

# The Performance Analysis of Low Latency Queueing Scheduling Algorithm for MANETs

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**Mukakanya Abel Muwumba\*, Odongo Steven Eyobu and John Ngubiri**

*Department of Computer Science, Makerere University, Kampala, Uganda*

**\*Corresponding Author:** Mukakanya Abel Muwumba, Department of Computer Science, Makerere University, Kampala, Uganda.

## Abstract

Delay is a major Quality of Service (QoS) metric in Mission Critical Applications and some of these include health, vehicle and inspection safety applications. Some of such applications run on Mobile Ad Hoc Network (MANET) set ups which comes with transmission challenges arising from the size of traffic packets, environmental conditions and others. These challenges cause transmission delays, packet loss and hence a degraded network performance. In this article we study a Low Latency Queueing (LLQ) Scheduling Algorithm that makes use of three priority queues each transmitting voice, video and text packets. For the purposes of improving delay performance piggy backing off video packets on voice transmission is used. The LLQ model is developed under two scenarios as follows: (I) when voice packet is delayed once and piggybacked with video on transmission. (II) when voice packet is delayed only if there is a partial video packet being transmitted. During scheduling of traffic voice packets are combined with the partial video packet. We investigate the performance variation of the LLQ in an M/G/1 queue under different scenarios and under two service distributions namely: Exponential and Bounded Pareto (BP). The numerical results for the first scenario revealed that the video packets experienced the least conditional mean response time/conditional mean slowdown, followed by voice and least were text packets under LLQ Algorithm. While for second scenario, it was observed that voice packets experienced the least conditional mean response time/conditional mean slowdown, followed by video packets and then text packets in that order under LLQ Algorithm.

**Keywords:** delay; video; voice; text

## Introduction

MANETs is a technology of wireless distributed networks which have unique properties of decentralization, infrastructure-less, self-organization, flexible networking and dynamic topology which makes them suitable for communication in the absence of fixed infrastructure [1, 2]. Unlike the traditional cellular networks (GSM, LTE-A, 5th-Generation, etc.). MANETs are capable of providing connectivity even when communication infrastructure has been destroyed [3].

MANETs have a challenge of Quality of Service (QoS) to different user applications. The Covid-19 pandemic and work from home policies greatly increased the volume of real-time voice and video services [4]. Another limitation is the short transmission range between the nodes hence communication with each other is via multiple hops [5]. Transmitting delay sensitive real time traffic like voice, video over MANETs is a challenging task because this traffic requires strict QoS guarantees.

Kakuba et al. [6] proposed the Low Latency Queueing (LLQ) algorithm to schedule video and voice packets. In [7] a size based LLQ scheduling algorithm which was an improvement to the Kakuba model was proposed. This paper studies the work in [7] and extends it to an LLQ algorithm in M/G/1 system with three queues (voice, video and text). Utilizes the optimization technique for computing the size of the partial video packet to be transmitted alongside voice packets in Queue 1 in order to fully maximize system utilization.

The rest of this paper is organized as follows. Section 2 describes the Related work. The LLQ model with three queues is presented in Section 3. The Results and Discussions are presented in Section 4. The Conclusion is presented in Section 5.

## Related Work

Teleconferencing and video on demand services in the Internet require huge amounts of bandwidth by the internet. With the aid of appropriate buffer handling techniques, the bandwidth can be managed efficiently [8]. An urgency-based packet scheduling was proposed to deliver delay sensitive data in mobile networks effectively [9]. A model has been developed to give a higher priority to voice and video traffic which is the most sensitive [10]. The model monitors and classifies all incoming traffic based on the level of their sensitivity.

Cisco Systems introduced an LLQ Algorithm which combines a single strict priority queue with Class-Based Weighted Fair Queuing (CB-WFQ). The high priority traffic is placed in the strict priority queue and it allows delay sensitive voice and video traffic to be scheduled first before the packets in the other queues. The LLQ algorithm had the following drawbacks. i.e., It could not guarantee the expected QoS level, if sensitive audio and video packets are processed in the single priority queue due to resource sharing between many applications; and the presence of bursty video packet interfered with the voice traffic from being transmitted successfully [11]. A novel Low Latency and Efficient Packet Scheduling (LLEPS) algorithm is developed to ensure low latency for real time audio and video streaming applications [12].

The behavior of queues and their traffic is monitored to address the buffer under-run problem. Shaimaa et al. [13] have demonstrated that the combination of Class-Based Weighted Fair Queuing and Low Latency Queuing (CB-WFQ-LLQ) improved the performance of multimedia applications. A scheduling algorithm that categorizes and prioritizes the real-time traffic was developed [14]. Sohail et al. [15] proposed Class-Based Low Latency Fair Queueing (CBLLFQ) Packet Scheduling Algorithm which utilizes a mapping criterion and an efficient queuing mechanism for voice, video and other traffic in separate queues. Rukmani et al. [16] proposed an Enhanced LLQ (ELLQ) with an additional Primary Strict Priority Queuing (PSPQ) for scheduling the video applications separately along with the dedicated Secondary (SSPQ) for voice applications.

Kakuba et al. [6] proposed an improved LLQ Algorithm modelled in a non-preemptive priority MMPP/G/1 queue system. Muwumba et al. [7] first adopted and improved on the Kakuba et al. [6] LLQ Algorithm in the M/G/1 queue system and then studied it under varying packet size distributions in MANETs. One limitation is this model is that it considered only two classes of traffic i.e., voice and video, and this is a research gap. This study bridges the gap by proposing an LLQ model with three Queues (voice, video and text) and investigating the performance variation of the model under different scenarios.

## The LLQ Model with Three Queues

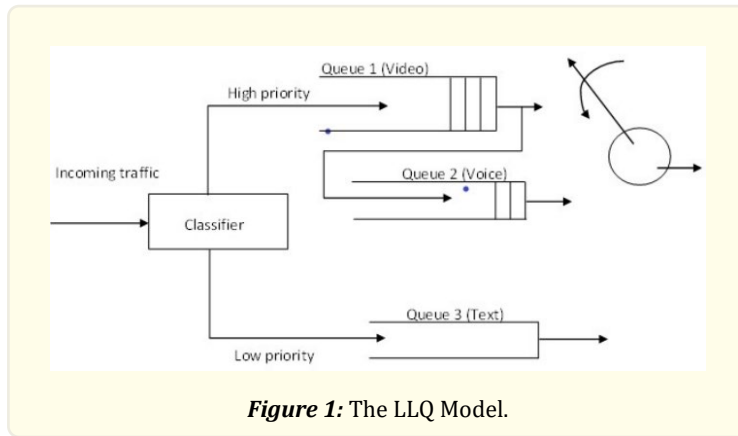
### *Preliminaries*

We considered the similar mathematical notations and metrics i.e., arrival rates, load, the conditional mean response time for voice and video packets as defined in [7]. The only new notations we are introducing are  $T(xD)$  is the conditional mean response time of text packets; CoV is the Coefficient of Variation.

Similarly, two service distributions were considered the Exponential and BP distributions. The Exponential distribution was used to represent low CoV empirical packet sizes. The probability density functions (pdf) of these distributions are given in [7, 17, 18] are omitted here due to space limitations.

**The Model**

The study assumed an LLQ model with three Queues i.e., Queue 1, 2 & 3 for voice, video and text packets respectively as represented in Fig. 1. The study assumes that in MANETs: (i) Link formation and data sharing is possible when any two nodes are within a communication range; (ii) Although there is mobility, nodes are not allowed to go beyond the communication range of each other to avoid link failure; (iii) The communication between mobile nodes is through bidirectional links; and (iv) The transmission range is constant and similar for all nodes.



**Figure 1:** The LLQ Model.

We indicated the following changes in the expressions for conditional mean response times voice, video and text packets. Firstly, we introduced the component for the CoV. Secondly, the optimization technique for computing the of the size of the partial video packet to be transmitted alongside voice packets in Queue 1. Upon arrival of voice packets at the mobile station, the LLQ scheduler will map them according to the order of increasing sizes until Queue 1 is full. When the next partial video packet does not fully fit in the remaining space in queue 1, LLQ uses the optimization technique which involves computing the unallocated space in Queue 1 and selecting a smaller partial video packet that fits the unallocated space in order to fully maximize system utilization.

**Scenario I: When voice packet is delayed once**

We assumed a tagged: voice packet of size  $x_s$  arriving at Queue 1; tagged video packet of size  $x_L$  arriving at Queue 2; and tagged text packet of size  $x_d$  arriving at Queue 3. The packets are delayed by mean residual time of the packets found in service and the mean waiting time of the earlier arrivals found in the Queue. hence; The expression for conditional mean response time of a voice packet of size  $x_s$  in Queue 1 if a voice packet is delayed once and piggy backed with video on transmission is given by;

$$T(x_s) = \frac{\rho}{\lambda} + \frac{CoV^2 + 1}{2} \left[ \frac{\lambda \overline{x_s^2}}{2m(1 - \rho_{x_s})} + \frac{\lambda \overline{x_{pr1}^2}}{2m(1 - \rho_{pr1})} \right] \quad (1)$$

The expression for conditional mean response time of a video packet of size  $x_L$  in Queue 2 if a voice packet is delayed once is given by;

$$T(x_L) = \frac{\rho}{\lambda} + \frac{CoV^2 + 1}{2} \left[ \int_2^i \frac{\lambda \overline{x_{pr1}^2} dx}{2m(1 - \rho_{pr1})} \right] \quad (2)$$

And the expression for conditional mean response time of text packet of size  $x_D$  in Queue 3 if a voice packet is delayed once is given by;

$$T(x_D) = \frac{\rho}{\lambda} + \frac{CoV^2 + 1}{2} \left[ \frac{\overline{\lambda x_S^2}}{2m(1 - \rho_{xS})} + \frac{\overline{\lambda x_L^2}}{2m(1 - \rho_{xL})} \right] \quad (3)$$

Equations 1, 2 and 3 represent the LLQ Algorithm with three queues under the first scenario.

In the simplest understanding we use the term Piggybacking to refer to a process where a voice packet transmitted by any node consists of at least its own state information and a header but also includes information of the partial video packet.

### **Scenario II: When voice packet is delayed only if there is a partial video packet being transmitted**

Now we assume a tagged: voice packet of size  $x_S$  arriving at Queue 1; video packet of size  $x_L$  arriving at Queue 2; and text packet of size  $x_D$  arriving at Queue 3. The traffic packets are delayed by mean residual time of the packets found in service and the mean waiting time of earlier arrivals plus partial video packets found in the Queues. Therefore, the expression for conditional mean response time of a voice packet of size  $x_S$  in Queue 1 if a voice packet is delayed only if there is a partial video packet being transmitted is given by;

$$T(x_S) = \frac{\rho}{\lambda} + \frac{CoV^2 + 1}{2} \left[ \frac{\overline{\lambda x_{pr1}^2}}{2m(1 - \rho_{pr1})} \right] \quad (4)$$

The expression for conditional mean response time of a video packet of size  $x_L$  in Queue 2 if a voice packet is delayed only if there is a partial video packet being transmitted is given by;

$$T(x_L) = \frac{\rho}{\lambda} + \frac{CoV^2 + 1}{2} \left[ \frac{\overline{\lambda x_S^2}}{2m(1 - \rho_{xS})} + \int_2^i \frac{\overline{\lambda x_{pr1}^2} dx}{2m(1 - \rho_{pr1})} \right] \quad (5)$$

And the expression for conditional mean response time of text packet of size  $x_D$  in Queue 3 if a voice packet is delayed only if there is a partial video packet being transmitted is given by;

$$T(x_D) = \frac{\rho}{\lambda} + \frac{CoV^2 + 1}{2} \left[ \frac{\overline{\lambda x_S^2}}{2m(1 - \rho_{xS})} + \frac{\overline{\lambda x_L^2}}{2m(1 - \rho_{xL})} + \frac{\overline{\lambda x_F^2}}{2m(1 - \rho_{xD})} \right] \quad (6)$$

Equations 4, 5 and 6 represent the LLQ Algorithm with three Queues under the second scenario.

## **Results and Discussion**

### **Introduction**

The LLQ Algorithm was analyzed under the Exponential and BP distributions. for the two scenarios. while being scheduled by the LLQ Algorithm. The performance metrics that were measured were the mean conditional response time and slowdown. The implementation of the LLQ Algorithm was done in in Matlab version R2021a.

### **Analysis under Exponentially Distributed Workloads**

We first evaluated the Algorithms under Scenarios I and II for the Exponential Distribution. Fig.2(a) and 2(b) show the performance of voice, video and text packets for the LLQ Scheduling Algorithm in terms of the conditional mean response time,  $T(x)$  as a function of packet size,  $x$  under the Exponential distribution. We note that the conditional mean response time increases with increase in packet

size for all the three traffic types (voice, video and text packets). This result confirms what earlier studies [19] indicated that the conditional mean response time,  $T(x)$  of a packet depends on the size of a packet.  $T(x)$  grows linearly with  $x$ , the packet size. Voice packets registered the best performance, followed by video and text packet was the least. The explanation for this trend is that when the partial video packet is being transmitted, there is delay created for the voice packets and performance gain for video packets. The result is spontaneous in that a delay in transmitting voice packets trigger a delay for data packets. For Scenario II: voice packet is delayed only if there is a partial video packet being transmitted; voice packets registered the best performance, followed by video and again text packets were the least.

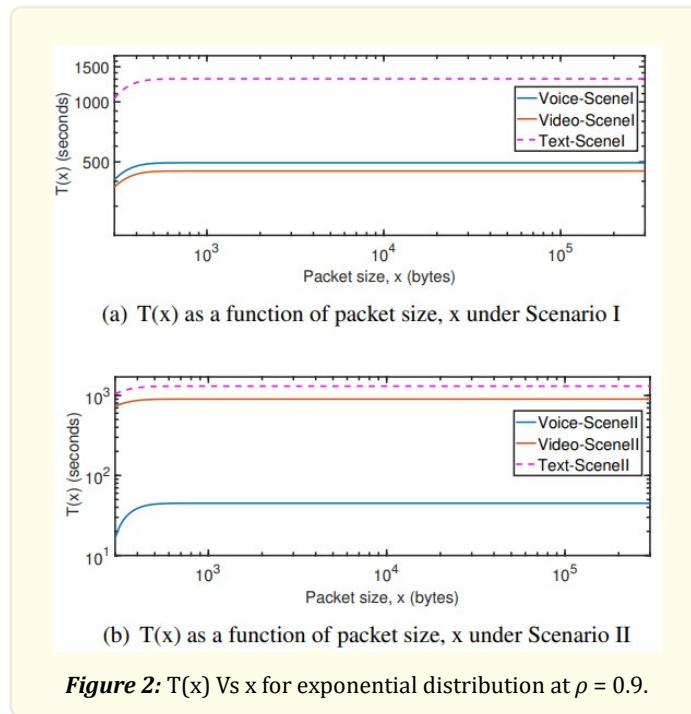
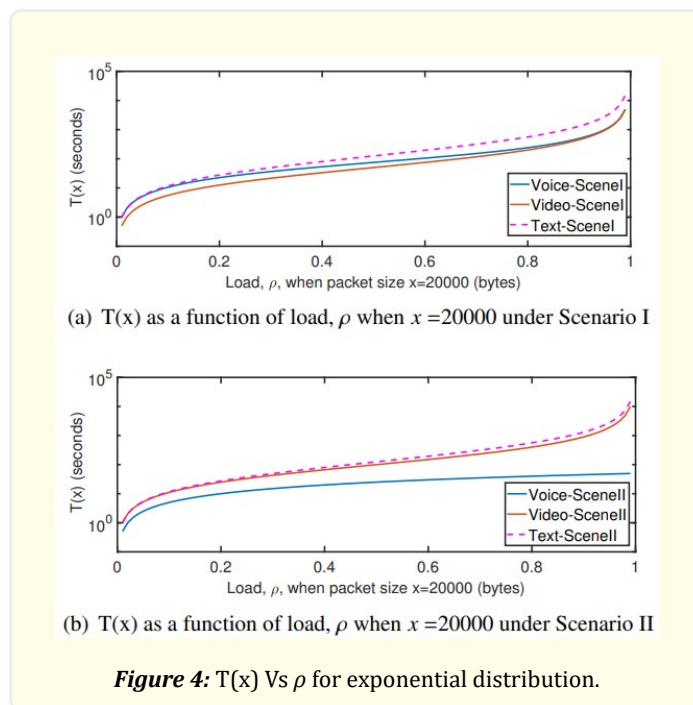
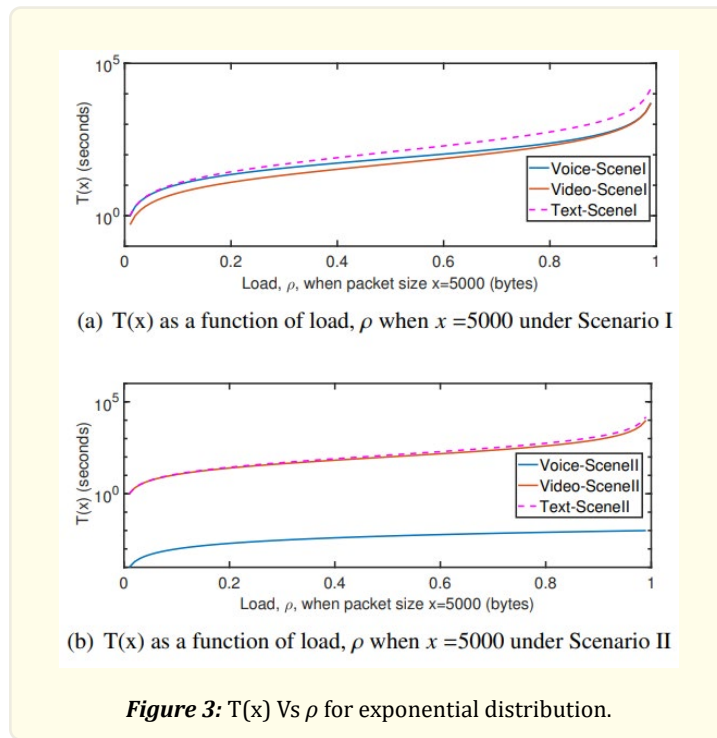
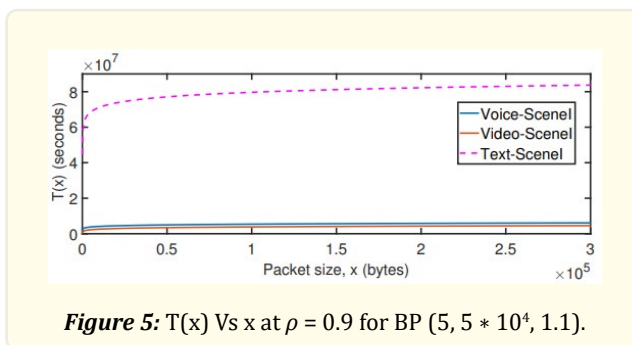


Fig.3(a) and 3(b) show the performance of voice, video and text packet for the LLQ Scheduling Algorithm in terms of the conditional mean response time,  $T(x)$  as a function of load,  $\rho$  when  $x=5000$  bytes under the exponential distribution. We observe that  $T(x)$  increases with increase in  $\rho$  for all the three traffic types. For Scenario I: When voice packet is delayed once and piggy backed with video on transmission; We can also observe from the figure that the LLQ Algorithm performs much better for video packets compared to voice and text packets regardless of the load. For Scenario II: When voice packet is delayed only if there is a partial video packet being transmitted; It can be observed that voice packets experience lower conditional mean response time under LLQ Algorithm. Video packets experience a slightly higher conditional mean response time, and text packets experience the highest conditional mean response time under LLQ Algorithm.

A similar trend is observed when the packet size  $x=20000$  bytes under the exponential distribution as shown in Fig.4(a) and 4(b).



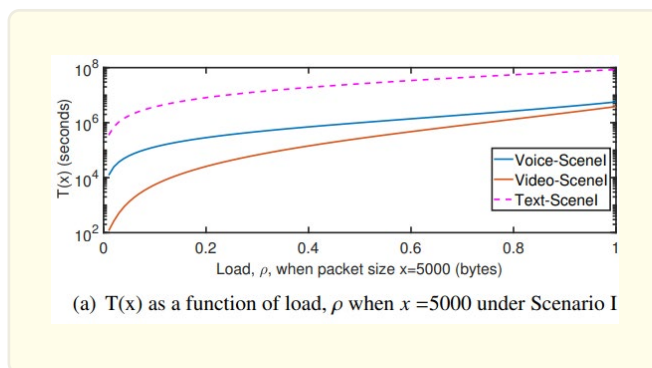


**Analysis under Heavy tailed Workloads**

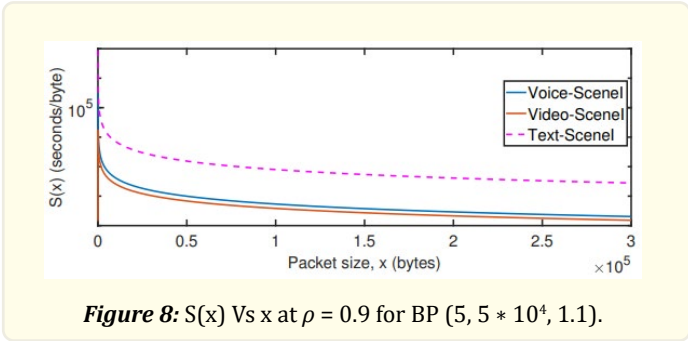
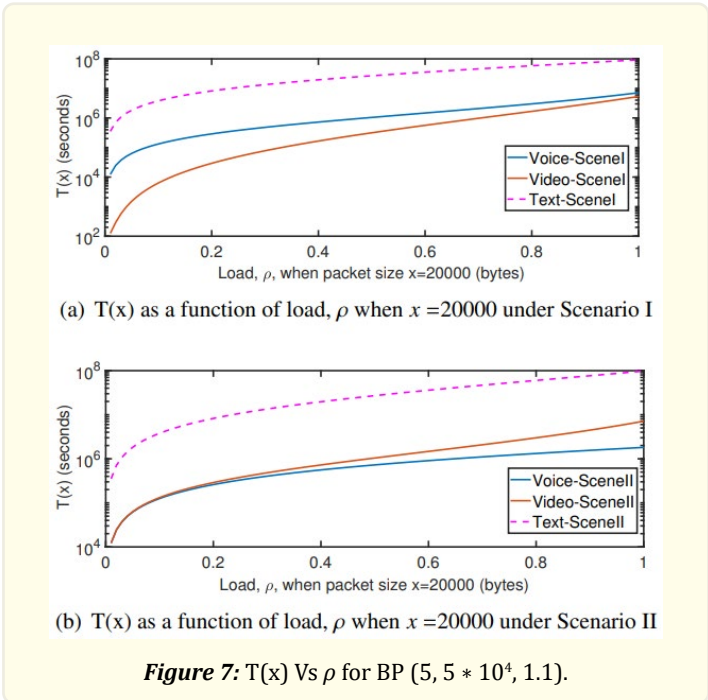
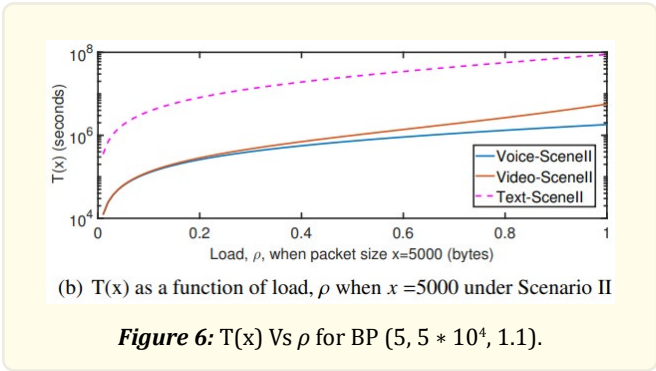
We now study the performance of the Algorithms under Scenarios I and II for the BP Distribution. The BP is one of the most commonly used heavy-tailed distributions. It is a well-known fact that heavy-tailed traffic significantly degrades the performance of Networks. Recent traffic measurements, have shown that the traffic of the Internet exhibit the heavy-tailed ness characteristic [20]. Heavy tailed distributions refer to distributions with tails that decay slower than the exponential distribution [21].

In Fig.5, we present the results of the LLQ Algorithm for the conditional mean response time for BP  $(5, 5 * 10^4, 1.1)$  as a function of packet size at  $\rho = 0.9$  for voice, video and text packets. We again note that the conditional mean response time increases with increase in packet size for all the three traffic types (voice, video and text packets). It is a known fact that  $T(x)$  grows linearly with the packet size,  $x$ . We observe that video packets registered the best performance, followed by voice and again text packets are the least. The explanation that follows is the partial video packet is transmitted alongside the voice packets in queue1 and this creates additional delay for voice packets. Text packets by default are delay tolerant and have to wait for voice and video packets to be serviced before they receive service. In Fig.6(a) and 6(b) we show the performance of voice, video and text packet for the LLQ Scheduling Algorithm for  $T(x)$  as a function of load,  $\rho$  when  $x = 5000$  bytes under the BP  $(5, 5 * 10^4, 1.1)$ .

We note that  $T(x)$  increases with increase  $\rho$  for all the three traffic types. For Scenario I: When voice packet is delayed once and piggy backed with video on transmission; We observe from the figure that the LLQ Algorithm performs much better for video packets compared to voice and text packets regardless of the load. For Scenario II: When voice packet is delayed only if there is a partial video packet being transmitted; It is observed that voice packets experience lower conditional mean response time under LLQ Algorithm. We observe that the performance of video packets is poor, and text packets had the poorest performance amongst the three traffic types. Obviously, the poor performance of text packets is attributed to the original design concepts of internet traffic that do not guarantee any strict QoS because this class of traffic is delay tolerant. A similar trend is observed when the packet size  $x = 20000$  bytes under the BP  $(5, 5 * 10^4, 1.1)$  as shown in Fig.7(a) and 7(b).









The results of  $S(x)$  for BP  $(5, 5 * 10^4, 1.1)$  as a function of packet size at  $\rho = 0.9$  are presented in Fig.8 for voice, video and text packets.

We note that  $S(x)$  decreases with increase in packet size for all the three traffic types. Just like Wierman [19], we use the figures showing ratios  $S(x) = T(x)/x$ . This is a useful measure because, in many cases, it is appropriate for response times to be proportional to packet size. We can also observe that video packets perform much better compared to voice and text packets. The explanation for this is the Algorithm does not allow pre-emption of jobs in service.

## Conclusion

The study considered three priority Queues namely; Queue 1 (voice packets); Queue 2 (video packets); and Queue 3 (text packets). The LLQ Algorithm was evaluated under varying workloads. We considered two Scenarios: Firstly, when voice packet is delayed once and piggy backed with video on transmission; And secondly, when voice packet is delayed only if there is a partial video packet being transmitted. We presented the results of the LLQ Algorithm: For Scenario I: The results revealed that the video packets experienced the least conditional mean response time/conditional mean slowdown, followed by voice and least were text packets. For Scenario II: It was observed that voice packets experienced the least conditional mean response time/conditional mean slowdown, followed by video packets and then text packets in that order under LLQ Algorithm.

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