Developed Model to Control Congestion on Converge Network

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***Abstract*— Congestion control techniques like Active Queue Management (AQM), Carrier Sense Multiple Access/Carrier Detection (CSMA/CD) have not proven to be very efficient in the presence of overwhelming complex converge network. Thus, vast packets in a complex converged network leads to collisions, network degradation and high degree of packet loss. Bandwidth utilization factor has a high effect on the network such that controlling the level of utilization via the management of the number of users and the amount of packets on the network rendered the latency very insignificant. As a consequence of this, high throughput and very minimal packet loss was achieved in the experiment. This was confirmed analytically by varying the utilization factor between 40% and 90% while keeping other parameters in the experiment constant.**

Keywords**:** Congestion, Active Queue Management, Converged Network, Bandwidth.

I. INTRODUCTION

The study on congestion control was pioneered by Van Jacobson and this was called the TCP/IP congestion control [1]. The High – Speed *TCP* [HS-TCP] for Large Congestion windows was introduced by Sally Floyd as a modification of TCP's congestion control mechanism for use with TCP connections with large congestion windows [2]. It overcomes Standard TCP's difficulty of achieving a large congestion window in environments with very low packet drop rates. High-speed TCP proposes a small modification to TCP's increase and decrease parameters. It is designed to have a different response in environments of very low congestion event rate, and to have the standard TCP response in environments with packet loss rates of at most 10−2. In environments with low packet loss rates (typically lower than 10−3, it is possible to ignore the more complex response functions that are required to model TCP performance in more congested environments with retransmit timeouts [3].

We considered and modeled an existing complex corporate converge network in order to evaluate and profound realistic solution to controlling congestion on the network. To derive the mathematical model the following postulations were considered:

- The delay along the network path was summarized into three classes: *propagation, serialization* and *queue delay.*
- The packet sizes from the nodes were of the same length and at the router the service time T_s, for the packets was constant.
- During network congestion, packet loss indications were exclusively "through triplicate acknowledgement", no timeout was considered.

Figure 1 shows the sources of delay on a converged network between the sources Host A to Host N on the network, with model processing delay, packet queuing and propagation delays along the link. Based on these assumptions, the net delay on the network was given as follows: *Net delay = Propagation Delay* (T_p) *+ Serialization Delay* (T_s) *+ Queue Delay* (T_q)*.*

The propagation delay is described as the time it takes a signal to physically traverse the network path from source to destination [4]. This is a function of the distance of separation between the sender and the receiver and also the speed of light. It is assumed that any signal passing through a Fiber or wire does so with two-third of the speed of light *'c'* [5]. Taken *x* as the distance across the network path from the source to the destination, propagation delay is given by:

Propagation delay, $T_p = \dfrac{x}{0.667c}$ *(sec),* (1)

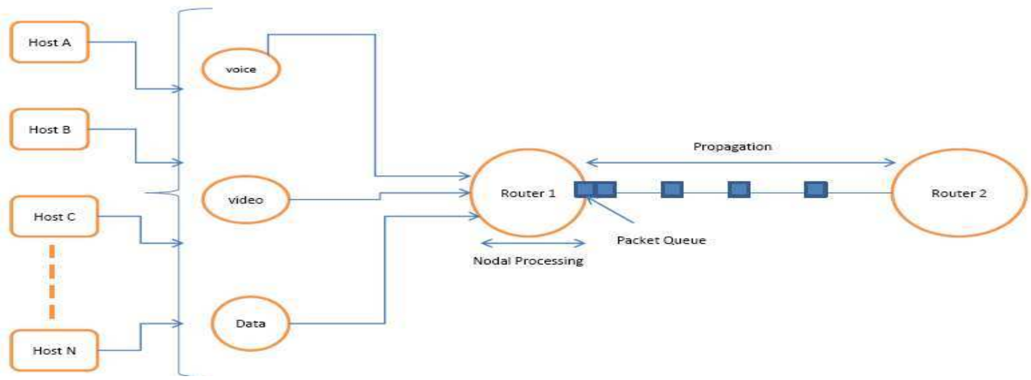


Figure I Sources of delay on a converged network. (Hosts A to Host N: packet flow sources)

For a sender to transmit a bit of signal into the outgoing queue, the serialization delay was given as follows [10 - 12]:

Serialization Delay, $T_s = \dfrac{N}{A} = \dfrac{N}{(1-\rho)B}$

as A = (1 − ρ) B and

$\rho = \frac{\lambda}{\mu}$

$$= \frac{N\mu}{(\mu-\lambda)B} \quad \textit{(sec)}, \qquad (2)$$

where *N* - number of packets size sent by the sender
A - Available bandwidth on the channel,
ρ- Bandwidth utilization factor
B - Bandwidth capacity of the link.
λ - Inter-arrival rate of the packets,
μ - mean service time at the node.

Queue delay, $T_q = \dfrac{l}{B} = \dfrac{\lambda^2}{(1-\rho)B\mu}$ (sec), also,

$l = \frac{\rho 2}{[1-\rho]}$ (3)

where *l* is the length of queue.

When two nodes are connected together, the queues experienced by each packet traveling through the medium are independent of any successful arrival of the packet [6]. Latency is the summation of delays inherent on the network [7]. Thus, the net delay was deduced as follows:

$$T_p + T_s + T_q = \frac{x}{0.667c} + \frac{N\mu}{(\mu-\lambda)B} + \frac{\lambda^2}{(1-\rho)B\mu} \qquad (4)$$

In most cases the Maximum Transmission Unit (MTU) was less than or equal to the Maximum Segment Size (MSS) (MTU ≤ MSS),
Therefore,

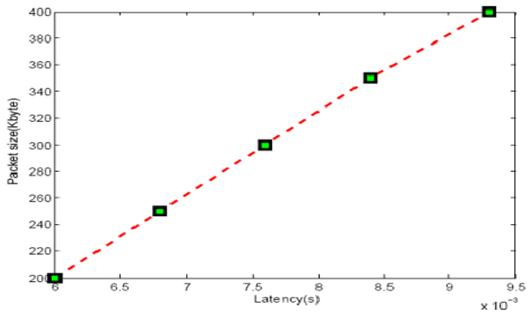
$$Throughput_{\max} = MTU \Big/ \left(\frac{x}{0.667c} + \frac{N\mu}{(\mu-\lambda)B} + \frac{\lambda^2}{(1-\rho)B\mu} \right) \qquad (5)$$

II. Analysis of Results and discussion

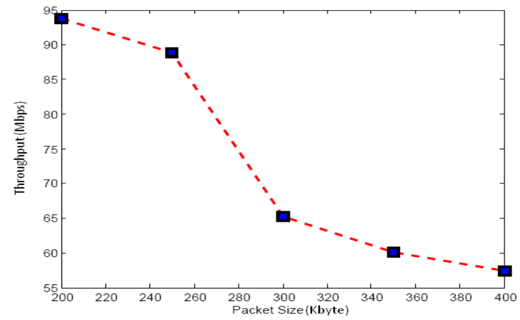
The net delay was obtained by considering the total delay experienced by the complex network i.e. *propagation delay, serialization delay, and queue delay*. To control the level of bandwidth utilization factor via the management of the number of users and the amount of packets on the network, data were generated from the mathematical model. The results were analyzed as follows: Firstly, we considered the following input parameters: Distance = 100m, Packet Size (Kbyte) = 200 – 400, MTU = 1500, Bandwidth Capacity = 100Mbps and Bandwidth utilization factor $\rho = 0.4$. The results of input parameters are shown in table I.

Table I Generated Results given $\rho = 0.4$

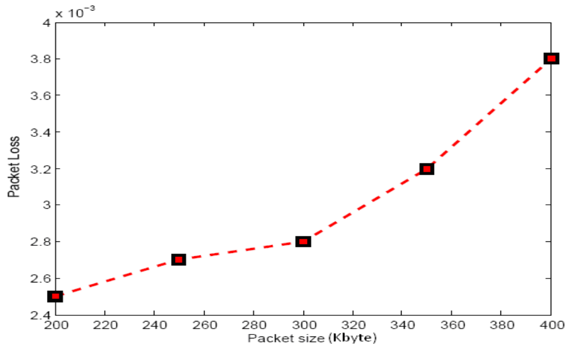
Packet Size, (Kbyte)	**Latency, (ms)**	**Throughput, (Mbps)**	**Packet Loss Rate, %**
200	0.006	93.75	0.0025
250	0.0068	88.9	0.0027
300	0.0076	65.2	0.0028
350	0.0084	60.1	0.0032
400	0.0093	57.4	0.0038



(a)



(b)



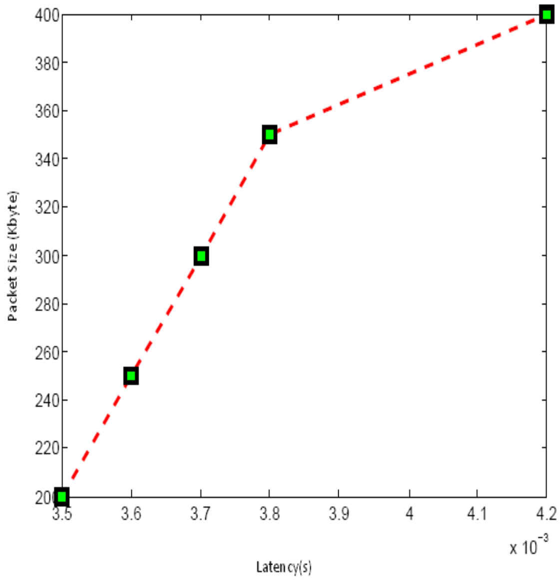
(c)

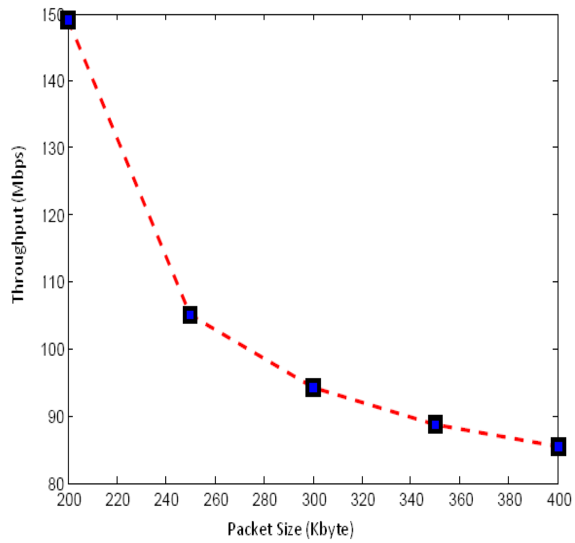
Figure 2 (a) Response of latency to packet size, (b) Graph of response of packet size to throughput, (c) Graph of packet size against packet loss.

Figure 2(a) shows the response of latency to packet size. Increased in the packet sizes resulted in an increased in the latency. It was observed that the increment was on a gradual note, which suggested that the network had not reached a congested state. Figure 2(b) represented the relationship between packet size and throughput. An increase in the size of the packets led to the reduction in throughput. It was observed from the graph that packet size was inversely proportional to throughput and the higher the packet size, the lower the throughput. A sharp change was observed in the throughput when the packet size was increased to 300Kbytes, with a reduction in value from 89Mbps to 65Mbps, this further point to a gradual packet build-up on the network. At the utilization value of $\rho = 0.4$, the packet loss underwent a steady increment, though mostly negligible, however it was of considerable value when compared in respect of the packet size between 200Kbytes and 400Kbytes with a difference of 13packets, Figure 2(c). This proved that the network performed efficiently when the network dropping of packets has a very negligible value.

Using the same mathematical model, however with *bandwidth utilization factor of* $\rho = 0.9$ considered, the average throughput reduced drastically as compared to the same packet sizes with lower bandwidth utilization factor of 0.4. The highest throughput obtained was 37.5Mbps as compared to 93.7Mbps when the utilization factor was 0.4. The latency also increased significantly from 0.042ms at 200kbyte to 0.079ms at 400kbyte. The packet loss rate was also significant, as the number of packet dropped in the network was higher (37 packets) as compared to when the utilization was 0.4. This indicates a better performance at lower bandwidth utilization factor of $\rho = 0.4$ compared to the utilization of $\rho = 0.9.$

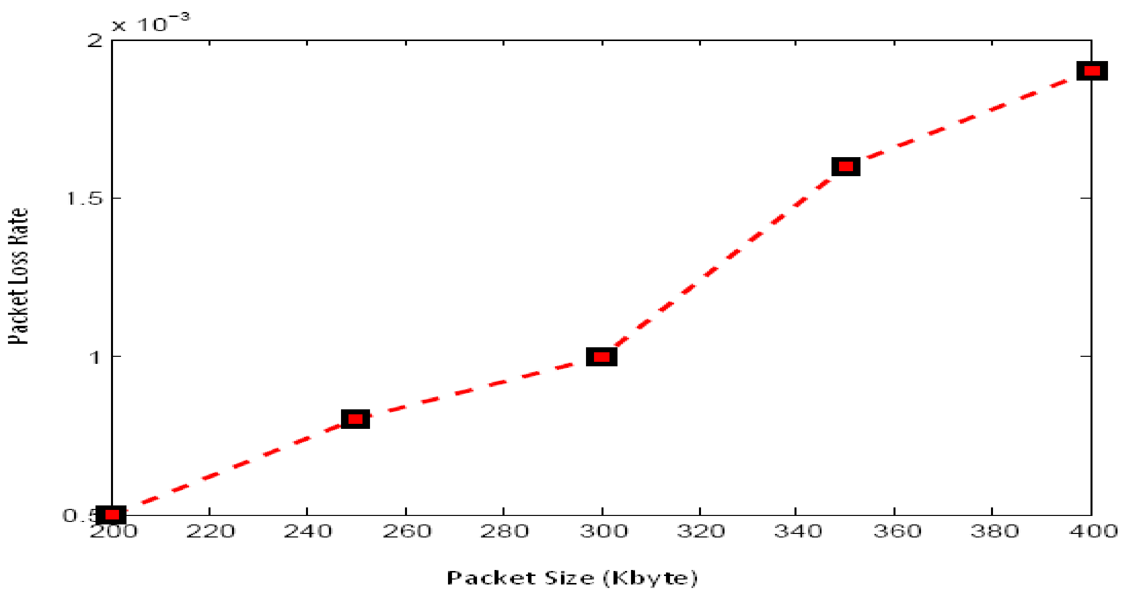
Secondly, we considered the following input parameters: Distance=100m, Packet Size (Kbyte) =200 – 400, MTU=1500, Bandwidth Capacity=150Mbps and Bandwidth utilization factor $\rho = 0.4$. The result shows in Table 2.


Packet Size (Kbyte)
Latency(s)
x 10^-3


Throughput (Mbps)
Packet Size (Kbyte)

(b)

(a)


x 10^-3
Packet Loss Rate
Packet Size (Kbyte)

(c)

Figure 3 (a) Graph of packet size against latency, (b) Graph of throughput against packet size and (c) Graph of packet size against packet loss with utilization factor of 0.9 with 150Mbps.

Table II Generated results for Bandwidth capacity = 150Mbps & $\rho = 0.4$

Packet Size (Kbyte)	**Latency (ms)**	**Throughput (Mbps)**	**Packet Loss Rate**
200	0.0035	149	0.0005
250	0.0036	105	0.0008
300	0.0037	94.2	0.001
350	0.0038	88.6	0.0016
400	0.0042	85.4	0.0019

The results in Table II are plotted in Figure 3 (a), (b), (c). This shows an increased in the bandwidth capacity. In this case the highest achievable throughput was 149 at 200 Kbyte. While the throughput for an average file size of 400 Kbyte gave 85.4 with a lower packet loss as against what was observed when the bandwidth was . Also, the latency was small compared to what we had in Figure 2. This clearly indicates a better network performance when the bandwidth was increased from 100Mbps to 150Mbps keeping bandwidth utilization factor constant at 0.4.

The bandwidth capacity increment generated a direct improvement on the throughput as compared with when the bandwidth was 100Mbps, as there was better network performance with a higher throughput and a distinctive lower packet loss. The latency value is very low compared to when the bandwidth was 100Mbps. These further points to better network performance with a higher bandwidth capacity of 150Mbps at a utilization factor of 0.4 compared to when the bandwidth was at 100Mbps.

Also, the mathematical model was used with the same input parameters as Table 2, however, with utilization factor of $\rho = 0.9$. The maximum throughput achieved was 53.5Mbps at 200kbyte, and the minimum 43.7Mbps at 400kbyte. Latency rose to as high as 0.038 compared to 0.0042 when the utilization was 0.4. This is an important model to ascertain the behavior of complex networks, considering the effect of utilization.

III. CONCLUSION

In this paper, a model for controlling the impact of link congestion on a converged network was presented. The model expresses throughput, latency and packet loss rate as a function of TCP and network parameters. The model was used to generate results with some assumptions with regards to distance, constant range of packets, network bandwidth, and the variation of the bandwidth utilization factor between 40% and 90%. It showed some encouraging results when the bandwidth utilization was modified at 40%, but not when the utilization rises to 90%. At 90% the network began to show very feasible signs of congestion with low network performance.

REFERENCES

[1] Jacobson V. *"Congestion Avoidance and Control" ACM, Proceedings of SIGCOMM, Stanford, CA, 1988.*

[2] S. Floyd. *"Highspeed TCP for large congestion window", Internet Draft draft-floyd-tcp-highspeed-01.txt, February 2003.*

[3] Eric He, Rajkumar Kettimuthu, Sanjay Hegde, Michael Welzl, Jason Leigh, Chaoyue Xiong and Pascale Vicat- Blanc Primet, *"Survey of Protocols and Mechanisms for Enhanced Transport over Long Fat Pipes", Data Transport Research Group,2003/2004.*

[4] Benedict Chung Wong *"Two finite queues in tandem with pure delay and overflow" Technical Report submitted to Dept. of Industrial and System Engineering, Ohio State University, Columbus, Ohio, July 1975.*

[5] Michele C. Weigle, Pankaj Sharma, and Jesse R. Freeman IV, *"Performance of Competing High-Speed TCP Flows" Networking 2006. Networking Technologies, Services, and Protocols; Performance of Computer and Communication Networks; Mobile and Wireless Communications Systems, pp. 476-487, 2006.*

[6] William Stallings *"Queuing Analysis "A practical guide to an essential tool for computer scientists" Prentice Hall 2000.*

[7] S. N. John, R. E. Okonigene, A. Adelakun: *Impact of Latency on Throughput of a Corporate Computer Network, Proceedings: The 2010 World Congress in Computer Science, Computer Engineering, and Applied Computing (WORLDCOMP'10), Annual Summer Conference on Modeling Simulation & Visualization Methods (MSV'10), Las Vegas, Nevada, USA, July 12-15, 2010, pp.282-287.*