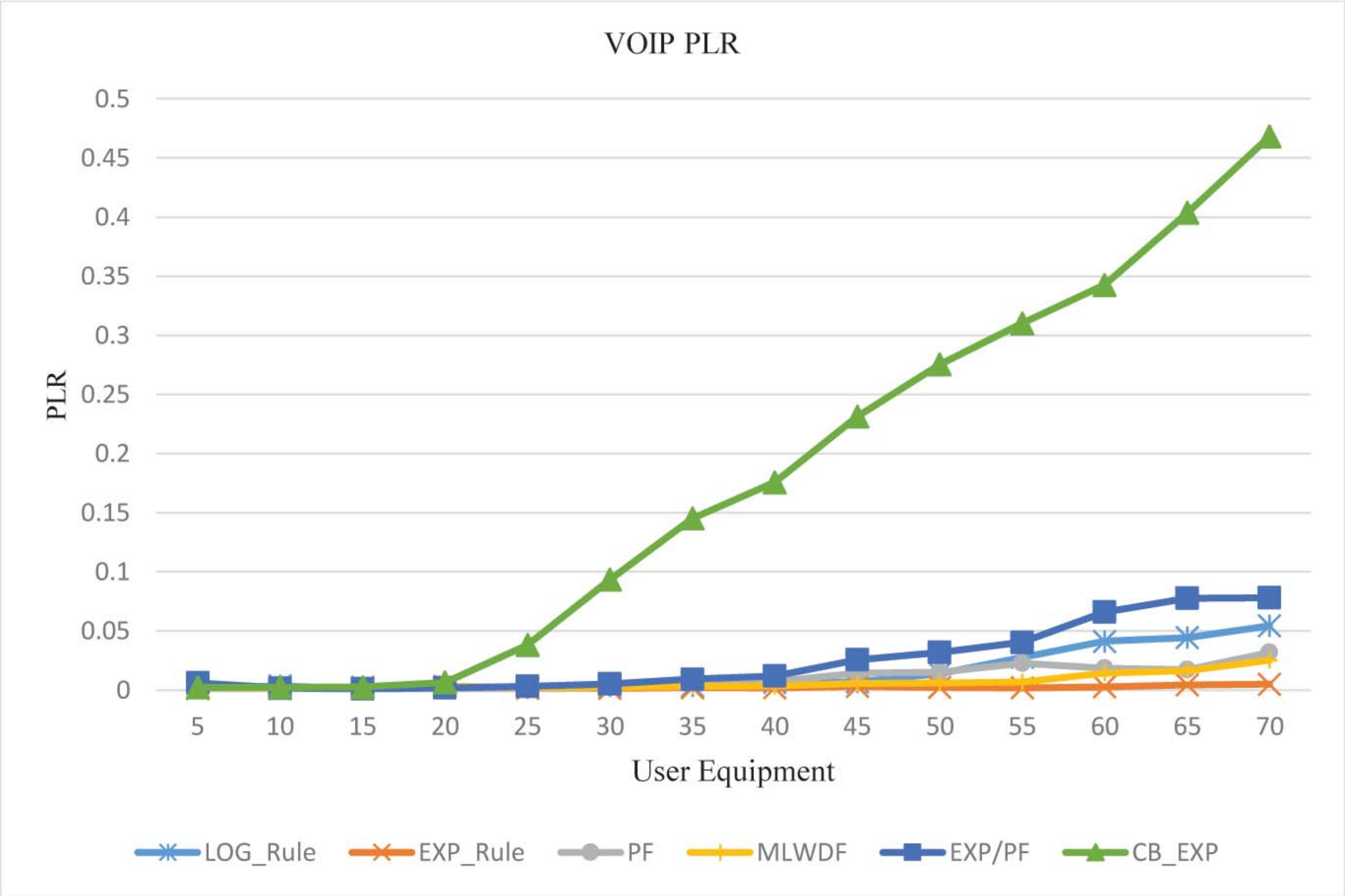


Fig. 9 Comparison of VoIP packet loss rates

In this section, the comparison is assumed to be the case that the serving cell receives no interference from adjacent cells. As mentioned, the three MLWDF, LOG-RULE and EXP-RULE are channel and QoS aware, but the PF metric can only monitor the status of the channel and is unaware of the QoS. Simulations were performed ten times for different methods and then averaged.

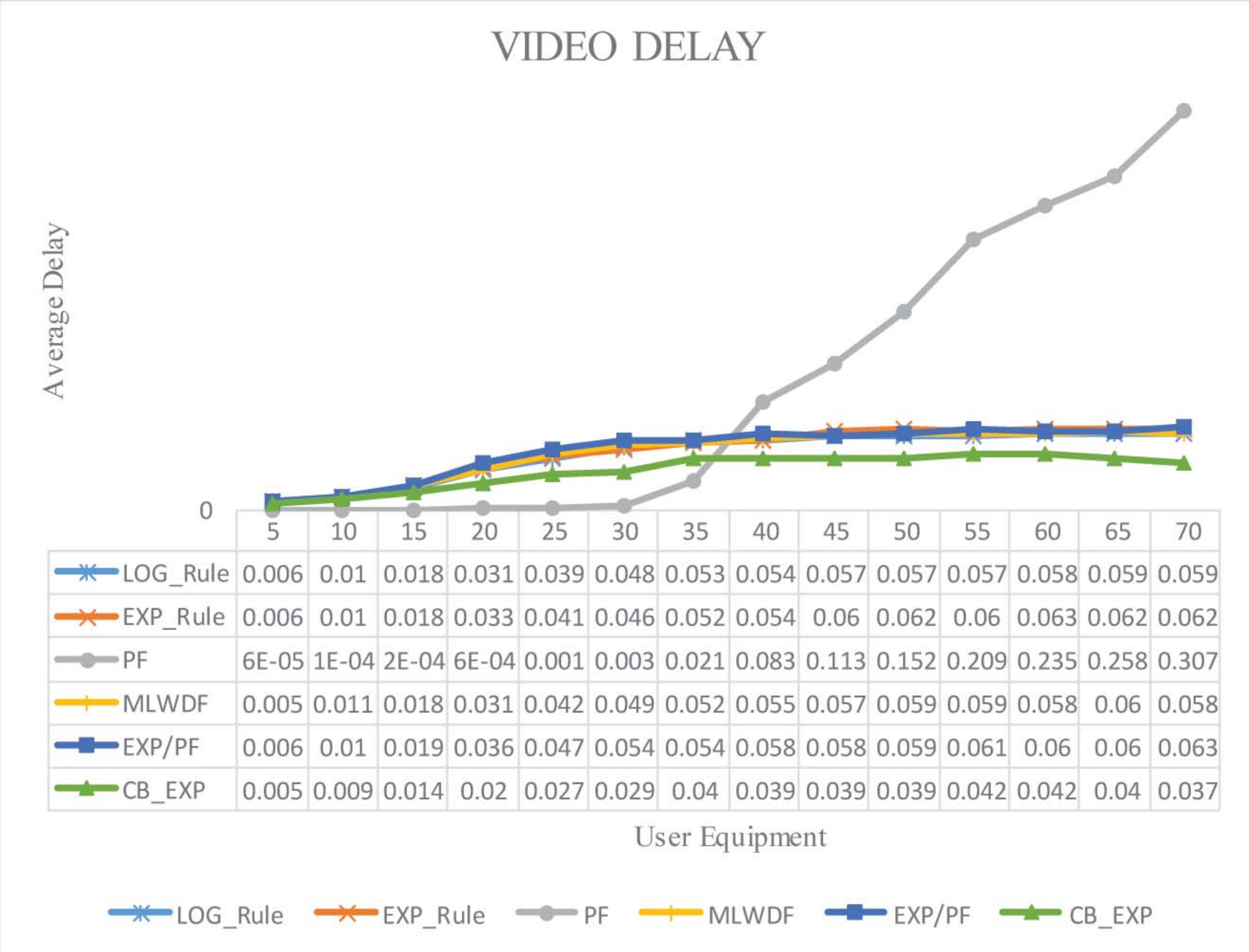
Fig. 8 shows a comparison of the video packet loss rate for proposed CB_EXP and existing metrics. It is shown that CB_EXP metric has better performance than all algorithms except EXP-RULE. For the number of users equal to 70, the proposed metric improves video packet loss rate compared to LOG-RULE, EXP / PF and PF by 11.3%, 17% and 30%, respectively. But its performance is 14% degraded compared to EXP- RULE. As discussed in the last section, in a scenario of serving cell with interference, proposed metric performs the best. In the single cell scenario with interference, it was concluded that proposed metric has better performance rather tEXP-RULE, especially for number of users more than 45.

The reason for the poor performance of the proposed metric in the second scenario, single cell without interference, is as follows. Based on real time channel conditions in this scenario and since SINR ratio for each user is suitable, the proposed metric decision for resource allocation based on channel status cannot make a significant distinction between users. Also, MT expression value is almost the same for all users. However in EXP Rule, given the average resource history allocated to each user, the user who has a higher resource receiving history, should receive less resources at the next TTI.

Fig. 9 shows the comparison results of VoIP packet loss rate obtained for each metric in scenario2. As depicted, the proposed CB_EXP metric does not improve VoIP packet loss rate. The main reason is due to the goal of improving VoIP delay and video packet loss for CB_EXP. Naturally, by reducing the VoIP packet delay, the VoIP packet loss rate increases. On the other hand, other existing metrics in this scenario are generally designed to support VoIP traffic meaning that their performance is improved specially for VoIP.

Fig. 10 shows the comparison results of video packet delay for proposed CB_EXP metric and other existing metrics in scenario2. As demonstrated, the proposed metric outperforms the other metrics remarkably. In other words, the proposed metric is the more appropriate for the case where video packets need low delay like online game applications. The maximum delay for online game application is less than 50 ms in order to have real time interactivity [29, 30].

Fig. 11 shows the comparison results for VoIP average delay. As depicted, proposed CB_EXP metric outperforms other metrics significantly. For 70 users, the proposed metric has improved the VoIP packet delay by 11%, 15%, 103% and 32% compared to MLWDF, LOG_Rule, EXP / PF and EXP_Rule, respectively.



	5	10	15	20	25	30	35	40	45	50	55	60	65	70
LOG_Rule	0.006	0.01	0.018	0.031	0.039	0.048	0.053	0.054	0.057	0.057	0.057	0.058	0.059	0.059
EXP_Rule	0.006	0.01	0.018	0.033	0.041	0.046	0.052	0.054	0.06	0.062	0.06	0.063	0.062	0.062
PF	6E-05	1E-04	2E-04	6E-04	0.001	0.003	0.021	0.083	0.113	0.152	0.209	0.235	0.258	0.307
MLWDF	0.005	0.011	0.018	0.031	0.042	0.049	0.052	0.055	0.057	0.059	0.059	0.058	0.06	0.058
EXP/PF	0.006	0.01	0.019	0.036	0.047	0.054	0.054	0.058	0.058	0.059	0.061	0.06	0.06	0.063
CB_EXP	0.005	0.009	0.014	0.02	0.027	0.029	0.04	0.039	0.039	0.039	0.042	0.042	0.04	0.037

Fig. 10 Comparison of video packets average delay

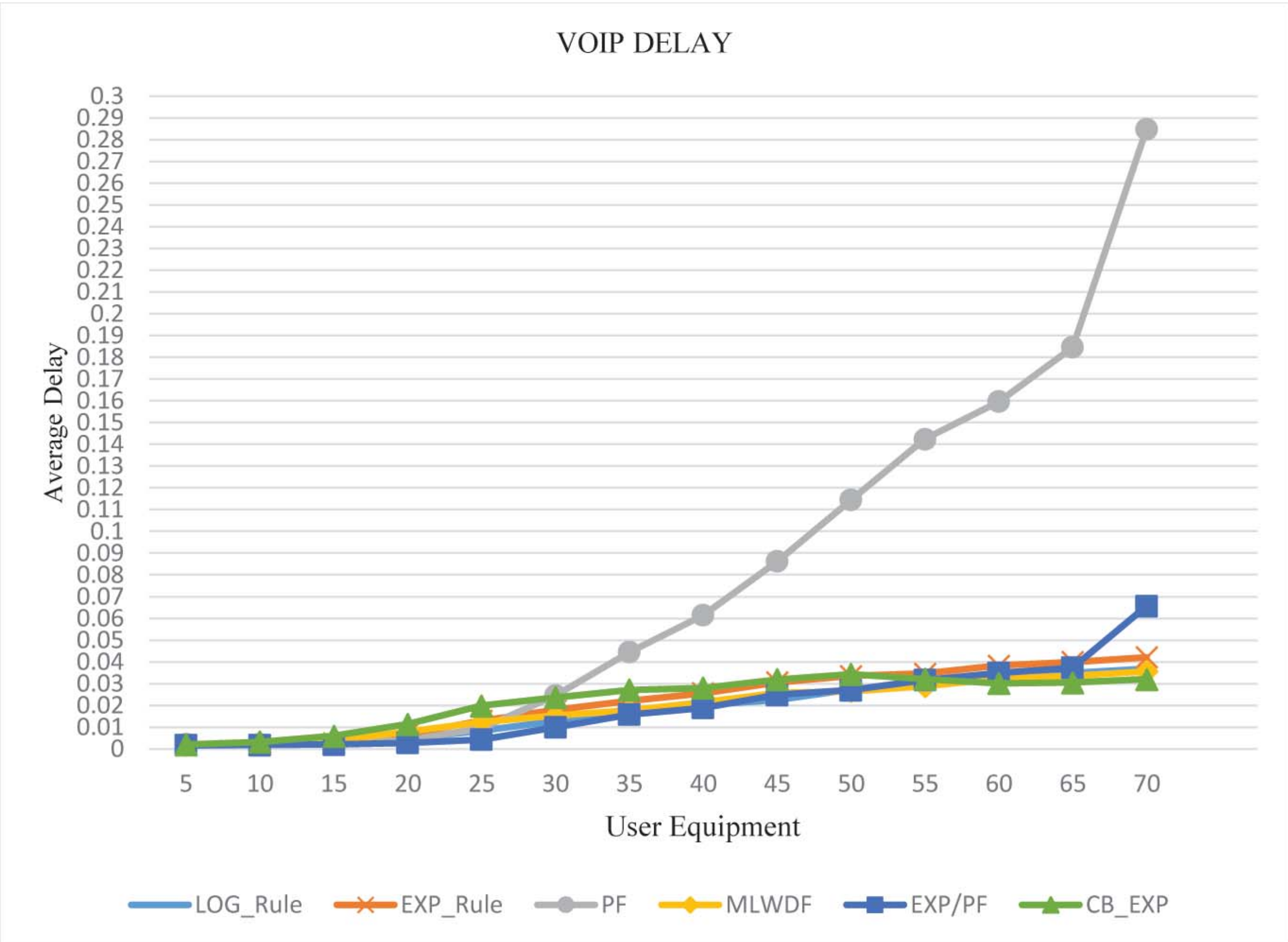


Fig. 11 Comparison of VOIP packets delay

CONCLUSION AND SUGGESTION

Due to the increasing number of cellular telecommunication networks users as well as limited availability of physical resources such as bandwidth, the need for efficient and high performance networks forces network designers to define effective standards for LTE. In this IP based network, many constraints exist for resource management and allocation policies. One of these most important policies is scheduling the service to various user packets. The new and specific form of LTE radio access network requires its own scheduling algorithms. Unlike networks with time-invariant channel, such as optical fiber-based networks, the time-varying feature of LTE channel, should be considered in scheduling. On the other hand, it is also necessary to consider the QoS requirements. Therefore, a suitable metric for LTE scheduling, should consider both simultaneously. Most of the existing metrics, are proposed based on focusing on one parameter for a particular traffic. Given that parameters such as delay and packet loss rate are usually in opposite performance direction, blind focus on improving one parameter may cause another parameter to degrade. On the other hand, according to the cell state in terms of interference, there was no specific classification for using a particular algorithm and scenario.

Based on these shortcomings, to improve the quality parameters, especially for real time video and VoIP, a scheduling metric is proposed in this paper, considering traffic type and channel conditions. This metric, gives higher priority to delayed video packets before expiring. Also, this metric performs reasonably well in the interference scenario by keeping the delay of the VoIP packets in an acceptable range.

Suggestions for future research are as follows: Presenting and classifying different suitable metrics for specific applications, improving scheduling metrics considering client buffer, switching between different metrics in different condition, comparing scheduling metrics performance results for different user speeds.

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