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Comparison of Various Queuing Techniques to Select Best Solution for Eradicating Congestion in Networks

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Abstract

Congestion is an aggregation of huge amount of data in the networks that results in delay in packet delivery ratio and even huge loss of data. Queue management is a technique to minimise the congestion rate so that the data is successfully transferred from the source and destination. There are two types of queue management techniques (a) In active queue management techniques the intimation is given by the queue to the senders to slow down the packet rate as its queue buffer is about to full (b) Passive Queuing the drops rate is more in comparison with active queuing because the senders does not have any idea about queue buffer size so they are not able to lower its delivery rate and queue buffer drops the received packet when the buffer is full.
We use Network simulator 2.34 for the performance comparison as the simulator. A number of simulations are done to study the performance of various queue management algorithms like RED, BLUE, REM, and SFQ on the basis of delay, packet loss, congestion window and throughput metric. We give detail analysis of result and give suitable reason for fluctuation in the graph. This helps the new researcher to build an efficient algorithm to eradicate the major problem of congestionbased upon pros and cons of QM algorithms.

Keywords—Congestion, RED, REM, SFQ, DROP-TAIL, BLUE Algorithm

I. INTRODUCTION

Incase with wired and wireless networks when there is huge packets in the medium for its transmission to the particular destination this leads to numerous problems like delay in packet forwarding, drop of packets, loss of confidential data etc. , this is called congestion. The major factors for congestion in the networks are lower bandwidth of network, propagation delay, when the rate of incoming packet is much more in comparison with rate of outgoing packet in queue buffer size.

It is very important to control congestion in the networks for the sake of preventing attacks by the intruder like distributed denial of services, click fraud etc. moreover congestion may results in financial loss if formed by the attacker for the sake of revenge. To control congestion, several techniques are used, such as exponential back off, congestion control in TCP, priority schemes, and queue management.Their description is as follow:

(a) **Exponential back off:** A sensing technique used in CSMA/CA. In order to control the congestion sensing of channel is done by the sender before the data transmission. If the channel is idle than packet is forward by the sender [1]. Exponential back off waiting time is calculated in case with busy channel condition.

(b) **Congestion control in TCP:**Congestion is controlled on the basis of congestion window in TCP using slow start, congestion avoidance, fast retransmit and fast recovery phases [2]. Initially slow start phase occur and based upon availability of transmission medium between source and destination the size of congestion window is decided.

(c) **Priority scheme**: Assigning priorities to the packets based upon the requirement of data [3]. The data with lowermost priorities are dropped if required so that the appropriate information is transferred to receiver successfully.Higher priority is assigned to the control packets.

Attackers may also use congestion mechanism like DDOS attack, Botnet etc. to create heavy load on network for the purpose of revenge, data loss as well as financial loss. To root out the issue of congestion in network and to improve the network performance various algorithm had been proposed based queue management techniques.

1.1 QUEUE MANAGEMENT ALGORITHMS:

1.1.1 Active Queue Management (AQM) has been proposed as a router-based mechanism for early detection of congestion inside the network. The algorithms for AQM are RED, BLUE and REM.

(a) RED (Random Early Detection): Random early Detection seeks to prevent the router's queue from becoming fully used by randomly dropping packets, and send signals to the sender to slow down before the queue is entirely full [4]. RED also performs tail drop, but does so in a more gradual way. Once the queue hits a certain average length, packets enqueued have a configurable chance of being marked (which may mean dropped). Thischance increases linearly up to a point called the max average queue length, although the queue might get bigger.

Table 1: RED Algorithm

for each packet arrival calculate the average queue size Avg if Tmin ¡= Avg ¡= Tmax calculate drop probability p a with probability p a: mark (or drop) the arriving packet else if Avg = Tmax mark (or drop) the arriving packet

(b) BLUE ALGORITHM: Blue algorithm is completely based upon packet loss and link utilization whether RED is based upon instantaneous average queue length. In case of BLUE when the buffer overflow takes place the marking probability is increased thus more will be the packet drop ratio [5]. One major problem of blue is that once a flow is marked, it is tainted forever. If later the flow restrains itself, BLUE still tries to reduce its sending rate through packet drops.

Table 2: Blue Algorithm

Upon link idle event:	**Upon packet loss event:**

if((now-
last update)>freeze time)
Pm = Pm-d2;
Last update = now;
	if((now–
	last update)>freeze time)
	Pm = Pm+d1;
	last update = now;

Where Pm: Marking/dropping probability d1: Amount of increase by Pm d2: Amount of decrease by Pm now = current time Last update: last time Pm was changed	

(c) REM ALGORITHM: The motive of Random exponential marking to achieve maximum utilization with minimum delay and packet loss. It works on two principle (a) Match Rate Clear Buffer: In which user rate is compared with network capacity. (b) Sum Prices: Here outgoing links are summed and the marking probability is directly proportional to sum of link prices.

1.1.2 Passive Queue Management (PQM): the sources connected with PQM algorithms knows about the buffer status only when packet drop occurs [6]. The main advantages of using PQM is that it is easy to implement in network with less computational overheads. It does not defines the fairness among senders as single user may consumes the whole buffers. The techniques used in PQM are Drop Tail and SFQ.

(a) DROP TAIL ALGORITHM: In drop tail scheme based on first in first out (FIFO) queue policy packets are enqueued at the tail of a queue as they arrive and dequeued from the head of queue when there is capacity on the link [7]. Drop tail is the policy of dropping the arriving packet when the queue is full.

<u>Advantages</u>

- Very simple method, places an extremely low computational load on the system.
- Packets are not reordered and max delay is determined by max depth of queue.

<u>Disadvantages</u>

- A burst flow can consume the entire buffer space of queue.
- It impacts all flows equally.

(b) SFQ ALGORITHM:SFQ is called "Stochastic" because it does not really allocate a queue for each session, it has an algorithm which divides traffic over a limited number of queues using a hashing algorithm [8].Stochastic Fairness Queuing (SFQ) is a simple implementation of the fair queuing algorithms family. This leads to very fair behavior and disallows any single conversation from drowning out the rest.

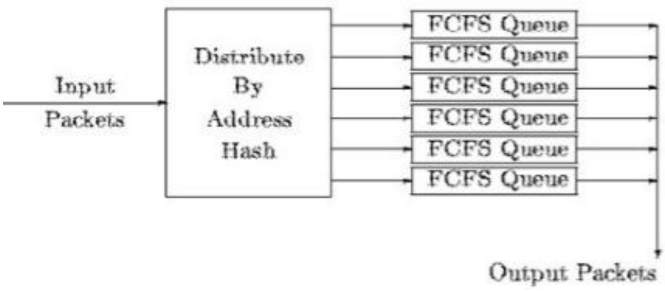


Figure 1: Stochastic Fair Queue

2. LTERATURE REVIEW:

Long Chengnianet. al. [9] present fairness performance comparison between HSTCP with DropTail router and that with AQM router. With the assumption of synchronous feedback, the RTT unfairness of TCP AIMD congestion avoidance algorithm is obvious. However, the RTT unfairness may be exponential increased in HSTCP with DropTail router. By constructing the duality model of HSTCP with AQM router, we highlight that AQM can improve the fairness performance of HSTCP. Our quantitative result shows that the unfairness of HSTCP/AQM is inverse proportional to its RTT.

Ceco, A et. al. [10]Modern data networks are faced with a congestion problem, which causes delays and data packet losses. To overcome this problem, among other congestion avoidance mechanisms, the algorithms for active queue management at the routers in the network have been proposed and implemented. In this paper, several of these algorithms are discussed. Their performances are compared on the basis of ns-2 simulations.

Hussain, S.M. et. al. [11]DDoS (Distributed Denial of Service) attacks pose a big threat to the availability of services on the Internet. DDoS degrades the performance of a network; disconnects the host and performs bandwidth depletion and resource depletion attacks. Different queuing algorithms exhibit varying performance during flooding attacks on a network. This paper presents the effect of UDP flooding on the performance of the different queuing algorithms such as Droptail (DT), Random Early Discard (RED), Deficit Round Robin (DRR), Fair Queue (FQ) and Stochastic Fair Queue (SFQ). The study shows that SFQ performs better for UDP traffic as compared to the rest of the queuing techniques.

Biyaniet. al. [12] Using the ns simulator, we perform a detailed study of the fairness and the smoothness properties of SimdNR, an enhanced version of SIMD. Our study includes TFRC and TCP congestion control protocols in both steady-state and highly dynamic network conditions through RED and DropTail routers. Our results show that SimdNR is fair to TCP through RED routers in steady-state scenario. But, SimdNR is not fair to TCP over drop tail routers. Also our results show that SimdNR demonstrates less smoothness than TFRC in steady-state scenario and superior smoothness than TFRC in dynamic network conditions.

3. RESULTS AND COMPARISON

3.1 PERFORMANCE METRICS:

3.1.1 PACKET LOSS: Packet loss occurs when one or more packets of data travelling across a computer network fail to reach their destination. The larger the value of packet loss, the more difficult it is for transport-layer protocols to maintain high bandwidths, the sensitivity to loss of individual packets, as well as to frequency and patterns of loss among longer packet sequences is strongly dependent on the application itself.

3.1.2 THROUGHPUT: The throughput is usually measured in bits per second (bit/s or bps), and sometimes in data packets per second or data packets per time slot. This measure how soon the receiver is able to get a certain amount of data send by the sender. It is determined as the ratio of the total data received to the end to end delay. Throughput is an important factor which directly impacts the network performance.

3.1.3 DELAY: Delay is the time elapsed while a packet travels from one point e.g., source premise or network ingress to destination premise or network degrees. The larger the value of delay, the more difficult it is for transport layer protocols to maintain high bandwidths

3.2.SIMULATION SCENARIO

We have design a simulation scenario for extensive research. The simulation scenario are

Figure 2. Simulation scenario

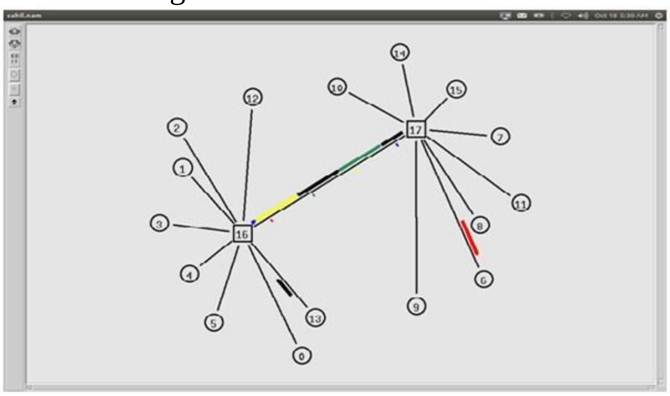


Table 2: simulation parameter

Parameter	**Value**
number of senders	6 TCP and 2 UDP flow
bandwidth between senders and routers	10 Mb
bandwidth between the two routers	1.5 Mb
maximum buffer size	50
delay between senders and routers	3 ms
delay between the two routers	20 ms
max bound on TCP agent window size	25
TCP packet size	1460 bytes
Udp packet size	1000 bytes
Traffic type	CBR(Constant bit rate)
RED max threshold	equals to the maximum buffer size
RED linterm	1
RED q weight	0.002

3.3 SIMULATION RESULTS:

We have compared RED, BLUE, REM and SFQ in various simulation scenario with performance metrics like Throughput, Average end to end delay, Congestion window size and packet loss.

3.3.1 THROUGHPUT

Below are the comparison of various AQM techniques. Those graphs are printed by analyzing the trace file. The green lines are instant throughput. And red lines are average throughput. With the graphs, we can clearly see how the throughput changes.

Figure 3: Throughput analysis of RED queue

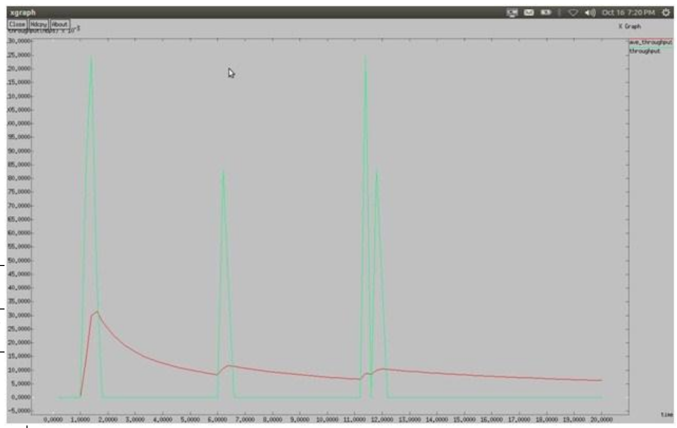


Figure 4: Throughput analysis of REM queue

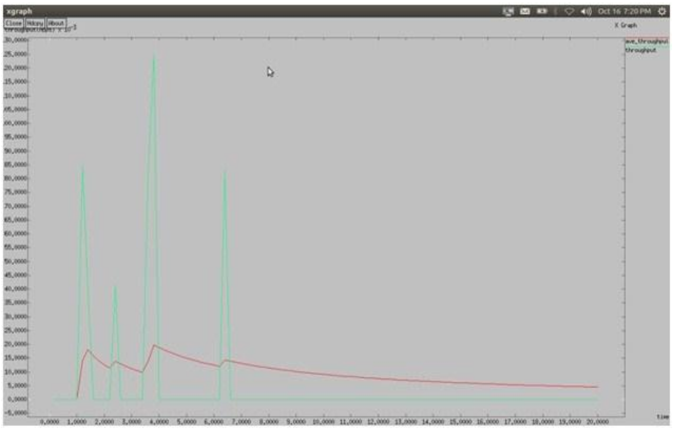
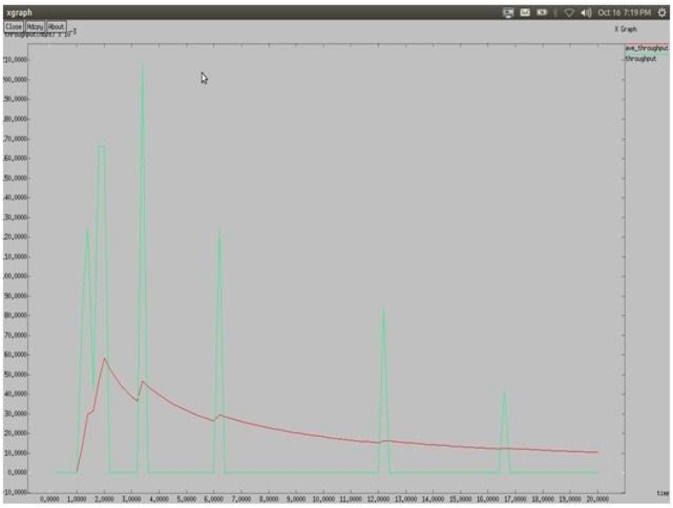


Figure 5: Throughput analysis of Blue queue



SFQ has highest throughput because it maintains a queue for all the flow, the chance of packet contention for standing in queue has decreased. So only it faces contention for the link and packet will be dropped when the single queue for the flow has full that's why the throughput of the SFQ has much more compare to other queuing mechanism.

3.3.2 CONGESTION WINDOW

Presents the congestion window size for different TCP sources in RED, REM, Blue and SFQ techniques. Here we observed that the size of congestion.

Figure 6.Congestion Window analysis of RED queue

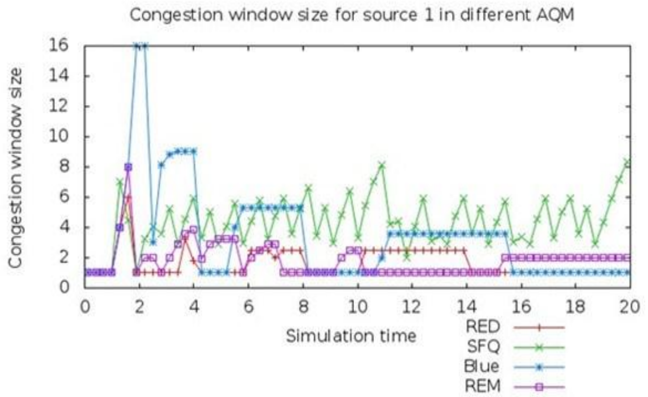


Figure 7. Congestion Window analysis of BLUE queue

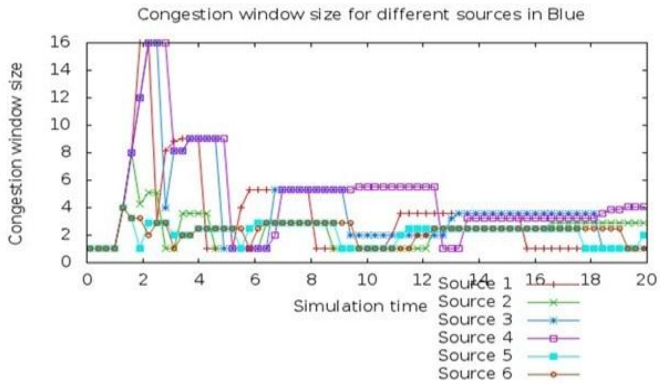


Figure 8. Congestion Window analysis of REM queue

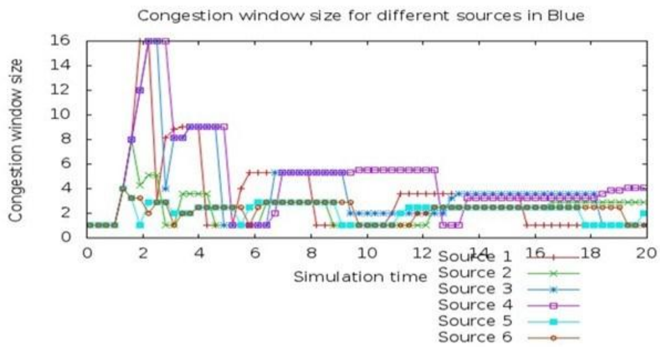
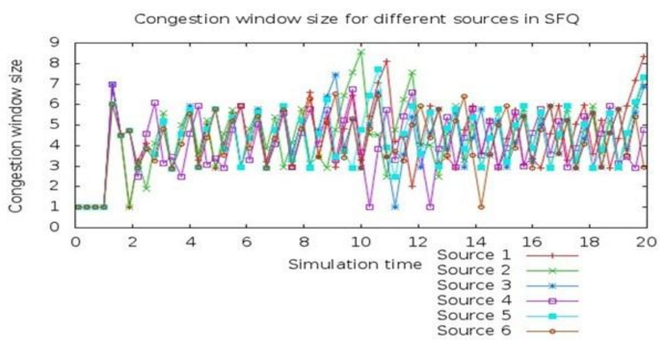


Figure 9.Congestion Window analysis of SFQ queue



Window more fluctuates in case of SFQ because in this case the more data present in the network and TCP connection send more data. Here the bottleneck link does not drop the packet of sources for the less space in the queue. In case of Blue the size of congestion window is more because it has less packet loss due to its underline technology. In case of RED and REM the congestion window size is constant so it have less congestion variance in the network. RED and REM uses different type of strategy to control the packet loss in the network. The fluctuation of congestion window size gives the idea about how much of the traffic available in the network as well as what number of packet was dropped because of packet drop it can be data as well as Ack packet.

3.3.3 Avg end-to-en Delay of AQM

The below figures presents the average delay of different AQM techniques on the simulation time.The RED and SFQ have low end to end delay on different flow so these two queue mechanism can be used in delay sensitive application. On the other hand Blue has higher end to end delay compare to other algorithm so these algorithm cannot be used in delay sensitive network.

Figure 10.Delay analysis of RED queue

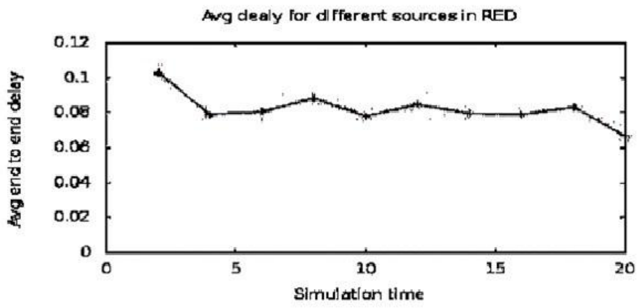


Figure 11.Delay analysis of REM queue

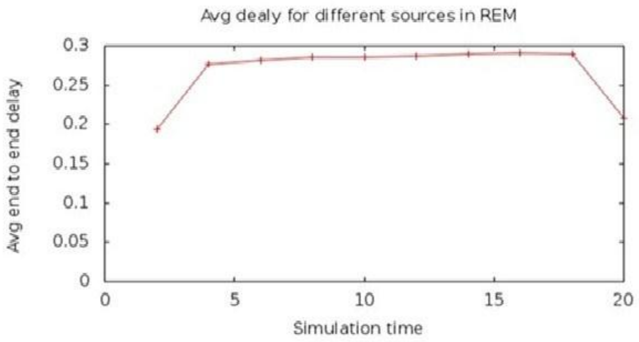


Figure 12.Delay analysis of Blue queue

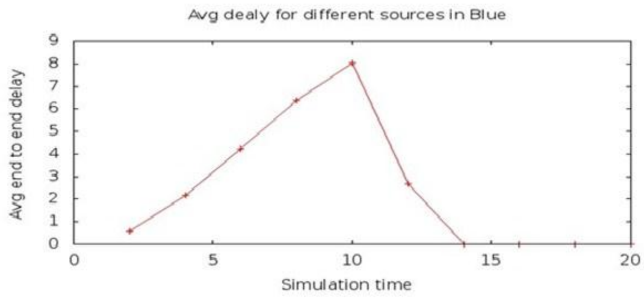
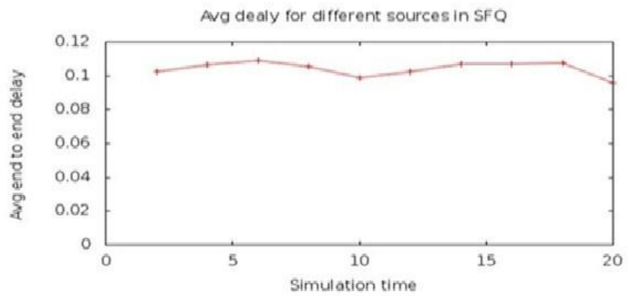
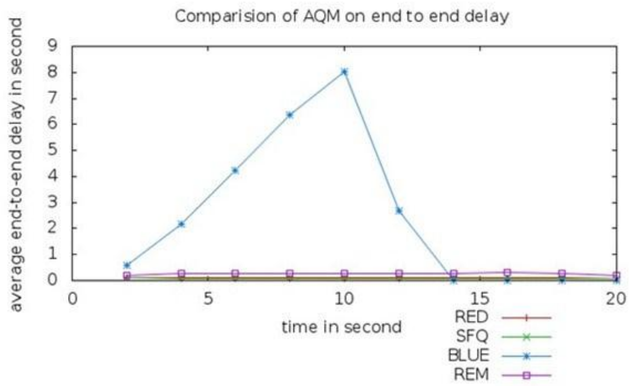


Figure 13.Delay analysis of SFQ queue



The Figure 14 Presents average end to end delay of four AQM techniques. In which show that Blue has higher end to end delay compare to other queue techniques.

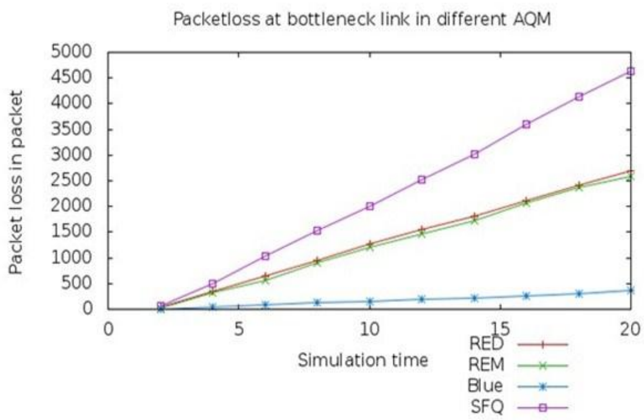
Figure 14. Delay analysis of different queue



3.4 PACKET LOSS

The below figure shows about the loss rate occurred in RED, Blue, SFQ, and REM respectively. It has been observed that loss rate smoothly increasing as in time. We got the drastic change in loss in case of SFQ because of the increase in traffic due to increase throughput of the network. It has been concluded that SFQ could achieved higher loss rate at higher bandwidth (at some specific bandwidth but it could not be happen). But Blue shows smooth decrease in loss rate over increase in time. So Blue provides reliable service to the UDP flow

Figure 15.Packet loss analysis of different queue



4. CONCLUSION

In this paper, the performance of four AQM schemes, selected from amongst the many published over the past years, has been evaluated. We have compared RED, BLUE, REM, and SFQ algorithms. SFQ can be used for those network where we have to utilize the link capacity on different sources equally. Blue queue is used for those network where minimum packet drop is required. RED and REM queue model perform well for delay sensitive network. AQM algorithms are absolutely useful because the management of packets to avoid congestion occasionally requires exceeding hardware capabilities.

For future work, we plan to extend the simulation for the new algorithm which would comprise all the advantage of each algorithm. There would be hybridization of RED, Blue SFQ, and REM to provide the better results.

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