# Adaptive Weighted Lottery Earliest Deadline First Scheduler (AWLEDF) with Feedback Control-congestion Mechanism

Ma'en Saleh

Electrical & Computer Engineering, Tafila Technical University
P. O. Box 179, Tafila 66110, Jordan

*Abstract*— **Different network technologies were developed to provide guaranteed quality of service (QoS) requirements for different classes of data flows that passing through a network. Accordingly, a variety of scheduling algorithms were proposed. In this research, we propose an efficient scheduling algorithm (AWLEDF) that integrates a multi-layer scheduler with a network congestion control methodology to be installed on the edge router of a packet switched network. The Multi-layer scheduler combines the earliest-deadline-first (EDF) and the lottery schedulers to provide guaranteed QoS for three classes of data flows: video, audio, and best effort (text) traffics, while the congestion control methodology implements a feedback mechanism that monitors the buffer resources at the edge router. Compared to FCFS, EDF, and Lottery-FCFS schedulers, AWLEDF shows high efficiency in providing QoS requirements in terms of miss-rate and average packet delay, while protecting the network from being congested efficiently through measuring the buffer utilization at the edge router.**

*Keywords*— **Scheduling, Control-Congestion, NPMs, QoS, Lottery.**

## I. Introduction and Related Work

The type of service provided by the network depends on the technology adopted by the network itself, and the type of data flow and its requirements when admitted to the network. The earliest type of data traffic was not sensitive to any quality metrics, and then the only guarantee that was provided by the network was the serving guarantee even if the service was delayed. Such service was called the best effort service for non-real-time data traffics [1]. Nowadays, different categories of data flows share the same network technology. Each flow requests a service with different quality of service (QoS) requirements [2].

Network technologies adopt different scheduling algorithms to provide the best service to its clients (flows) within the requested QoS requirements [scheduling]. In [3], we perform a performance study for two scheduling algorithms first-come-first-served (FCFS), and earliest-deadline-first (EDF) when applied for real-time (RT) and non-real-time traffics (NRT) in a packet switched network. The study shows the efficiency of EDF over FCFS in terms of miss-rate, delay, and buffer utilization QoS metrics. Dynamic Multilevel Priority (DMP) scheduler was proposed for packet wireless sensor networks (WSN) to serve both RT and NRT traffics [4]. Compared to FCFS and static multi-level schedulers, DMP shows efficiency in minimizing packet's delay and end-to-end delay. In [5], three EDF schedulers were employed for hard-real-time (HRT) clustered network: global, partitioned, and clustered earliest-deadline-first (G-EDF, P-EDF, and C-EDF). Empirical results show that C-EDF is the most preferable scheduler for both HRT and soft-real-time (SFT), where strength QoS requirements are guaranteed by such scheduler. Global fair lateness (G-FL) algorithm was proposed in [6] to schedule soft real-time tasks in a clustered network. Compared to G-EDF, G-FL maximizes the tardiness bounds for the real-time tasks and thus, increaser the probability of meeting their QoS requirements. In [7-8], we used the differentiated earliest-deadline-first (Diff-EDF) scheduler as the scheduling unit in a security-aware scheduler for packet switched networks. Compared to ECFS and EDF schedulers, the Diff-EDF scheduler shows high efficiency in minimizing the miss-rate and total average delays for different real-time data flows.

Different scheduling algorithms were proposed based on integrating two or more scheduling algorithms in a single unit with a layer of communication between the sub-schedulers. Such method depends on queuing the arrived data units in different queues such that, each queue is operated by a specific scheduling algorithm. Accordingly, there should be a layer of control to switch from one queue to another. Such algorithms include: smoothed round robin (SRR) [9], hybrid genetic packet Scheduler (HGPS) [10], tri-state multilayer scheduler TSM that combines FCFS, FQ, and EDF scheduling algorithms in a multi-queue scheduler system (text, video, and audio)[11], and multi-level (ML) scheduler based on EDF and worst fair queue (WFQ) to serve hard and soft real-time traffics in a heterogeneous network [12].

Different network performance metrics (NPMs) can be used to reflect the over whole performance of the network such as throughput, bandwidth, packet loss, and delay [13]. Such metrics are badly affected by the congestion of the network. In order to preserve such metrics, while providing the best service to the admitted flows, different congestion control schedulers were proposed such as: sliding mode congestion control scheduler (SM) [14], joint transmission scheduling and congestion control for wireless networks [15], distributed link scheduling for wireless network [16], and joint congestion control, routing, and scheduling for heterogeneous network [17].

In this research, we propose a congestion control aware scheduler (AWLEDF) that uses the hybrid scheduler (LEDF) to provide the best QoS requirements for different classes of data flows, while protecting the network from being congested through deploying a buffer estimation feedback mechanism at the edge router of a packet switched network.

The performance of the system was shown through a simulation comparison with three other scheduling algorithms: FCFS, EDF, and lottery-FCFS.

## II. Network Topology

In our research, we model a packet switched network with *M* source nodes communicating with M destination nodes. The destination nodes are connected to a local area network (LAN) and communicate through an edge router with the external world. Such router is the core unit of our research, where our proposed system was installed at that layer of the network; hence the main goal is to schedule packets efficiently to their destinations in the LAN environment.

Since we are dealing with packet switched network, the technology that we used in our research is the IEEE 802.1Q Ethernet technology that defines the frame format of the data unit exchanged through the network (Ethernet Packet). Accordingly, we assume that each source node is generating either real-time traffic (video, audio) or non real-time traffic (best effort) with the following specifications: (1) A maximum packet size of 1500 bytes based on the maximum IEEE 802.1Q Ethernet packet's payload; (2) Packet deadline ($D$) that is the remaining time of the packet once it arrives the edge router before it can be considered as expired packet; (3) The packet's inter-arrival time to the edge router that was modelled exponentially based on the traffic sending rate by the source node ($\lambda$). The performance metrics for our research could be represented from different views such as, traffic's miss rate (miss rate) and average total queuing time for packets at edge router (total delay). Such metrics play a key role in evaluating the over whole performance of the real-time network, since it directly affects the congestion of the network.

## III. Lottery-EDF Scheduler (LEDF)

Our real-time scheduler was modelled based on two scheduling phases: external phase and internal phase. The external phase was designed based on the lottery scheduling algorithm, while the internal phase was earliest deadline first (EDF) based scheduler. The scheduler was installed at the default gateway router (edge router) to perform the process of serving the arrived data packets within their QoS requirements.

### A. *External Phase (Lottery Scheduler)*

Lottery scheduling algorithm provides an efficient resource management methodology, where system resources are shared among a number of processes (clients). According to lottery scheduling algorithm, each process (client) receives a share to a given resource based on a number of tickets it holds for that resource [18-19]. Accordingly, a process that has more tickets for a resource will have the opportunity to use the resource more than other processes. The differentiation in the number of tickets given for the processes reflect the differentiation in the weight for each individual process that is, the process with the highest priority will be given the highest number of tickets.

In our research, the processes are of three main data classes: video, audio, and best-effort (text). Such processes are sharing the edge's router installed memory, where arrived data packets are queued in the memory before being served by the scheduler. The service provided by the scheduler is by choosing the packet to be served and forwarded through the LAN to its destination or dropping the packet if it missed its deadline.

Our scheduler divided the edge router's buffer ($Q$) into three buffers (queues): video queue ($Q_v$), audio queue ($Q_a$), and text queue ($Q_t$). Each queue holds the corresponding arrived data type where video packets queued at the video queue, audio packets to the audio queue, and text packets to the text queue. The sizes of the queues are defined as: video buffer size ($B_v$), audio buffer size ($B_a$), and text buffer size ($B_t$) such as:

$$B = B_v + B_a + B_t \tag{1}$$

Each data type will be assigned a number of tickets reflects the priority scheme followed by the lottery scheduler in serving the real-time flows. In our model we assigned the real-time traffic more priority than the best effort traffic, and among real-time traffic, we assigned the video packets the highest priority. By modeling the priority scheme using a random number generator that generates instances between 0 and 1, we define the priorities for each flow as the following: video traffic priority ($P_v$), audio traffic priority ($P_a$), text traffic priority ($P_t$) such as:

$$P_v + P_a + P_t = 1 \tag{2}$$

The lottery scheduler depends on the previous priorities to choose the queue to fetch from for serving. Assuming that the random generator generates an instance $x$, such that: $x \in [0,1]$ and $P_v > P_a > P_t$ then:

$$\boxed{\begin{array}{c} \text{If } x \in [0, P_t] \rightarrow Serve\_Text \\ \text{If } x \in [P_t, P_t + P_a] \rightarrow Serve\_Audio \\ \text{If } x \in [P_t + P_a, P_t + P_a + P_v] \rightarrow Serve\_Video \end{array}}$$

### B. *Internal Phase (EDF scheduler):*

This is one of the priority-based scheduling algorithms, where the remaining time of the packet is the priority key. In such scheduler, the packet that is closer to expire will be given a higher priority than all other packets in waiting queue of the scheduler. According to the lottery scheduler, the arrived data packets will be queued based on their types without specifying the method of ordering the packets. In our scheme, we adopt the earliest deadline first algorithm to be employed internally at each queue that is, the packets will be ordered in each queue based on their remaining time for expiration. No peremption algorithm is adopted here, such that the new arrived packet that is closer to expire than the already under serving packet will not pre-empt it from the top of the queue. The EDF algorithm will keep the packets from missing their deadline, and then achieving low flow's miss rate. As a result, the QoS requirements will have a higher chance to be guaranteed by the scheduler.

## IV. AWLEDF Scheduler

The LEDF scheduler provides a mechanism to provide the data packet generators with the best QoS requirements. Actually, the previous scheduler doesn't give any attention to the status of the network such as network congestion due to a heavy traffic load. The congestion may be caused due to filling one of the edge router queues, and then the new arrived packet to that queue will be dropped after being pending for a little amount of time. According to TCP/IP protocol, when the source forwards a packet, it waits for an acknowledgement of recipient from the destination. If the packet was dropped by the scheduler, the acknowledgment will not be sent. Accordingly, the source will regenerate the packet, which may cause a repeated traffic through the network. Such overhead may congest the network.

In our research, we add a congestion control methodology to our LEDF scheduler that monitors the buffers status of the edge router (video, audio, and text buffers) through a feed back from the buffers themselves. Such congestion control can be implemented by two methods: (1) Request from the scheduler to the edge router asking from the status of the buffer (the available buffering system) periodically every time instance (T); (2) Asynchronous control message from the buffer system when a critical situation happens to its status. In our research, we implement the second technique to avoid the overhead generated by the periodic control messages from the queue system. Accordingly, an adaptive weighted lottery EDF scheduler was proposed (WLEDF) by integrating the LEDF to congestion control methodology to be installed at the gateway (edge) router. In order to implement our methodology, the scheduler was divided into two separated sub-units: buffer and controller sub units with both data and control communication buses between them to exchange data and control messages. The internal design of the scheduler is shown in Fig.1.

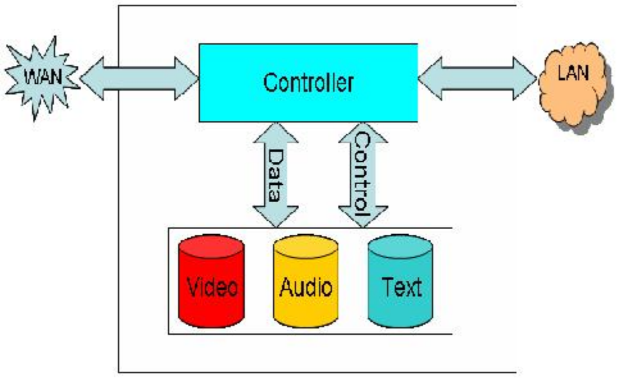


Fig. 1 Internal Structure of AWLEDF Scheduler

## V. System Methodology

The process started when a new packet arrived to the scheduler, the controller forwards the packet to the buffer through the data bus. The buffer unit queues the packet based on its type to the corresponding sub-buffer (video, audio, or text) and rearranges the sub-buffer according to the EDF algorithm. The controller runs the random generator mechanism that selects one of the sub-buffers to serve from according to the process explained in section III-B. Once the sub-buffer was selected, the controller sends a control-message through the control bus to the buffer sub-unit requesting a packet from the reselected sub-buffer. The buffer responds to the request and forwards the packet from the top queue (closer to expire) of the specified sub-buffer through the bi-directional data bus to the controller. The controller checks if the packet was expired or not. If the packet was expired, the controller drops the packet; otherwise, the controller serves the packet and forwards it to the destination according to its media access control (MAC) address. Fig.2 shows the network topology studied in this research.

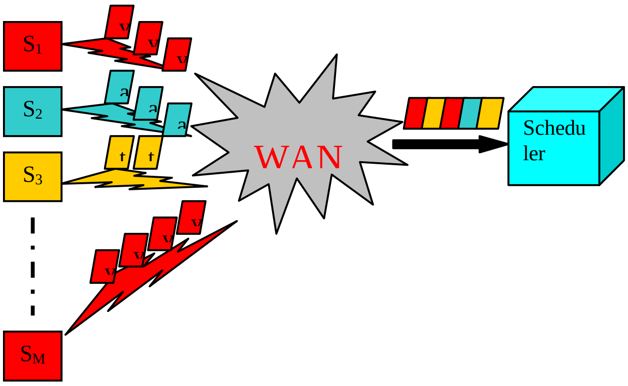


Fig. 2 Network Topology

In order to operate the congestion control mechanism, we assume a threshold for each sub-buffer that can't be exceeded by the queued packets as the following: (1) $\cdot_v$ : The threshold for the video sub-buffer; (2) $\cdot_a$: The threshold for the audio sub-buffer; (3) $\cdot_t$: The threshold for the text sub-buffer. The buffer unit keeps track of the queued packets at each sub-buffer by defining three counters: $\Phi_v$: The counter for video sub-buffer; (2) $\Phi_a$: The counter for the audio sub-buffer; (3) $\Phi_t$: The counter for the text sub-buffer. When the buffer queues a packet, it increments the corresponding counter, and when it retrieves a packet from the top of the sub-buffer, it decrements the corresponding counter. The congestion mechanism is operated if the following condition is not met:

$$\cdot_x * B_x > \Phi_x \tag{3}$$

Where $x \in \{v, a, t\}$.

In such case, the buffer sends a control message through the control bus to the controller to operate the congestion control mechanism. Upon receiving the control message, the controller specifies the sub-buffer that generates such message $Q_z$, where $z \in \{v, a, t\}$ and defines the priority vector $P^{\wedge}$ for the uncritical buffers. The controller then modifies the priorities using the priority upgrading parameter ( $^{-}$ ) defined by the controller unit at the initiating process as the following:

$$P_z = P_z +^{-} \tag{4}$$

$$P_x = P_x -^{-} \tag{5}$$

Where $^{-}$ is the priority upgrading parameter and $P_x$ is the lowest priority in $P^{\wedge}$ that is $\geq$ 0.2. If ($P_x < 0.2$), then:

$$P_y = P_y -^{-} \tag{6}$$

Where $P_y > P_x$, and $P_y \in P^{\wedge}$

In our research, we assume that the buffer should wait for $T$ time units (defined by the controller at the initiating process) before sending a new critical control message from any critical sub-buffer. If the controller doesn't receive a new control message after $T$ time units from the buffer, then it retains the previous priorities just before the critical situation occurred. If the controller receives three successive controller messages from the same critical sub-buffer ($Q_z$), it then upgrades its corresponding threshold ( $\cdot_z$ ) using the threshold upgrading parameter (  ) defined by the controller unit at the initiating process as the following:

$$\cdot_z = \cdot_z + \qquad (7)$$

Where   is the threshold upgrading parameter, such that: $(\cdot_z + \ ) < 0.9$.

In order to evaluate the performance metrics for our system (miss rate and average waiting time of packets at scheduler), the controller defines two counters for each node in the local area network (LAN). Such counters store information about the packets served for each node. The two counters are defined as: (1) Served counter $C^x_s$ that will be incremented every time a packet is served for node $x$; (2) Total waiting time counter ($C^x_w$): a counter that holds the waiting time for all packets served for node $x$, such that:

$$C_w^x = \sum_{z=1}^{C_s^x} t_s^z - t_a^z \qquad (8)$$

Where $t_s^z$ is the time the packet $z$ was forwarded after serving from the scheduler to node $x$, and $t_a^z$ is the arrival time of packet $z$ to the scheduler. Accordingly, the miss rate of the flow generated for node $x$ ($\Psi_x$) is defined as:

$$\Psi_x = \frac{C_s^x}{a_x} \qquad (9)$$

The average waiting time in ms of the packets served for node $x$ ($\P_x$) is given by:

$$\P_x = \frac{C_w^x}{C_s^x} \qquad (10)$$

## VI. Simulation & Numerical Results

In this paper, we analyze a simulated heterogeneous packet switched network, where $M$ transmitter nodes (sources) are communicated with $M$ receiver nodes (destinations). The simulation was carried out for different values of $M$ sources/destinations with a step of 3, such that $M$ = 3, 6, 9, 12, --------, 27, 30. In each simulation step, we assume that the number of video, audio, and text streams is equal. Accordingly, we will have M/3 video flows, M/3 audio flows, and M/3 text flows for each simulated step. The deadline distribution function for each flow was simulated as a uniform distribution over an interval [a, b]. The interval bounds for each flow were defined as the following: [20 ms, 140 ms] for video, [150 ms, 270 ms] for audio, and [280 ms, 380 ms] for text. In each simulation step, we assume that the sending rate ($\lambda$) for the $M$ sources is equal to 1000 packets/s.

According to the congestion strategy, we assume that the edge router has buffer resources ($B$) that depend on both $M$ and $\lambda$, such that:

$$B = \frac{M * a}{8} \qquad (11)$$

Accordingly, each traffic queue has an equal size, such that:

$$Q_v = Q_a = Q_t = \frac{B}{3} = \frac{M * a}{27} \qquad (12)$$

The initial priorities for data traffics were defined as the following: video traffic priority ($P_v$) = 0.5, audio traffic priority ($P_a$) = 0.3, text traffic priority ($P_t$) = 0.2. Threshold values for the sub-queues were defined as: $\cdot_v$ = 0.6, $\cdot_a$ = 0.65, $\cdot_t$ = 0.7. The priority upgrading parameter ( ¯ ) = 0.1. The threshold upgrading parameter (  ) = 0.5. The waiting time ($T$) before sending successive critical control messages was set to 15 ms.

In order to measure the performance of our proposed algorithm, we implement three other schedulers to compare with: (1) FCFS, where the first arrived packet will be served first regardless the type and the deadline requirements; (2) EDF, where the packet closer to expire will be served first; (3) WLFCFS, this scheduler has the same implementation of our proposed one with a difference in the internal scheduling policy, such that EDF was replaced with FCFS.

Simulation results show the performance of our proposed algorithm at the edge layer of the network (gate-way router) in terms of miss-rate, average total packet delays, edge router buffer utilization, and congestion control triggers. Fig.3 shows that AWLEDF scheduler has the lowest miss rate among all other schedulers. It's obvious that the miss rate levels are increasing as the number of $M$ nodes is increasing at the LAN connected to the gateway router, where more packets are arriving to the scheduler with higher opportunities to miss their deadlines. Since more packets are arriving as $M$ increase, the waiting time for the packets in the scheduler will increase as shown in Fig.4. Simulation results prove that AWLEDF scheduler decreases the delay time for the queued packets in the scheduler better than FCFS, EDF, and AWLFCFS schedulers as shown in Fig.2. Such previous results show the possibility of protecting the network from being congested, and thus preserve the NPMs.

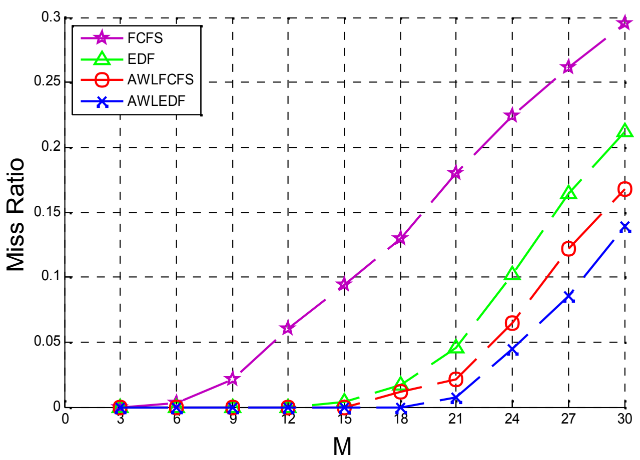


Fig. 3 Miss Ratio Metric

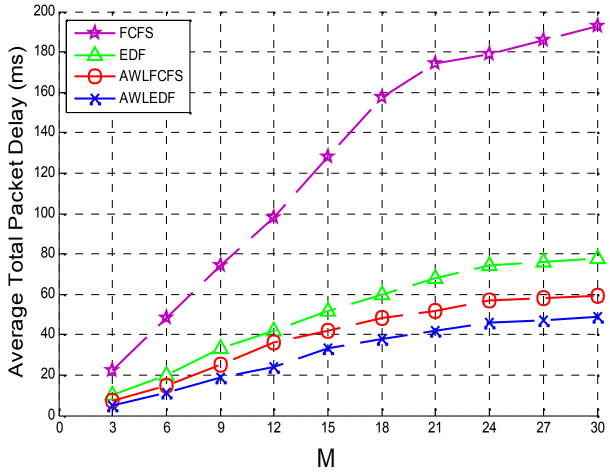


Fig. 4 Average Total Packet Delay

In Fig.5, we estimate the buffer utilization at the edge router using the four schedulers (FCFS, EDF, AWLFCFS, and AWLEDF). Simulation results show that AWLEDF provides the most efficient utilization for the buffer, which decreases the probability of congesting the network. AWLFCFS was the second best choice, where a control-congestion methodology is applied in such scheduler. From the other side, FCFS was the worst one, and thus it wasn't suitable for real-time networks with QoS requirements.

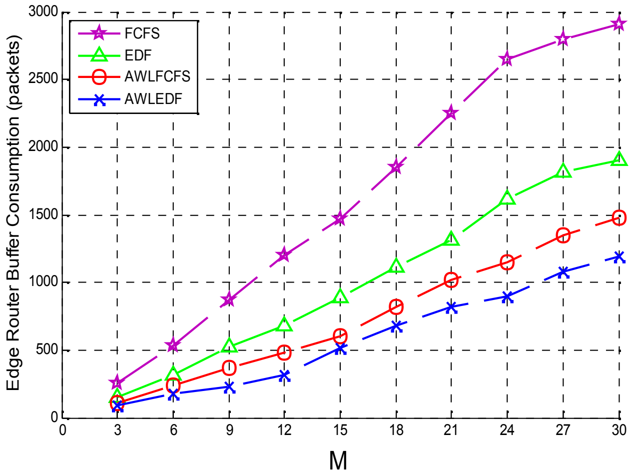


Fig. 5 Buffer Consumption at Edge Router

Although that the congestion control mechanism provides a method for protecting the NPMs, it adds an overhead to the system that may affects the QoS requirements. We perform a simulation that measures the number of triggers caused by the congestion control mechanism for both AWLEDF and AWLFCFS for different values of the predefined waiting time before two successive critical control messages (*T*). Fig.6 shows that AWLEDF scheduler minimizes the number of triggers for the control-congestion unit over the AWLFCFS scheduler, and thus minimizes the overhead of operating the congestion control mechanism. It also shows that the number of triggers will increase if we perform an extensive monitoring for the network by using a smaller value of *T*.

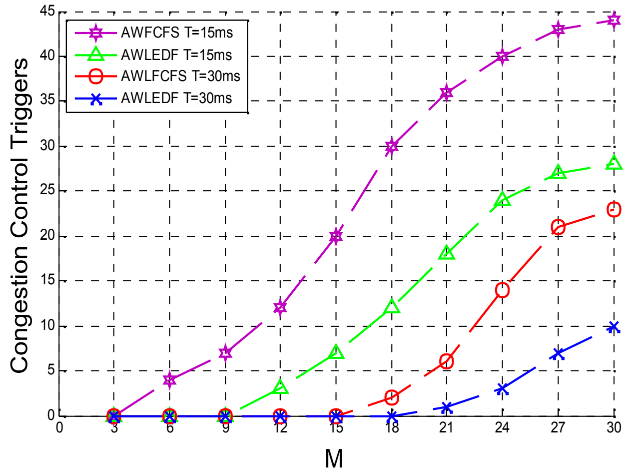


Fig. 6 Number of Congestion Control Triggers

## VII. CONCLUSIONS

In this research, a hybrid multi-layer scheduling algorithm (LEDF) was integrated with a congestion control mechanism to provide a congestion-aware scheduler (AWLEDF) that provides both RT and NRT traffics with a guaranteed QoS requirements, while preserves the NPMs in terms of miss-rate and average delay. As a result, the proposed system protects the packet switched network from being congested by heavy traffic load efficiently. Simulation results proves that AWLEDF is much efficient than FCFS, EDF, and Lottery-FCFS schedulers when applied for a heterogeneous real-time packet switched networks with heavy traffic load. The future work includes adding a layer of security-awareness for such system to protect the real-time traffics from security threats in the LAN environment.

## REFERENCES

[1] S.R. Boddu, A. Mukhopadhyay, P. Chandhar, B.V. Philip, and S.S. Das, "Analysis of fractional frequency reuse in OFDMA networks for real time and best effort traffic," *IEEE International Conference on Communications (ICC)*, pp. 5366 – 5370, June 2013

[2] K.T Phan and L. Tho," Online QoS-based dynamic scheduling in multi-channel wireless networks," *IEEE Conference on Wireless Communications and Networking (WCNC)*, pp. 586 – 590, April 2013.

[3] M. Saleh and L. Dong, "Comparing FCFS & EDF scheduling algorithms for real-time packet switching networks," IEEE International Conference on Networking, Sensing and Control (ICNSC), pp. 698 – 703, April 2010.

[4] N. Nasser, L. Karim, and T. Taleb, "Dynamic multilevel priority packet scheduling scheme for wireless sensor network," *IEEE Trans. on Wireless Communications*, vol. 12, no. 4, pp. 1448 – 1459, April 2013.

[5] A. Bastoni, B.B. Brandenburg, and J.H. Anderson, "An empirical comparison of global, partitioned, and clustered multiprocessor EDF schedulers," *IEEE Conference on Real-Time Systems Symposium (RTSS)*, pp. 14 – 24, Dec. 2010.

[6] J.P. Erickson and J.H. Anderson, "Outstanding paper award: fair lateness scheduling: reducing maximum lateness in G-EDF-Like scheduling," *Euromicro Conference on Real-Time Systems (ECRTS)*, pp. 3 – 12, July 2012.

[7] M. Saleh and L. Dong, " Real-time scheduling with security enhancement for packet switched networks," *IEEE Trans. on Network and Service Management*, vol. 10, no. 3, pp. 271 – 285, Sep. 2013.

[8] M. Saleh and L. Dong, "Adaptive security-aware scheduling using multiagent system," in *2012 IEEE International Conference on Communications.*

[9] K. B. Johnsson, D. C. Cox, ***"QoS*** scheduling of mixed priority non real-time taffic," VTC 2001,

[10] S. Abedi and S. Vadgama, "Hybrid genetic packet scheduling and radio resource management for high speed downlink packet access," *The 5th International Symposium on Wireless Personal Multimedia Communications*, vol. 3, pp. 1192 – 1196, Oct. 2002.

[11] A. M. Z. Bidoki, S. V. Azhari, and N. Yazdani, "A decomposed hierarchical logarithmic scheduling algorithm for input-queued switches," *10th International Conference on Telecommunications (ICT)*, vol. 2, pp. 1662-1669, 2003.

[12] L. Wang, G. Min, D. Kouvatsos, and X. Jin, "An analytical model for the hybrid PQ-WFQ scheduling scheme for WiMAX networks," 1st International Conference on Wireless Communication, Vehicular Technology, Information Theory and Aerospace & Electronic Systems Technology (Wireless VITAE), pp. 492-498, 2009.

[13] P. Kim, "A measurement mechanism of IP performance metrics for network mobility," *2010 International Conference on Control Automation and Systems (ICCAS)*, pp. 1613-1616, 2010.

[14] Z. Ding and D. Wu, "Sliding mode based joint congestion control and scheduling in multi-hop Ad Hoc networks with multi-class services", *2010 IEEE Global Telecommunications Conference (GLOBECOM)*, pp. 1-5, Dec. 2010.

[15] D. Bethanabhotla, G. Caire, and M.J. Neely, " Joint transmission scheduling and congestion control for adaptive streaming in wireless device-to-device networks," *2012 Conference Record of the Forty Sixth Asilomar Conference on Signals, Systems and Computers (ASILOMAR)*, pp. 1179-1183, Nov. 2012.

[16] S. Dong, Q. Yang, F. Fu, and K.S. Kwak, "Distributed link scheduling for congestion control in multihop wireless network, *2013 International Conference on Wireless Communications & Signal Processing (WCSP)*, pp. 1-5, Oct. 2013.

[17] P.L. Vo, N.H. Tran, C.S. Hong, and K. Chae, "A joint congestion control, routing, and scheduling algorithm in multihop wireless networks with heterogeneous Flows," *2011 International Conference on Information Networking (ICOIN)*, pp. 347-351, Jan. 2011.

[18] D. G. Sullivan, R. Haas, and M.I. Seltzer, " Tickets and Currencies Revisited: Extensions to Multi-Resource Lottery Scheduling," *Proceedings of the Seventh Workshop on Hot Topics in Operating Systems*, pp. 148-152, Mar. 1999.

[19] C. Xien and J. Zheng, "Novel method in automatic examination for distance education," 2010 Second International Workshop on Education Technology and Computer Science (ETCS), vol. 1, pp. 582-585, March 2010

**Maen Saleh** (M'10) received the B.S. and M.S. degrees in computer engineering from Yarmouk University in 2005 and 2007, respectively. He received the Ph.D. degree in electrical and computer engineering from Western Michigan University in 2012.

From 2006 to 2007, he was a Lecturer at the Computer Science Department at Yarmouk University. From 2007 to 2008, he was a Lecturer at the Electrical and Computer Engineering Department at Tafila Technical University. From 2010 to 2012, he was a Teaching Assistant at the Electrical and Computer Engineering Department at Western Michigan University. In 2012, he joined the faculty of Tafila Technical University as an Assistant Professor of Electrical and Computer Engineering.

Dr. Maen research interests include real-time scheduling for packet switched networks, security-aware scheduling, real-time agent-based systems, and QoS for heterogeneous networks.