ISSN: 2449 – 0539

BAYERO JOURNAL OF ENGINEERING AND TECHNOLOGY (BJET) VOL.14 NO.2, pp 81-91 AUGUST, 2019

Development of an Improved Modified Largest Weight Delay First Algorithm for Long Term Evolution (LTE) Network.

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Abstract-- **Long term evolution (LTE) is one of the fastest growing technologies which support different type of applications . One of the key radio resources management mechanisms in the LTE network is the radio scheduler, which coordinate the access to radio resources, the decision of scheduling scheme play a major role in end to end LTE network performance and user Quality of service (QoS). Many resource scheduling scheme have been proposed and implemented in time past, among which is the modified largest weight delay first (MLWDF) algorithm which is suitable for real time and non-real time applications, this algorithm have been found not to support application sufficiently at overload network condition this is because it consider only head of line packet delay (HOL) as QoS parameter in making scheduling decisions which is not sufficient enough to improve overload network performance. This research evaluate some of the well-known algorithm in literature and make an improvement on the MLWDF by first adopting the mathematical model of MLWDF algorithm and then make an improvement on the adopted model by incorporating a scheduling priority ratio which is the ratio of (HOL) and delay deadline together with natural exponential term which help to grow the metrics for the users as their delay tolerance is approaching threshold thereby preventing packet loss and increase throughput in overload network condition. The proposed algorithm was simulated using LTE-Sim simulator and compared with other schedulers such as Flow bandwidth MLWDF for LTE downlink transmission (I-MLWDF), Improved MLWDF scheduler for LTE downlink transmission (Mod_MLWDF) and MLWDF using packet loss ratio, packet delay, throughput, fairness index and cell-spectral efficiency as performance metrics. The result shows that the proposed algorithm improved the video flow throughput at user index 30 by 38.33%, 59.35% and 60.45% and at user index 50 by 47.93%, 77.16% and 77.36% for VoIP flow it improve the throughput at user index 50 by 1.69% and 1.49%, for non-real time flow i.e, IMS signaling the proposed algorithm shows poor performance at user index 10, 30 and 50 by 66.91% ,88.19% and 90.71% respectively.**

Keywords – MLWDF, I-MLWDF, Mod_ MLWDF, QoS, Priority ratio,Scheduling.

INTRODUCTION

In recent times, the number of mobile subscribers and the volume of traffic generated by them have increase drastically. This has brought about the introduction of a packet based broadband system referred to as Long Term Evolution (LTE) networks. The LTE architecture includes the Evolved Universal Terrestrial Radio Access Network (E-UTRAN) and Evolved Packet Core (EPC) network[1]. E-UTRAN is a radio access network of 3GPP's Long-Term Evolution (LTE).It is a new air interface system, which provides High speed data-rate, lower latency and is optimized for packet data. In LTE, Two duplexing schemes are used, time division duplexing (TDD) and frequency division duplexing (FDD).Using LTE-TDD, a single frequency channel is assigned to both the transmitter and the receiver .LTE-FDD requires paired spectrum with sufficient frequency separation to allow simultaneous transmission and reception. The E-UTRAN Network require high speed data-rate and reliable transmissions with bandwidth efficiency .To meet these requirements Multiple input multiple-output(MIMO)system have been implemented in which multiple antennas are used in both transmitter and receiver and up to four antennas can be used by a single LTE cell [2] .The E-UTRAN architecture consists of eNodeBs that interfaces with the user equipment (UE) [3].LTE network provides spectrum flexibility where the transmission bandwidth can be selected between 1.4 MHz and 20 MHz depending on the available spectrum[1]. The peak data rate, which is the important parameter by which different technologies are usually compared, generally depends on the amount of spectrum used. The allowed peak data rate for the DL and UL is equal to 100 Mbps and 50 Mbps respectively[2,4]. LTE targets to provide spectral efficiency two to four times better than 3G systems (15 bps/Hz in DL and 3.75 bps/Hz in UL), and reduces radio access delays.The LTE network uses Orthogonal Frequency Division Multiple Access (OFDMA) for downlink transmission (OFDMA) and Single carrier frequency division Mutiple Access (SC-FDMA) for Uplink transmission[2].

This research is organized as follows section II contains the review of literature, section III described the general procedure for downlink resource scheduling in LTE network, section IV contained the methodology used to develop the proposed algorithm, in-depth description of the propose model, the traffic model used, the Lte-Sim and the performance metrics, section V contained the performance analysis and section VI handled the conclusion.

LITERATURE REVIEW

The scarce nature of radio resource has led to many packets scheduling algorithm both in uplink and downlink in other to meet up with the QoS requirement of the network users in [1] the authors propose proportional fair packet scheduling algorithm the algorithm was design initially for a Code Division Multiple Access (CDMA) network, in this research a forward link data throughput performance of high data rate wireless access system is presented on the forward link of the propose system,[1,2,4,10-12], In downlink transmission, radio resources are arranged in both frequency and time domains and are referred to as resource blocks (RBs).In the frequency domain, a RB consists of 12 consecutive subcarriers of 15 KHz each (180 KHz total bandwidth) while in the time domain, it is made up of a time slot of 0.5 ms duration. A time slot consists of 7 OFDM symbols which can be either seven (normal cyclic prefix) or six (extended cyclic prefix) [5]. The normal cyclic prefix is used in urban cells and high data rate applications while the extended cyclic prefix is used in special cases like multi-cell broadcast and in very large cells e.g. rural areas, low data rate applications [6].In literature, researches have been done on radio resource allocation in LTE downlink network. Different procedures and decisions have been used to design and test the performance of schedulers. The key design aspects range from complexity, scalability, spectral efficiency, fairness, to QoS provisioning. Depending on the research goal, schedulers prioritize the users based on criteria such as channel condition, packet delay, service type, resource allocation policies. Although, not all the parameters are used at the same time to achieve the set goals. Packet scheduler at radio base station (evolved Node B in LTE specification) is in charge of assigning portions of spectrum shared among users [7]. The performance of the network differs according to the algorithms used for scheduler. Designing an effective scheduler is therefore an important task in order to differentiate the performance of one wireless system from another. The packet scheduler in LTE aims to maximize the spectral efficiency and makes the negative impact of channel quality drops into negligible several packet scheduling algorithms are proposed to support the increasing traffics. This work focus attention on resource scheduling algorithm in downlink of LTE network for real time flow and a satisfactory level of performance for the non real-time flow, an improvement of an existing algorithm i.e, MLWDF is carried out to get the propose algorithm by in cooperating scheduling priority ratio in to the existing algorithm the propose algorithm will be compare to Mod_MLWDF, I-MLWDF and MLWDF in other to test it performance using throughput, delay, packet loss ratio and fairness index as performance metrics.The propose algorithm put delay deadline as well head of line packet into consideration in making scheduling decision this is aim at reducing packet loss due to dealing expiration, keeping the head of line delay bellow the delay threshold and to increase the chances of real time services to get access to radio resources at every transmission time internal. .Data is transmitted to deferent access terminal (AT) in TDM fashion. The rate transmitted to each Access Terminal (AT) is variable and depends on each ATs measured Signal to Interference Plus Noise Ratio (SINR) ATs send to the access point (AP) the index of the highest data rate which can be received reliably. The scheduler at AP terminals determines to be served base on the reported data rate request from the terminals and the of data that has already been transmitted to each terminal. The scheduler attempts to take advantage of temporal variation of the channel by scheduling transmission to ATs during time periods where the ATs see strong signal levels. The scheduler send data to the mobile that has the highest DRC/R, where DRC is the rate requested by the mobile in a giving time slot and R is the average rate received by the mobile over a window of appropriate size, by so doing each user is served in slots in which its requested rate is closer to the peak compared to its recent requests.Suppose there are N users and let R_i(t) be the estimate of the average rate for user i at slot t.

i =1,……,N (2.1)

Also let us suppose at slot t, the current DRC (i.e., request rate) from user i is DRC_i(t), again

i =1,……..,N (2.2)

The algorithm work as follows:

1. Scheduling: The user with highest ratio of DRC_i(t)/R_i(t) out of all N users will receive transmission at each decision time in case of ties among the user they are randomly broken any user for whom there is no data to send is ignored in this calculation.
2. Update Average Rate: For each user i
   R_i(t-1) = (1-1/t_c)R_i(t) + 1/t $_c$(2.3)

A user that is not currently receiving service has zero (0) for his current rate of transmission even users for whom the scheduler has no data to send also get their average rate updated. Also note that the scheduling step is executed each time a new transmission begins but the update average rate is done in each slot, even if the slot is in the middle of a multi-slot transmission.

The size of the time window t_c determine between maximum throughput and satisfying fairness of each users. The drawback of this algorithm is it sole reliance on channel condition in making scheduling decision. Hence, there is need for other scheduling algorithms.

In [8] Comparative study of scheduling algorithms for LTE networks were carried out the authors make an attempt to study and compare the performance of PF,MLWDF and EXP/PF scheduling algorithms, the evaluation is centered on the influence of speed on the performance of the algorithm with respect to Video, VoIP and Best effort flow respectively.

The authors in [9] propose modified largest weight delay first algorithm MLWDF, this algorithm was design to support multiple real time data users in CDMA-HDR system. It combine both channel condition and state of the queue with respect to delay in making scheduling decisions, it ensures that the probability of delay packets does not exceed the discarded bound the maximum allowable packet loss ratio i.e.[2,5,9].

$$P_r\{W_i > \tau_i\} \leq \delta_i$$

Where W_i is user head of line packet delay, τ_i and δ_i are the delay threshold and the maximum probability of exceeding it respectively.

In each time slot the scheduler serve queue j for which equation (2.4) is maximal.

$$M_j^{MLWDF} = \alpha_i W_i \frac{r_i(t)}{\bar{r}_i} \quad (2.4)$$

Where W_i is head of line packet delay for i-th user, $r_i(t)$ is the channel capacity with respect to i-th user, and

$$\alpha_i = -\frac{(\log\delta_i)}{\tau_i} \tag{2.5}$$

$\overline{r_i}$ is the average channel rate with respect to user i .

Despite the ability of the algorithm to increase the throughput of the real-time user it is also accompanied with the following drawbacks the algorithm has totally fail to put into considering the dynamic nature of the users which make them move from one point to the other which in turn effect their channel conditions. The algorithm also fail to give priority to those users whom as a result of movement from one place to another or station in a place with poor channel condition particularly those user that are stations at the cell edge even when their head of line packet delay is tending towards the maximum allowable delay which if not scheduled in current Transmission time interval (TTI) would lost it packet, hence the need for improvement the algorithm.

In [13] the authors propose an algorithm that make an attempt to improving the MLWDF algorithm the improvement was carried out by incorporating user demand variations and hybrid automatic repeat request HARQ prioritization parameter, the authors came up with two model, one for the user demand variation parameter and the other for the HARQ parameter but were not able to simulate that of HARQ because the Lte-Sim version5 used for the simulation does not incorporate HARQ module ,after the simulation of the user variation parameter model it was compared with MLWDF and PF using Video, VoIP and IMS Signaling flows.

$$m_{ik}^{iM-LWDF} = \alpha_i \frac{1}{P_{QCI}} D_{HOL,i} \cdot \frac{[d_{i,k}(t)]}{[R_{i,k}(t)]} \tag{2.6}$$

$$m_{ik}^{iM-LWDF} = \alpha_i \frac{1}{P_{QCI}} D_{HOL,i} \cdot \frac{[d_{i,k}(t)]^{a}}{[R_{i,k}(t)]} \tag{2.7}$$

Equation 2.6 and 2.7 are the user variation and HARQ variation parameters model.

The authors in [14] also make an attempt in making improvement on MLWDF algorithm by incorporating flow bandwidth to the parameter of the average data rate of the user before the current transmission time interval TTI in the MLWDF algorithm model as shown in the equation 2.8.

$$m_{ik}^{iM-LWDF} = \alpha_i D_{HOL,i} \cdot \frac{d_k^i(t)}{R_i(t-1)\beta} \tag{2.8}$$

Where β is the bandwidth of flow.

III DOWNLINK PACKET SCHEDULING IN LTE NETWORKS

The per-RB metrics comparison that serves as the transmission priority of each user on a specific RB is taken into account for resource allocation for each UE. For example the *k*-th RB is allocated to the *i*-th user if its metric $M_{i,k}$ is the largest one among all i-UEs, i.e., if it satisfies the equation:

$$M_{i,k} = \max_i\{M_{i,k}\} \tag{2.9}$$

Figure 2.2 show Generalize model of packet scheduling in downlink of LTE the whole process of downlink scheduling can be divided in a sequence of operations that are repeated, in general, every TTI.

1) The eNodeB prepares the list of flows which can be scheduled in the current TTI. Flows could be formulated only if there are packets to send at MAC layer and UE at receiving end is not in the idle state.
2) Each UE decodes the reference signals, reports CQI (Channel Quality Indicator) to eNodeB which helps to estimate the downlink channel quality. The eNodeB can configure if the CQI report would correspond to the whole downlink bandwidth or a part of it which is called sub-band.
3) Then the chosen metric is computed for each flow according to the scheduling strategy using the CQI information. The sub-channel is assigned to that UE that presents the highest metric.
4) For each scheduled flow, the eNodeB computes the amount of data that will be transmitted at the MAC layer i.e. the size of transport block during the current TTI. The AMC (Adaptive Modulation and Coding module) at MAC layer selects the best MCS (Modulation and Coding Scheme) that should be used for the data transmission by scheduled users. Link adaptation involves tailoring the modulation order (QPSK, 16-QAM, 64-QAM) and coding rate for each UE in the cell, depending on the downlink channel conditions.
5) Physical Downlink Control Channel (PDCCH) is used to send the information about the users, the assigned Resource Blocks, and the selected MCS to terminals in the form of DCI (Downlink Control Information).
6) Each UE reads the PDCCH payload .If a particular UE has been scheduled; it will try to access the proper PDSCH payload. The users are prioritized by packet scheduler on the basis of a scheduling algorithm being used. These algorithms while making scheduling decisions, takes into account the instantaneous or average channel conditions, Head of Line (HOL) packet delays, status of receiving buffer or type of service being used [15].

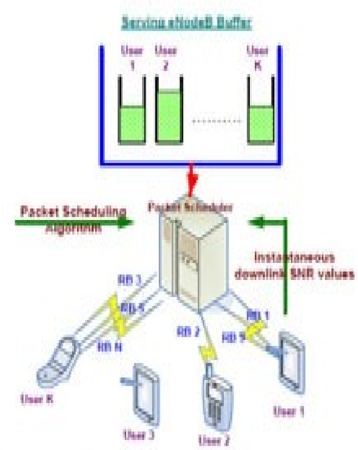

Figure 2.1 Generalize model of packet scheduling in downlink of LTE.

METHDOLOGY

A. THE PROPOSED IMPROVE MODIFIED LARGEST WEIGHT DELAY FIRST ALGORITHM (INPR_MLWDF)

This algorithm take into account the dynamic nature of the users which in turn affect their channel conditions as a result of moving from one point to the other and those users that resides at the cell edge in packet scheduling. Natural exponential function was introduced in the adopted model i.e, MLWDF Algorithm which help the propose algorithm metrics to grow for the users with poor channel conditions and growing head of line packet delay which the adopted model i.e, (MLWDF) fail to consider.

B. THE PROPOSE MODEL

Equation (2.4) i.e., the mathematical model for MLWDF Algorithm was adopted W_i is head of line packet delay for i-th user , $r_i(t)$ is the channel capacity with respect to i-th user, $\overline{r_i}$is average channel capacity before the current Transmission time interval (TTI) and

$$\alpha_i=-\frac{(log\delta_i)}{\tau_i} \quad (2.10)$$

$\tau_i$is the delay threshold

$\delta_i$is maximum probability of exceeding delay threshold.

Taking a critical look at the adopted model shows that it is Channel/QoS Aware scheduling algorithm model, this implies that the model has two part the part that deal with users channel state and the part that deals with users QoS requirement. The improvement was made on the QoS requirement part of the adopted model to eliminate starvation of users particularly those users who don't have better channel condition as a result of mobility from one point to the other or resident at the cell edge, the probability that the packet head of line delay of this users exceeding the delay threshold is bigger which can cause high packet loss ratio, high delay and low throughput.

To eliminate this problem the QoS information send to eNodeB by the users during their Channel Quality Indicator (CQI) reporting were considered Particularly the head of line packet delay and the delay threshold for the i-th user which is very important parameter that must be take into consideration to give priority to users who don't have better channel conditions during packet scheduling at eNodeB.
We considered an important scheduling priority ratio $\frac{W_i}{\tau_i - W_i}$ that grows the adopted model for users who have large head of line delay.
Normally the priority of the user to be schedule is directly proportional to the head of line packet delay of the user and inversely proportional to the delay deadline which is the deference between the delay threshold and the head of line packet delay, the bigger the value of $\frac{W_i}{\tau_i - W_i}$ the higher the metrics must be. However in other to eradicate starvation in the system we applied natural exponential to $\frac{W_i}{\tau_i - W_i}$ which gives

$$\exp\left(\frac{W_i}{\tau_i - W_i}\right)$$

when the differences between the delay threshold τ_i and the head of line packet delay W_i becomes smaller or the delay tolerance of the packet is reaching the end the natural exponential function grow the value of the proposed algorithm for users with poor channel conditions thereby decreases the chance of packet missing the deadline and from being dropped from the buffer.
From the adopted model:

$$\text{Let}W_i(t)=\exp\left(\frac{W_i}{\tau_i - W_i}\right) \quad (2.11)$$

Finally substituting equation (2.11) into equation (2.4) i.e, the adopted model give the proposed INPR_MLWDF algorithm.

$$M_i^{INPR\_MLWDF}=\alpha_i\frac{r_i(t)}{\overline{r_i}}\exp\left(\frac{W_i}{\tau_i - W_i}\right) \quad (2.12)$$

Where W_i is head of line packet delay for i-th user , $r_i(t)$ is the channel capacity with respect to i-th user, $\overline{r_i}$is average channel capacity before the current TTI and

$$\alpha_i = -\frac{(log\delta_i)}{\tau_i}$$

$\tau_i$is the delay threshold

$\delta_i$is maximum probability of exceeding delay threshold.

C. LTE SIMULATOR (LTE-Sim)

LTE-Sim is an open source frame work to simulate LTE network written in c++ language, it encompasses several aspects of LTE network such as Evolve Universal Terrestrial Radio Access Network (E-UTRAN) and Evolve Packet Core (EPC).LTE-Sim supports single and multi cell environments, QoS management, multi-users environment, user mobility, handover procedures and frequency reuse techniques. Three kind of network nodes are modeled in LTE-Sim this are user equipment (UE),enhance node base stations (eNodeB), mobility management entity (MME).The simulator also incorporate support for management of data radio bearer, Adaptive modulation and coding scheme (AMC) and Channel Quality Indicator (CQI) feedback. Finally the simulator implement some of the well-known scheduling algorithms such as Proportional Fair (PF),Exponential Proportional Fair (EXP/PF) and Modified Largest Weight Delay First (M-LWD).[19,20]

D. TRAFIC MODEL

The eNB is located at the center of the macro cell and it communicates using anOmni-directional antenna in a 10 MHz bandwidth. Each UE uses at the same time a video flow a VoIP flow and an IMS signalingflow . For the video flow a trace-based application that emits packets based on realistic video trace files with a rate of 242 kbps. The maximum transmission unit is set to 500 bytes, as in [20]. For VoIP a G.729 voice stream with a rate of 8.4 kbps was considered. The Voice flow is a bursty application that is modeled with an ON/OFF Markov chain [17],the simulation parameters is show in table 1.1

Table 1.1 Simulation Parameters

PARAMETER	VALUES
Simulation duration	100s
Modulation type	QPSK,16QAM and 64QAM
Frame structure	FDD
Mobile speed	3kmph
Radius	1km
Bandwidth	10MHz
Slot duration	0.5ms
Transmission time interval	1ms
Number of resource block	50
Maximum delay	1ms
Video bit rate	242kbps
VoIP bit rate	8.4kbps
Minimum number of users	10
Maximum number of users	50
Interval between users	10

The LTE propagation loss model is composed by 4 different models (shadowing, multipath,
penetration loss and path loss)[16]
– Path loss: PL = 128 : 1 + 37 : 6 log(d) where d is the distance between the UE and the eNodeB in km.

ISSN: 2449 – 0539
BAYERO JOURNAL OF ENGINEERING AND TECHNOLOGY (BJET) VOL.14 NO.2, pp 81-91 AUGUST, 2019

– Multipath: Jakes model
– Penetration Loss: 10 dB
– Shadowing: log-normal distribution (mean = 0dB, standard deviation = 8dB)

E. PERFORMANCE METRICS

The performance of any network depend on the QoS experience by the users in network, the following performance metric were used in this research.

1. Delay

This is a measure of the time difference between the time the packet leaves the sending end to the time the packet was receive at the receiving end measured in second (s).

2. Packet Loss Ratio (PLR)

This measured the percentage of packets of data traveling across a physical channel which could not reach their destination [6],.It can be calculated using the relationship bellow:

$$PLR=\left(\frac{P_{transmit}-P_{recieve}}{P_{transmit}}\right)*100 \quad (2.13)$$

Where,$P_{recieve}$ is the size of the received packet

$P_{transmit}$is the size of the transmitted packet

3. Fairness Index

This is term used to measure fairness among users it is used to determine whether users are receiving a fair share of the network resources, it can be measured using the famous Raj jain Fairness index formulae bellow [18],.

$$F=\frac{\left(\sum_{i=1}^{N} x_i\right)^2}{N\sum_{i=1}^{N} x^2} \quad (2.14)$$

Where, X_i is the amount of data that service i transmits successfully N is the total number of service i. Suppose F = 1, it means that the resources allocation can meet all the users and the system fairness is best. The higher the value of F is, the better the fairness between users, the value of F range from 0 to 1

4. Throughput

This measures the rate of useful bit successfully transmitted through a network,

it can be calculated using the relation bellow,[6].

$$Throughput = \frac{P_{transmit}}{t} \quad (2.15)$$

Where, $P_{transmit}$ is the size of the transmitted packets and t is the time taking to transfer by each user.

PERFORMANCE ANALYSIS

The following performance metrics were used for the result analysis. Delay, Packet Loss Ratio (PLR),Throughput and Fairness index, Video, VoIP and IMS signaling flow were used for the performance evaluations, for the purpose of analysis the propose algorithm was represented by Propose, MLWDF by Mathew_et_al, I-MLWDF by Chrisantus and Mod_MLWDF by Cosmos.

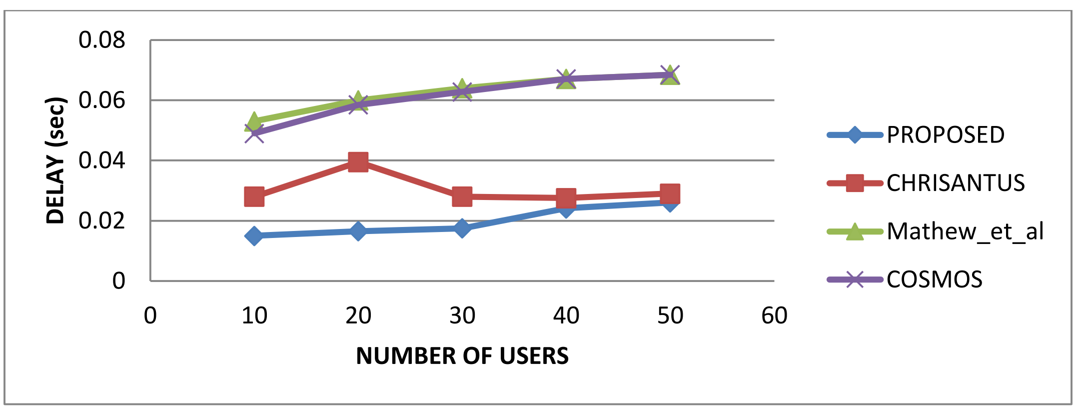

Figure 5.1 Video Delay

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BAYERO JOURNAL OF ENGINEERING AND TECHNOLOGY (BJET) VOL.14 NO.2, pp 81-91 AUGUST, 2019

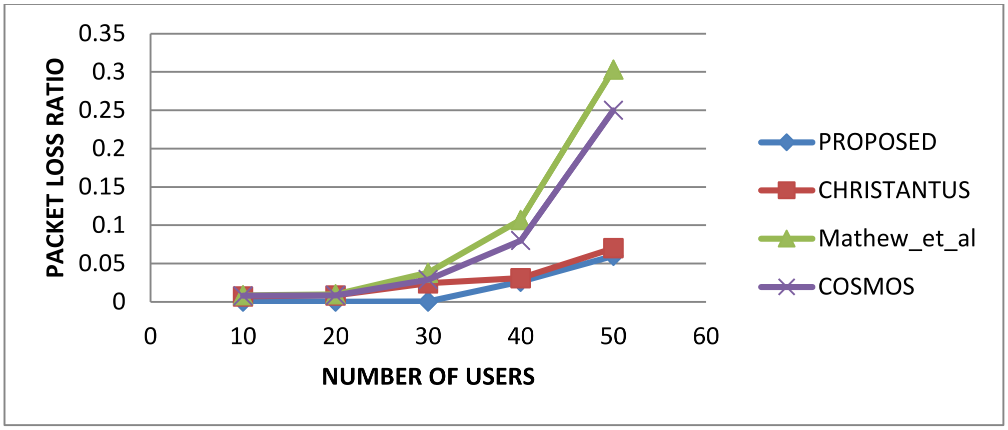

Figure 5.2 Video Packet Loss Ratio

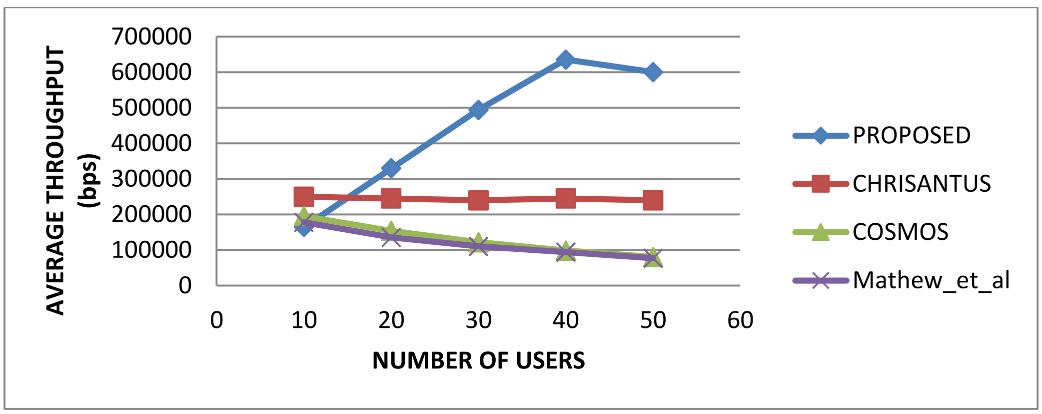

Figure 5.3 Average Video Throughput

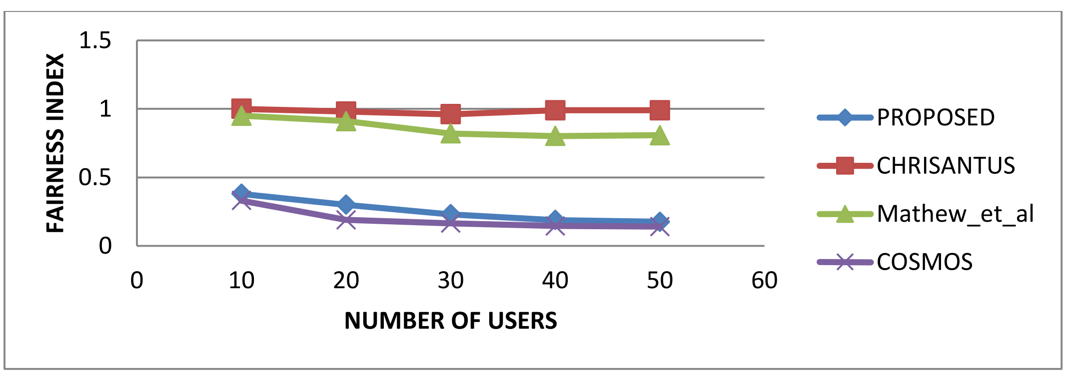

Figure 5.4 Video Fairness Index

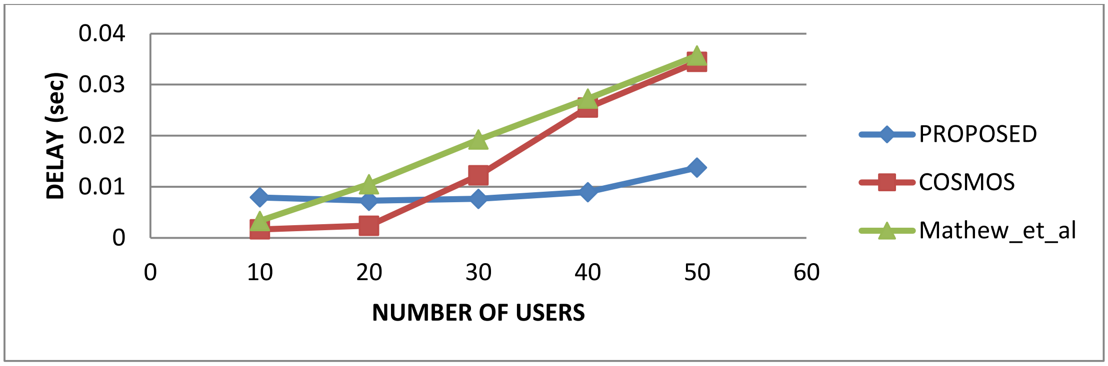

Figure 5.5 VoIP Delay

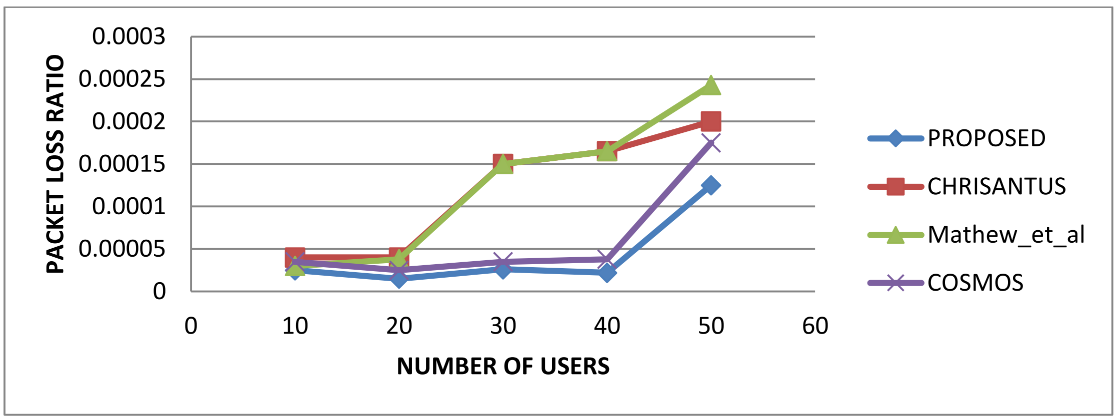

Figure 5.6 VoIP Packet Loss Ratio

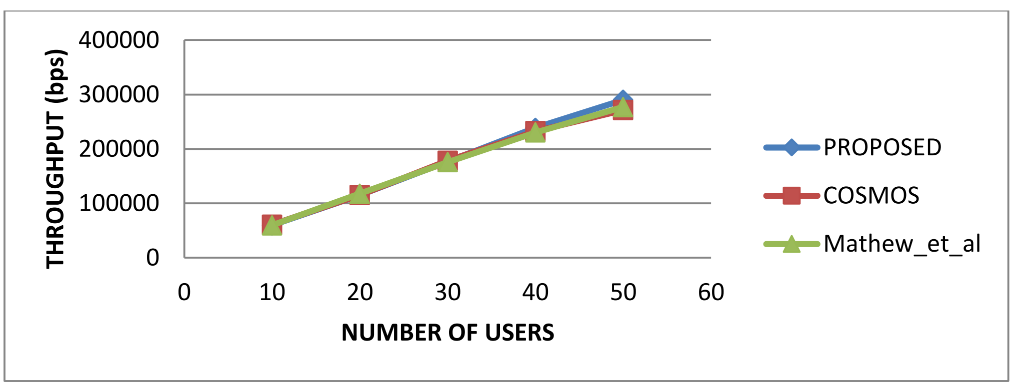

Figure 5.7 VoIP Throughput

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BAYERO JOURNAL OF ENGINEERING AND TECHNOLOGY (BJET) VOL.14 NO.2, pp 81-91 AUGUST, 2019

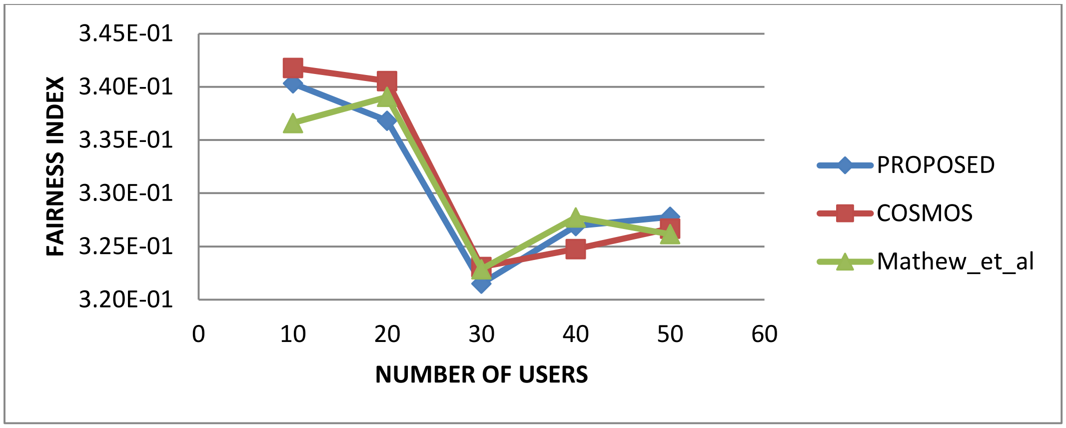

Figure 5.8 VoIP Fairness Index

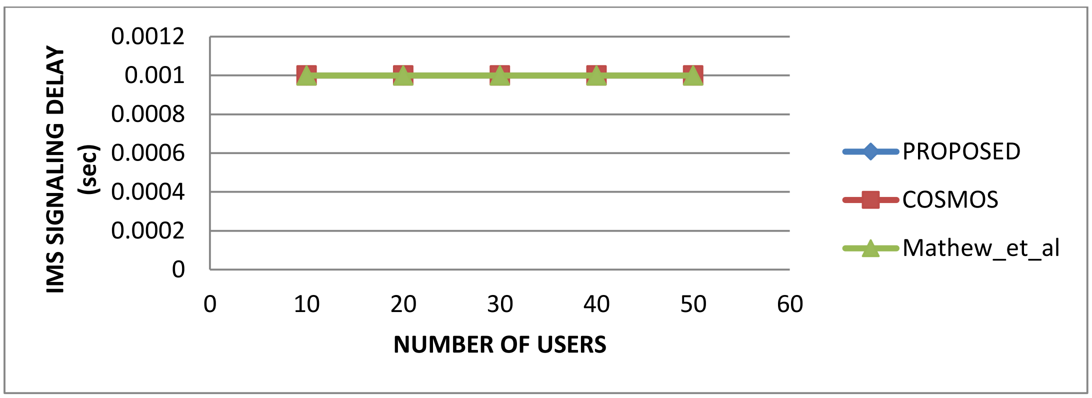

Figure 5.9 IMS Signaling Delay

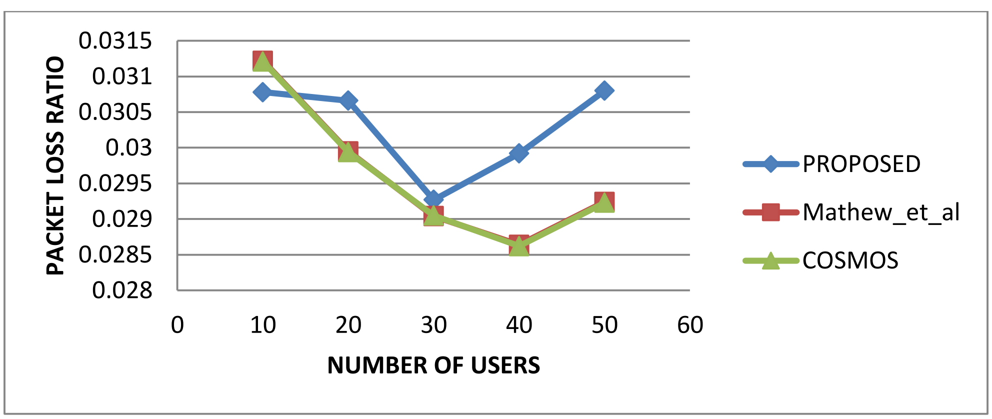

Figure 5.10 IMS Signaling Packet Loss Ratio

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BAYERO JOURNAL OF ENGINEERING AND TECHNOLOGY (BJET) VOL.14 NO.2, pp 81-91 AUGUST, 2019

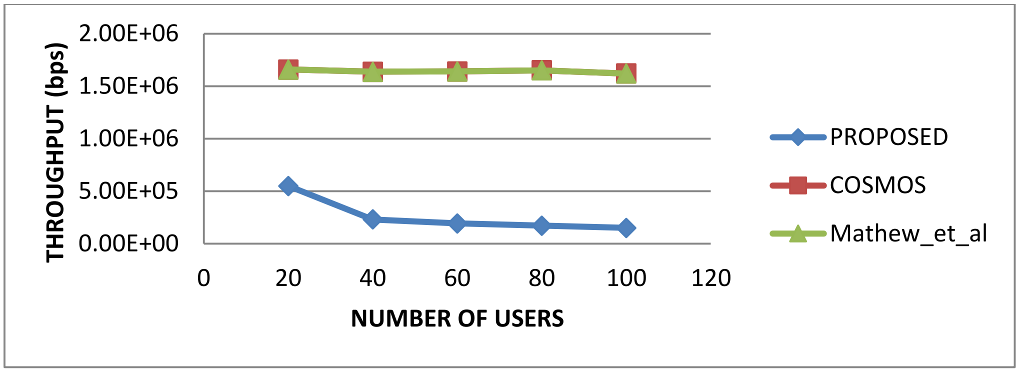

Figure 5.11 IMS Signaling Throughput

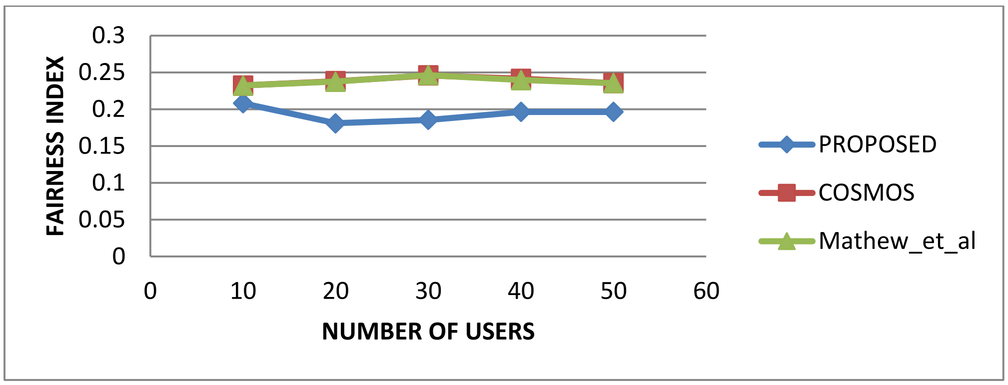

Figure 5.12 IMS Signaling Fairness Index

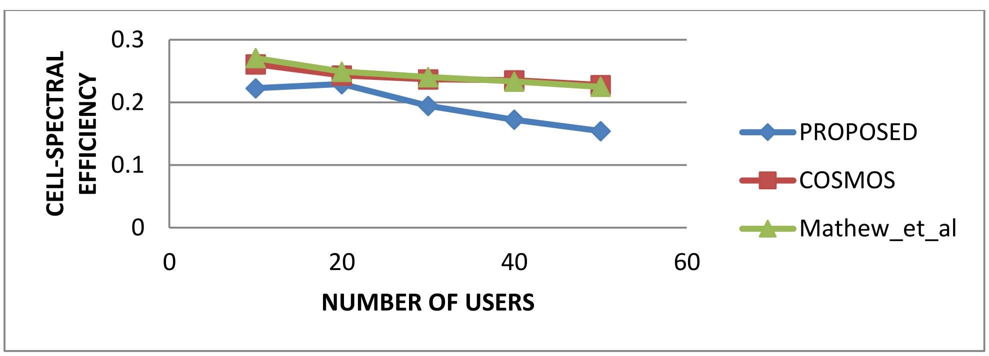

Figure 5.13 Cell-Spectral Efficiency

Figure 5.1 show the delay experience by video flow as the number of user in the network increases the delay increases for all the algorithms. However the propose algorithm maintain very low delay throughout the simulations this is because the propose algorithm put delay deadline and head of line packet delay in consideration when taking scheduling decisions after comparison it was observed that the propose algorithm was better than Chrisantus ,Cosmos and Mathew_et_al at user 10, by 15.10% , 33.12% and 34.80% at user index 30 by 9.82% , 46.03% and 47.01% and at user index 50 by 3.56% , 41.35% and 42.21%.

In figure 5.2 the relationship between the video PLR and the number of users with respect to all the algorithm were shown it was observed that as the number users in the system increases the packet loss ratio increases across all the algorithms, the propose algorithm and Chrisantus were able to hold the PLR bellow the allowable threshold of 3% [16] for up to user 43 with the propose algorithm having the lowest PLR this is because it was able to schedule user before their delay deadline there by reducing their PLR, when comparison were made it was observed that the propose algorithm was better than Chrisantus, Cosmos Mathew_et_al at user index 10 by 0.86%, 0.59% and 0.62% at user index 30 by 4.94%, 5.11% and 6.30% and at user index 50 by 1.63%, 19.01% and 24.50% respectively. The Video throughput is shown in figure 5.3 it is observed that while other algorithm throughput were deteriorating from the beginning to the end of the simulation as the number of users in the system was increasing the propose algorithm as increasing in throughput up to user 40 before it begin to deteriorating this due to the fact the propose algorithm was able to schedule much users before their deadline expirations, comparison between the algorithms shows that Chrisantus, Cosmos and Mathew_et_al are better than the propose algorithm at user index 10 by 25.72%, 3.48% and 2.80% while the propose algorithm was better than Chrisantus , Cosmos and Mathew_et_al at user index 30 by 38.33% , 59.35% and 60.45% and at user index 50 by 47.93%, 77.16% and 77.36% respectively.

Figure 5.4 depict how fair the algorithms are to the users in terms of Video flow it was observed that Chrisantus maintain high fairness index throughout the simulation, this is due to the incorporation of flow bandwidth to the average rate parameter in the algorithm, comparison show that Chrisantus was better than Cosmos, propose and Mathew_et_al algorithm at user index 10 by 68.90% , 66.82% and 9.30% at user index 30 by 79.21% , 74.34% and 16.20% and at user index 50 by 82.54% , 80.12% and 19.41% .In figure 5.5 graphical representation of VoIP delay was displayed all the algorithm experience increase in delay as the number of users increases but the propose algorithm maintain lowest delay up to user 40 before it begins to rise in comparing the performance of the algorithms it show that Cosmos and Mathew_et_al better than the propose algorithm at user index 10 by 19.35% and 18.20% while the propose algorithm was better than Cosmos and Mathew_et_al at user index 30 by 13.50% and 23.30% and at user 50 by 38.13% and 38.95%. Figure 5.6 show the relationship between PLR and the number of users regarding VoIP flow all the algorithm experience increase in packet loss as the number of users increases but they are all within the acceptable packet loss rate for VoIP i.e 1.2% [16], when comparison were carried out it was observed that the propose algorithm was better than Chrisantus, Cosmos and Mathew_et_alat user index 10 by 0.85%, 0.89% 0.91% at user index 30 by 11.50% , 0.98% and 11.76% and at user index 50 by 8.09%, 5.20% and 13.86% .In figure 5.7 depict the graphical representation of VoIP throughput it was observed that the throughput of all the algorithm are close this is because the VoIP flow was model using on ON/OFF markov chain model this implies that there was no transit packet arriving at the buffer during the OFF state, however where comparison were made Cosmos and Mathew_et_al were better than the propose algorithm at user index 10 by 0.69% and 0.65% and at user 50 by 0.11% and 0.15% while the propose algorithm was better Cosmos Mathew_et_al and at users index 50 by 1.69% and 1.49%. The fairness index of the VoIP flow is shown in figure 5.8 it was observed that the fairness index decrease as the number of users increases after comparison it shows that Cosmos was better than propose and Mathew_et_al algorithm at user index 10 by 0.42% and 3.6% and at user 30 by 0.48% and 0.15% while the propose algorithm was better than Cosmos and Mathew_et_al at user index 50 by 0.34% and 0.37%.

For the IMS signaling Cosmos performance is the same as Mathew_et_al because at priority level 1 in standard QCI table equation 2.5 will become exactly like equation 2.3.In figure 5.9 the delay of IMS Signaling is displayed all the algorithm were kept at buffering delay threshold i.e., 1ms.Figure 5.10 shows the packet loss ratio regarding IMS signaling the PLR continue to increase as the number of users increases when comparison were made the propose algorithm was better than Cosmos and Mathew_et_al at user index 10 by 16.50% at user 30 and 50 Cosmos and Mathew_et_al are better than the Propose algorithm by 8.76% and 36.63% .The throughput of the algorithms regarding IMS signaling is shown in figure 5.11 the propose algorithm maintain very low throughput throughout the simulations this is because the parameters of the propose algorithm were not design to handle packet which exceed the delay threshold while Cosmos and Mathew_et_al have the capacity to schedule packet that have exceed their delay threshold after comparison it shows that Cosmos and Mathew_et_al were better than propose algorithm at user index 10,30 and 50 by 66.91% , 88.19% and 90.71% respectively.

Figure 5.12 depict the fairness index of IMS signaling with the propose algorithm having very low fairness after comparison it show that Cosmos and Mathew_et_al were better than the Propose algorithm at user index 10, 30 and 50 by 10.34% , 24.62% and 16.60%.The cell spectral efficiency is shown in figure 5.13 it is observed that Cosmos and Mathew_et_al are better than the propose algorithm.

VI. CONCLUSION

An improvement on an existing scheduling algorithm i.e., MLWDF was carried out to get the propose algorithm with the main purpose of improving the QoS performance of real-time flows and a satisfactory level of performance for the non-real time flows in the network by incorporation of scheduling priority ratio. Future work could be to improve on the propose algorithm to increase its performance in terms of non-real time flow, Cell-spectral efficiency and to consider implementing the algorithm for uplink transmissions.

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