PACKET SCHEDULING STUDY FOR HETEROGENEOUS TRAFFIC IN DOWNLINK 3GPP LTE SYSTEM

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ABSTRACT

Long Term Evolution (LTE) network deploys Orthogonal Frequency Division Multiple Access (OFDMA) technology for downlink multi-carrier transmission. To meet the Quality of Service (QoS) requirements for LTE networks, packet scheduling has been employed. Packet scheduling determines when and how the user's packets are transmitted to the receiver. Therefore effective design of packet scheduling algorithm is an important discussion. The aims of packet scheduling are maximizing system throughput, guaranteeing fairness among users, andminimizing either or both PacketLoss Ratio (PLR)and packet delay. Inthis paper, the performance of two packet scheduling algorithms namely Log Maximum-Largest Weighted Delay First (LOG-MLWDF) and Max Delay Unit (MDU), developed for OFDM(Orthogonal Frequency Division Multiplexing)networks, has been investigated in LTE downlink networks, and acomparison of those algorithmswith a well-known scheduling algorithm namely Maximum-Largest Weighted Delay First(MLWDF) has been studied.The performance evaluation was in terms of system throughput, PLR and fairness index. This study was performed forboth real time (voice and video streaming)and non-real time (best effort)perspectives. Results show that for streaming flows,LOG-MLWDF shows best PLR performance among the considered scheduling schemes, and for best effort flows, it outperforms theother two algorithms in terms of packet delay and throughput.

KEYWORDS

Packetscheduling, LTE,DSA, PLR, throughput

1. INTRODUCTION

Long Term Evolution (LTE) is a development by the Third-Generation Partnership Project (3GPP) to meet the needs of theInternational Mobile Telecommunication Union (ITU). Some of its key features include increased data rate, a scalable bandwidth, increased coverage and capacity, and reduced latency that result in better Quality of Services (QoS) in communication. It employs OFDMAfor downlink transmission. LTE radio access network consists of eNodeBs, which are responsible for radio resource management function including packet scheduling. Packet scheduling is responsible for smart selection of user packetsand allocation radio resources

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appropriately. There are two main categories of multimedia services: Real-time (RT) and Non-Real Time(NRT) services. RT services areeither delay-sensitive (e.g. VoIP) or loss-sensitive (e.g. buffered streaming) or both(e.g.video conferencing). Best effort services, anexample of NRT services,have higher delay requirements and spare resourcesare allocated to them. Dynamic subcarrier allocation, which is dependent on Channel Quality Information (CQI), has been broadly investigated for single carrier and multicarrier wireless networks in [1-6] and [12-14]. Maximising system throughput sometimes resultsin unfair allocation to the users located far from the Base Station (BS) or the users suffering from poor channel condition. Therefore a sufficient trade-off between throughput and fairness is essential. In [5], time, frequency and multiuser diversity in OFDM networks that lead to considerable efficiency have been discussed. Furthermore, cross-layer optimization based on utility function has also been introduced. The scarce bandwidth, fading channels and tough QoS requirements of users make resource allocation a demanding problem.

1.1. Related Work

For different service demands, a number of scheduling plans have been suggested.Three single-carrier packet scheduling algorithms such as Round Robin(RR), Maximum Rate(Max-Rate) andProportional Fair (PF) have been investigated in[7],[8] and [9].MLWDF and Exponential Rule (EXP) were developed in [6] and [10] considering packet delay in Guaranteed Bit Rate(GBR) services over Code Division Multiple Access-High Data Rate (CDMA-HDR) systems.Channel-Dependent Earliest Due Dead line (CD-EDD) [11] algorithm in a mobile cellular system has been developed to study different delay requirements in GBRservices.In [6], M-LWDF scheduling employs both channel and delay information to prevent long queuing delay. In [12] and [13], the Max Delay Unit (MDU)scheduling scheme has been proposed in OFDM networks, and its performance has been compared withthe performance of MLDWF in [14]. In [15], LOG-MLWDF has been proposed and its performance has been investigated and compared withprevious methods proposed forOFDM networks.

Performance of MLWDF has been investigated in downlink 3GPP LTE system in [16, 17]. In this paper, three methods of packet scheduling will be investigated such as MDU, MLWDF and LOG-MLWDF suggested for 3GPP LTEsystem.

The rest of this paper is organized as follows.The formulation of the problem and packet scheduling algorithms are given in Section 2.The simulation environment and results are discussed in Section 3 and Section 4 respectively. Finally, Section 5 contains the conclusions.

2. SYSTEMMODEL AND PROBLEM FORMULATION

System model in 3GPP LTE downlink and packet scheduling algorithms have been described in this section.

2.1. System Model

In 3GPP LTE system a pair of Resource Blocks (RBs) called Physical Resource Block (PRB), consisting of frequency and time domain resources,are assigned to users based on packet scheduling algorithms. The bandwidth of each RB is 180 kHz in frequency domain which is composed of 12 sub-carriers. The duration of a RB is 0.5msand consists of 7 OFDM symbols when normal prefix is used. Packet scheduling is carried out at eNodeB in 1ms intervals and two consecutiveRBs are allocated to a user in that duration. At each time interval,users transmit their Channel State Information (CSI) to eNodeB within a RB.In this work, it has been assumed that the CSI report is based on the calculated SNR (Signal to Noise Ratio) value of the subcarrier at the central frequency of each RB, and assumed to be free of errors. All subcarriers of a RB are assumed to have the same SNR. According to the reported SNR, the eNodeBperforms scheduling for all users. In this paper, it has been assumed that all usershave buffers of infinite size in which packets are queued for transmission on a first in first out (FIFO) basis. The selection of the user (at each time slot and on each RB) is based on reported SNR and the priority (delay) of the user. Utility functions are used for selection of users based on SNR and delay. Figure 1 presents the architecture of the proposed scheduling schemes. The number of bits in two subsequentRBs is determined based on reported downlink SNR. In[18],the number of bits per symbol of user i at time t on a subcarrier within RB_j $(nbits_{i,j}(t)/symbol)$is used to compute the data rate of the user at time t.$dr_i(t)$,iscalculated according to thefollowingformula:

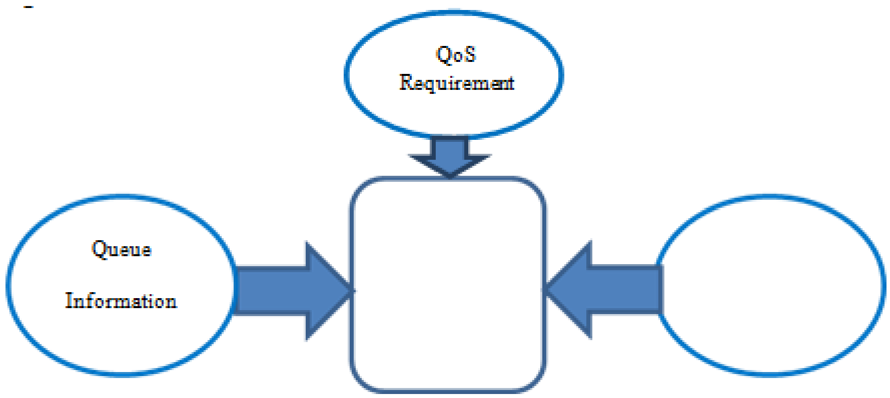

Figure1. Architecture of the proposed algorithms

$$dr_i(t) = \frac{nbits_{i,j}(t)}{symbol} * \frac{nsymbols}{slot} * \frac{nslots}{TTI} * \frac{nsc}{RB} \qquad (1)$$

Where

$\frac{nslots}{TTI}$is the number of slots per TTI,

$\frac{nsc}{RB}$is the number of subcarriers per RB.

In Table1, minimum instantaneousdownlink SNR values and the related data rates have been given. User’s priority is decided by packet scheduler based on channel condition, HOL(Head OfLine) packet delays and quality of service, etc. HOL packet delay is calculated by taking the difference of the current and arriving time of a packet. If the HOL delay exceeds a specific threshold then the packets will be discarded.

Table1. Immediate downlink SNR and data rate

MinimumInstantaneous Downlink SNR Value(dB)	Modulation and Coding	Data Rate(kbps)
1.7	QPSK(1/2)	168
3.7	QPSK(2/3)	224
4.5	QPSK(3/4)	252
7.2	16QAM(1/2)	336
9.5	16QAM(2/3)	448
10.7	16QAM(3/4)	504
14.8	64QAM(2/3)	672
16.1	64QAM(3/4)	756

2.2. Packet Scheduling Algorithms and Problem Formulation

In this section, the mathematical model for MDU, M-LWDF and LOG-MLWDF scheduling algorithms are described. MDU and LOG-MLWDF have been described in [12] and [15] for OFDM networks and are suitable candidates to be used in 3GPP LTE networks.Model of the system includes one BS and M users. A multipath fading channel is assumed to be the model of wireless simulated channel. The formula given below explains the channel impulse model:

$$h_i(t,\tau) = \sum_k \gamma_{k,i}(t)\delta(\tau - \tau_{k,i}). \qquad (2)$$

Where $\tau_{k,i}$is the delay of kth path and $\gamma_{k,i}(t)$ is the complex amplitude that is widestationary,narrowband and Gaussian random process. The paths are independent from each other. The channel frequency response is:

$$H_i(f,t) = \sum_k \gamma_{k,i}(t)e^{-j2\pi f\tau_{k,i}} \qquad (3)$$

Signal to Noise Ratio (channel gain) for user i, is as follow:

$$\rho_i(f) = \frac{|H_i(f)|^2}{N_i(f)} \qquad (4)$$

Where$N_i(f)$ is the noise power density.

Achievable rate of user i at frequency f,$C_i(f)$, (assuming BER and power density to beconstant) is presented as follow:

$$C_i(f) = \log_2\left(1 + \frac{\beta p(f)|H_i(f)|^2}{N_i(f)}\right) = \log_2(1 + \beta p(f)\rho_i(f))\,\frac{\frac{bits}{sec}}{Hz} \qquad (5)$$

Where:

$$\beta = \frac{1.5}{-\ln(5BER)}.$$

The throughput of user *i*is calculated as follows:

$r_i = \int_{D_i} C_i(f)\mathrm{df}.$

DSA(Dynamic Subcarrier Allocation)has been considered in this paper as the resource allocation scheme to dynamically allocate RBs to the users in order to maximize the average utility function as below:

$$Max\frac{1}{M}\sum_{i=1}^{M} U_i(r_i),$$

Subject to:

$$\bigcup_{i\in M} RB_i = [0, \beta],$$

and$RB_i \cap RB_j = \emptyset,\ i \neq j\ , \forall i, j \in M.$

Assignment of one RB does not have any impact on the assignments of other RBs. Utility functions can be linear or nonlinear functions of the rate. DSA method can be used inthe following formula when the utility functions are linear:

$$m(k,n) = argmax_{i\in\mu}\{U_i^{'}C_i[k,n]\} \qquad (6)$$

Where $m(k,n)$ indicates that RB_k is assigned to user m at time slot n, and $C_i[k,n]$ is the feasible data rate for RB_k at time slot n, and it is totally calculated by CQI. With nonlinear utility function RBs cannot be allocated independently andin that case, DSA method would be very complicated. When the utility functions are concave, iterative algorithms are used.

2.2.1. MDU Algorithm

MDU[12] deploys the idea that utility function can be calculated based on average waiting time and it improves the quality of service idea. The MDU scheme formula is given below:

$$max_{RB^n, i\in A^n, s\in B^n} \sum\nolimits_{i\in A^n, s\in B^n} \frac{|U_{i,s}^{'}(w_i)[n]|}{\lambda_{i,s}} r_i[n] \qquad (7)$$

Subject to the following conditions:

$\bigcup_{i \in A^n} RB_i^n \subseteq K$,

$RB_i^n \cap RB_j^n = \emptyset, i \neq j \;, \forall i, j \in A^n$.

$\lambda_{i,s}$: Average arrival rate of user i with the traffic type s,

$r_i[n]$: Data transmission rate of user i at time slot *n,*

$C_i[k,n]$: Achievable transmission rate on subcarrier RB_k for user *i,*

RB_i^n: Set of subcarriers assigned to user i at time slot *n,*

A^n: Set of queues at time slot *n,*

B^n: Set of the service types.

Three types of traffic such as voice, streaming and best effort have been considered in this work. Consecutive utility functions are used in MDU scheduling:

The marginal utility function for voice users is as below:

$$\left|U'_{i,v}(w_i)\right| = \begin{cases} w_i, w_i \leq 2.5ms \\ w_i^{1.5} - 2.5^{1.5} + 2.5, w_i \geq 2.5ms \end{cases} \tag{8}$$

The marginal utility function for streaming users is given by:

$$\left|U'_{i,s}(w_i)\right| = \begin{cases} w_i^{0.6}, w_i \leq 5ms \\ w_i - 5 + 5^{0.6}, w_i \geq 5ms \end{cases} \tag{9}$$

The marginal utility function for best effort users is as follows:

$$\left|U'_{i,s}(w_i)\right| = \begin{cases} w_i^{0.5}, w_i \leq 5ms \\ 5^{0.5}, w_i \geq 5ms \end{cases} \tag{10}$$

2.2.2. MLWDF Algorithm

The QoS requirements have been indicated in MLWDF[6] scheduling weights as below:

$$a_i = -\frac{\log \delta_i}{T_i},$$

$P_r\{W_i > T_i\} \leq \delta_i$.

Where

T_i:The delay threshold,

δ_i:The maximum probability of exceeding T_i,

W_i:The packet delay of user i.

MLWDF utility function is given by

$U_i = \gamma_i W_i(t) r_i(t)$(11)

Subjected to:

$\gamma_i = \frac{a_i}{\bar{r}_i}$.

W_i:The head of line packet delay for user i,

r_i(n): The channel rate of user i,

$\overline{r_i}$ (n): Average channel rate of user i.

When the precedence of service is high and the ongoing packet delay is considerable, the probability of obtaining service increases.

2.2.3. LOG-MLWDF Algorithm

LOG-MLWDF[15] is an extension of MLWDF algorithm in which logarithmic weight of the average delay has been added to MLWDF formula as below:

$$U_i = \gamma_i r_i(t)\left(W_i + log\left(Wave_i(t)\right)\right)^{1.5} \qquad (12)$$

Where:

$Wave(t) = \frac{1}{T}\sum_{i \epsilon N} W_i\,(t)$.

Table 2.3GPP LTE Downlink system parameters

Parameters	Values
Carrier Frequency	2GHz
Bandwidth	5MHz
Number of Sub-carriers	300
Number of RBs	25
Number of Sub-carriers per RB	12
Sub-Carrier Spacing	15kHz

Slot Duration	0.5ms
Scheduling Time(TTI)	1ms
Number of OFDM Symbols per Slot	7

3. SIMULATIONENVIRONMENT

In this paper a single cell of radius 150m with downlink parameters in Table 1 has been modelled to study the performance of MDU, MLWDF and LOGMLWDF. The model which is described in previous section has been considered in this work. A cell of 5MHz bandwidth with 25 RBs and 2 GHz carrier frequency is modelled. The parameters of the system are presented in Table 2.Two scenarios are considered in this research. In the first scenario, there are 40 voice and61 to 191 streaming users distributed uniformly within the simulation area.In the second scenario 1 to 36 best effort usershave been added to the network. Users are constantly moving between [1-100] km/h speeds in random directions. To guarantee that the users always remain in the cell border's area, a wrap-around method is used. The SNR reports to the serving eNodeBare assumed to be instantaneous and free of delay and error. The buffer size for all the streaming and best effort users is assumed to be infinite.On-Off activity model has been chosen for modelling the voice source as conversational voice model. The average duration of 1s and 1.35s has been considered for exponentially distributed interval model of the voice. 32kbps digital voice coding is generated within each talk spurt (on).In Table 3, the parameters of simulated streaming video (with 128kbps rate) have been given. MLWDF, LOG-MLWDF and MDU methodshave been simulated for 3000 time slots (1ms each) usingMATLAB.When a packet's delay is more than the waiting time threshold, it is considered to bediscarded. The permissible waiting time of apacket in eNodeB buffer (considered as the threshold of HOL packet delay), is set to 20ms, 10ms and 100ms for streaming, voice and best effort users respectively.

Table 3.Parameters of video streaming services [20]

Information type	Distribution	Distribution Parameters
Inter-arrival time between the beginning of successive frames	Deterministic (Based on 20fps)	50ms
Number of packets(slices) in a frame	Deterministic	8
packets(slice) size	Truncated Pareto (Truncated Pareto (Mean=50bytes, max=125bytes)	K=40bytes, $\alpha = 1.2$
Inter-arrival time between packets(slices) in a frame	Truncated Pareto (Mean=6ms, Max=12.5ms)	K=2.5ms, $\alpha = 1.2$

Path loss, shadowing and Rayleigh fading have been considered forcombined path loss and shadowing model [19]. BER is assumed to be 10^{-6} for this paper.

To evaluate the performance of packet scheduling algorithms,system throughput, Packet Loss Ratio (PLR) and fairness have been used as performance metrics. Acomparison among thealgorithms is made based on these metrics. Throughput of the system is defined as the size of correctly received packets (in bits) during thesimulation time.The formula is as below:

$$averagethroughput_i = \frac{1}{T}\sum_{i=1}^{N}\sum_{t=1}^{T} ptransmit_i(t) \qquad (13)$$

Where T is total simulation time for user i,N is number of users in service time and $ptransmit_i$ is total size of correctly received packets.PLR is defined as the ratio of total size of discarded packet to the total size of all packets that were received in the buffer of eNodeB during the interval. PLR of a system should be below thethreshold value to cover the QoS needs of a service. It is calculated according to following equation:

$$PLR = \frac{\sum_{i=1}^{N}\sum_{t=1}^{T} pdiscard_i(t)}{\sum_{i=1}^{N}\sum_{t=1}^{T} psize_i\,(t)} \qquad (14)$$

Where, $\mathrm{pdiscard}_i(t)$ is the total size of discarded packets(in bits)of user i in the simulation time, $\mathrm{psize}_i(t)$ is the total size of all packets(in bits) that reaches the eNodeB buffer of user i at simulation time T.N is total number of users and T is the simulation time.

Fairness is defined here as the difference between the most and least transmitted packets of two users divided by the total size of all packets that have arrived to the buffer of eNodeB. The formula is as below:

$$fainess = 1 - \frac{ptotaltransmit_{max} - ptotaltransmit_{min}}{\sum_{i=1}^{N}\sum_{t=1}^{T} psize_i(t)} \qquad (15)$$

In table 4, Guaranteed Bit Rate(GBR) and Non-GBR(NGBR) services and their different level of QoS requirements has been presented.

Table 4.Standardized QoS Class Identifiers (QCIs)[21]

	Resource Type	**Priority**	**Packet Delay Budget**	**Packet Error LossRate**	**Example Services**
1	GBR	2	100ms	10^{-2}	Conversational Voice
2		4	150ms	10^{-3}	Conversational Video(Live Streaming)
3		3	50ms	10^{-3}	Real Time Gaming
4		5	300ms	10^{-6}	None-Conversational Video (Buffered Streaming)
5	Non-	1	100ms	10^{-6}	IMS Signal

6	GBR	6	300ms	10^{-6}	Video (Buffered Streaming), TCP-based (e.g. email, chat..)
7		7	100ms	10^{-3}	Voice, Video (Live Streaming),Interactive Gaming
8		8	300ms	10^{-6}	Video(Buffered Streaming)TCP-based (e.g., www,e-mail, chat..)
9		9			

4. SIMULATION RESULTS

In this section, the simulation results are analysed. Figures 2 to 6 are related tothe first scenario in which there are no best effort users in the network. Figure2 and Figure 3 show the system throughput graphs for streaming and voice users of the three packet scheduling algorithms. Throughput of streaming users are the same in MLWDF and LOG-MLWDF algorithms (as seen in Figure 2), whereas for voice users, MLWDF has better throughput performance compared with LOG-MLWDF and MDU methods (as shown in Figure 3).

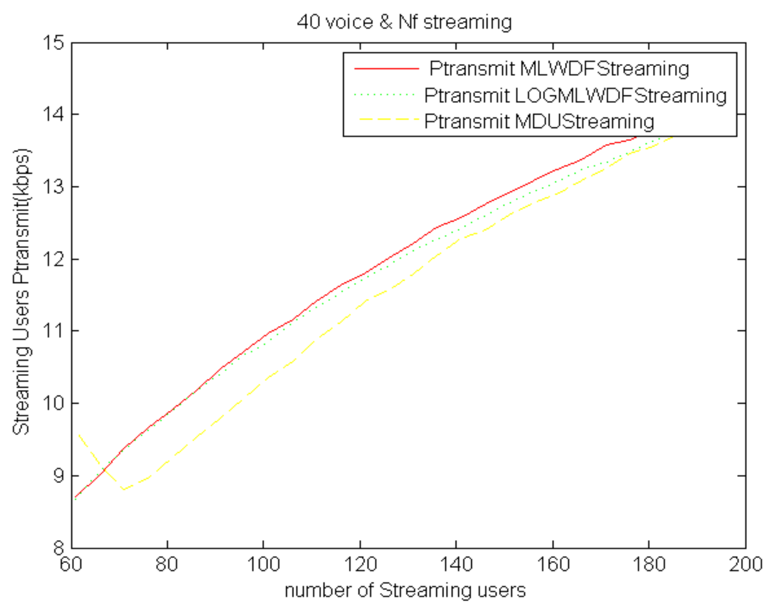

Figure 2.System throughput for streaming users vs. numberof steaming users

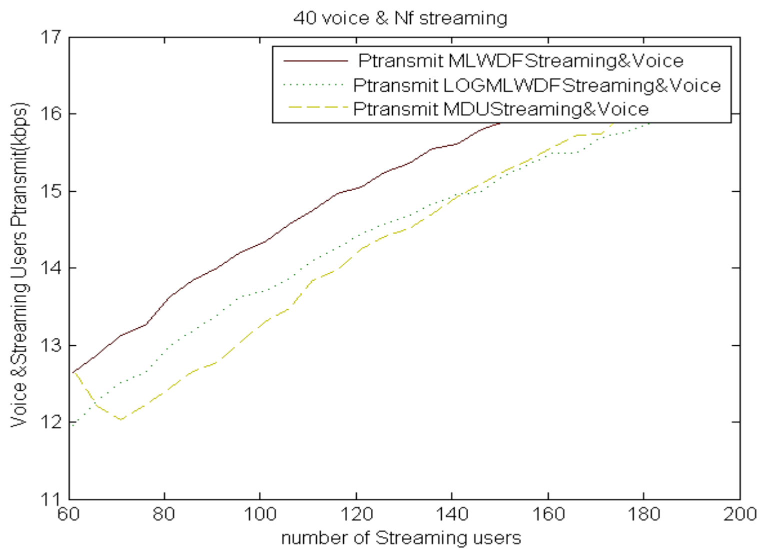

Figure 3.System throughput for streaming &voice users vs. number of streaming users

Figure 4 and 5showPLR performance for the three algorithms. It can be observed from Figure 4 that for all streaming users, LOG-MLWDF has satisfied thePLR threshold of10^{-3}[21]. With increasing number of streaming users, MLWDF and MDU have greaterPLR than LOG-MLWDF for streaming users. For voice users, LOG-MLWDF has greaterPLR than the other twoalgorithms (Figure 5) even thoughthe PLR is still below the permissible threshold for conversational voice of 10^{-2}[21]. The MLWDF followed by LOG-MLWDF have the highestPLR for streaming and voice users.

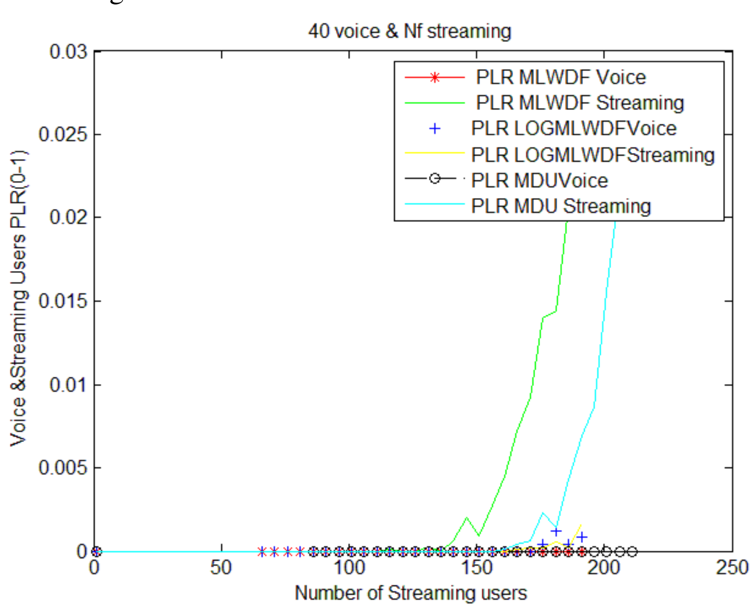

Figure 4. Packet Loss Ratio vs. number of streaming users

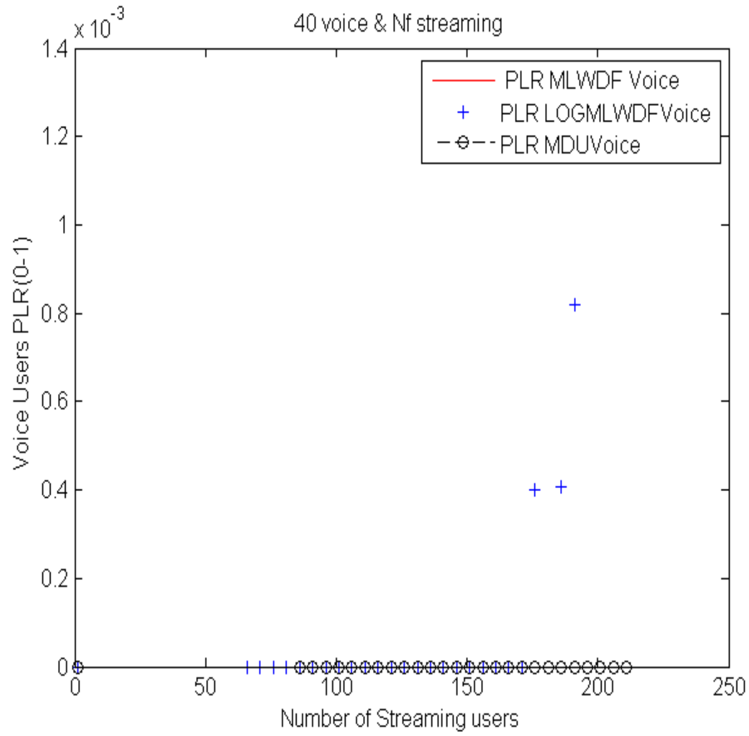

Figure5. Packet Loss Ratio vs. number of streaming users

In Figure 6 the fairness of the three algorithms has been compared. LOG-MLWDF has the worst fairness performance among the three methods. The fairness performance of MLWDF is slightly better than MDU algorithm when the number of users is low but it is slightly lower than MDU algorithm when the number of users is high.

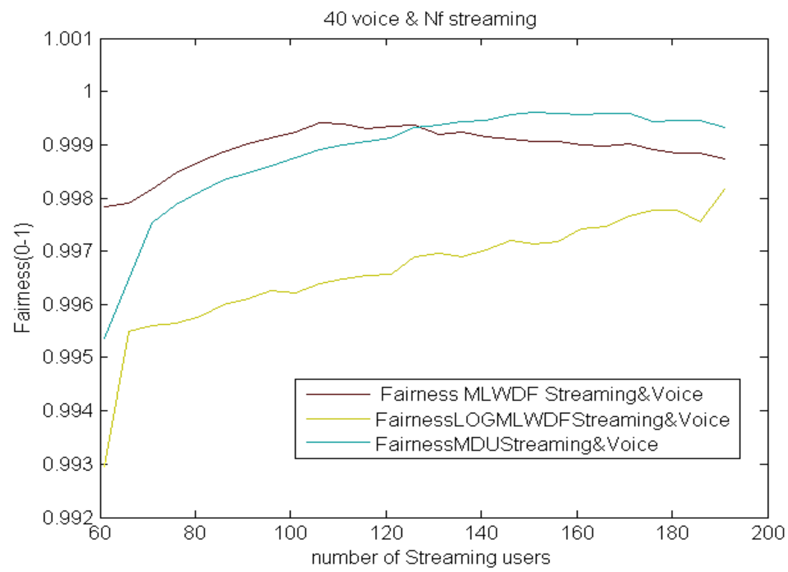

Figure 6.Fairness vs. number of streaming users

Considering the second scenario in which best effort users are added to the previous network simulation. The simulation results are given in Figure7 and Figure 8.From the percentile delay and throughput perspective of best effort users, it can be seen that LOG-MLWDF outperforms the previous algorithms.

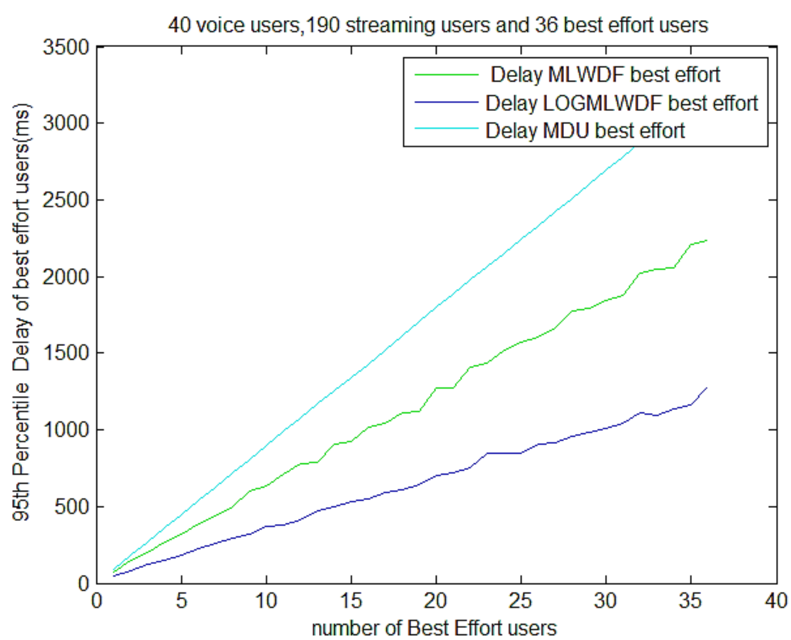

Figure 7. 95 percentile delay of best effort users vs. number of users

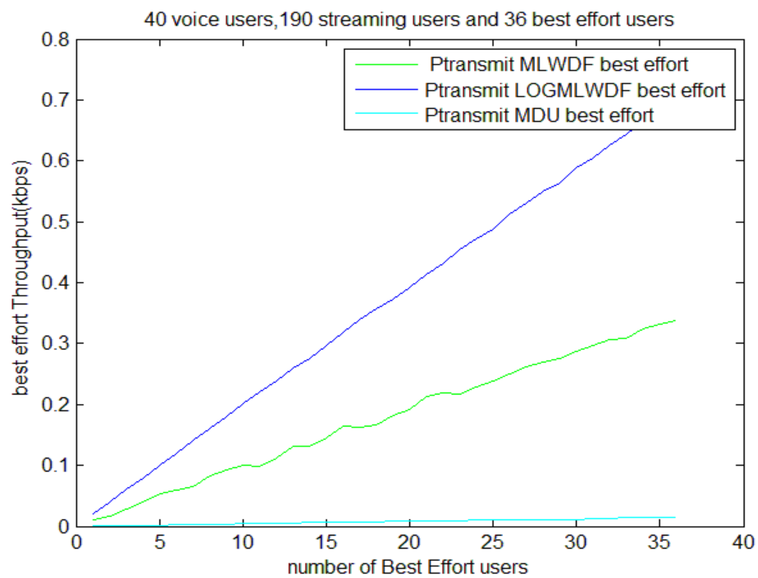

Figure8. Best effort throughput vs.number of users

In Figure 9, the three algorithms have been compared according to the fairness metric. It can be seen that LOG-MLWDF outperforms the other two algorithms and MLWDF has the least fairness compared with other algorithms.

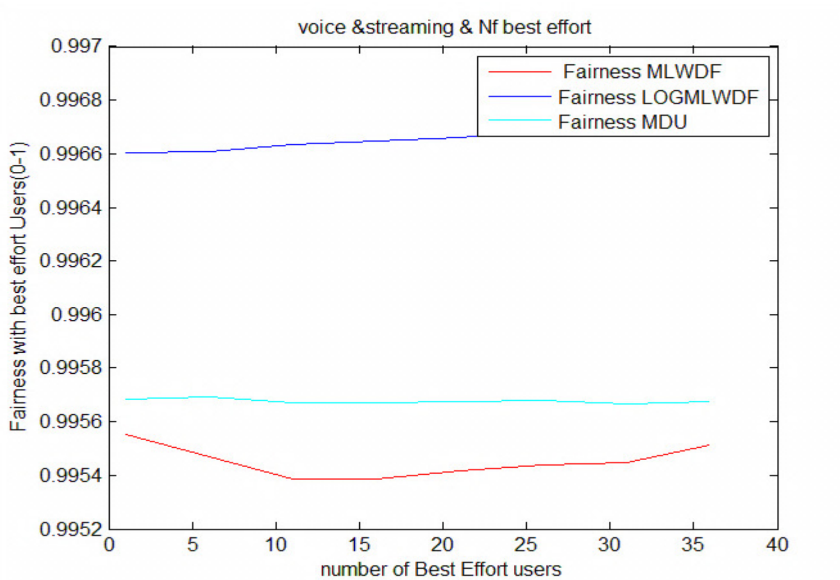

Figure 9. Fairness vs. number of users

5. CONCLUSIONS

This paper investigates the resource management of three scheduling algorithms in a downlink 3GPP LTE system. A comparison between well-known algorithms MDU,MLWDF and LOGMLWDF has been made considering two scenarios. This paper also analyses the strengths and weaknesses of LOG-MLWDF compared with the other two algorithms. In the first scenario, LOG-MLWDF shows better PLR performance for streaming users and its PLR performance remains below the threshold value for voice users.With the addition of best effort users it is apparent that LOG-MLWDF outperforms the two other algorithms in terms of 95 percentile delay, fairness of the wireless network and throughput for best effort users. In future work different scheduling scheme with the concept of Hybrid Automatic Repeat Request (HARQ) with different simulation time and more scenarios will be investigated.

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