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# Impact of Router's Buffer Size on Packet Loss Due To Congestion in TCP

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***Abstract*—continuously packet loss results in poor throughput. In wired network, packet loss is mainly due to congestion in path (neglecting any link failure). To avoid the packet loss queues of limited size are maintained at routers which stores packets for further processing rather than dropping it. In this paper, we will analyze the impact of buffer size or queue limit at router on packet loss. Results are taken with the help of ns2 simulator.**

**Keywords—** ***Wired network, TCP, Congestion, Router, Buffer's size***

## I. INTRODUCTION

In wired network, packet losses are due to congestion and duplicate acknowledgements and timeout give indication of congestion [1] and fast retransmit and fast recovery phases start and sender reduces size of congestion window to half and linearly increase congestion window [2] in congestion avoidance phase, which result in lower transmission rate to relieve the link congestion and which further cause low throughput. The congestion control algorithms in TCP [3] are based on assumptions that if links are reliable and hosts are stationary then in such network the channel error rate are very low.

Packet loss is condition in which data packets are transmitted correctly at source end, but never arrive at the destination end. This might be because of network conditions are poor or the packet was deliberately dropped at a router because of internet congestion. Each data packet has a limited time to reach its destination before it is considered lost. This is parameter to prevent overcrowding of packets on the network. This packet loss event can cause noticeable effects in all types of communications like in videoconference environment, it can create jitter, In audio communication, it can cause frequent gaps in received speech etc.

Routers are configured for best-effort packet forwarding. This means that the router receives packets and forwards all packets. If the router is unable to process a packet immediately, the packet is queued. If the queue is full, the packet is dropped. Packets are typically processed on a first-come, first-served basis.

Control, security, reliability and speed are the primary benefits of using wired network. One great advantage of having a wired network is the control it provides. If a physical connection is needed to access the network, the business is in full control of who and what gets online. In this paper, we will try to make wired network more reliable by identifying the packet loss and we will see the impact of varying router's buffer size on packet loss.

The rest of this paper is organized as follows. In section II, Literature review is given. Section III gives overview of congestion, Router buffer size, packet loss and also includes Environment Setup. Section IV, presents results and discussions. Conclusion and future work are provided in section V.

## II. LITERATURE SURVEY

In [1], acknowledgement timeout or duplicate acknowledgements indicate packet loss due to congestion in path. On timeout appropriate action must be taken to reduce load otherwise, network may get into infinite cycle of packet loss and result into low throughput. To reduce packet loss, author proposed a scheme named cute which help to reduce packet loss. They conclude that on timeout window size should be reset to one and this will limit the packet loss.

In [4], to reduce congestion, authors proposed a new queue management algorithm which is modification of drop tail algorithm (based on fixed queue limit) and based on variable output queue limit. They dynamically changed output queue length in wired network and analyzed that small change in virtual output queue length gives good performance and low packet loss when buffer size of input port is fixed.

In [5], authors discussed that buffer sizing based on bandwidth delay product can lead to both poor utilization and high loss rate or over estimation of buffer requirement. So, they proposed an analytical framework for the optimal choice of the router buffer size in which Lagrange function corresponds to a linear combination of the average sending rate and average delay in the queue. They conclude that as the number of long-lived TCP connections sharing the common link increases, the required minimum buffer size to achieve full link utilization reduces.

In [6], authors described Stanford model which is use to achieve full utilization of network link. They discussed issue on use of small buffer's size in router. Size of router buffer is set to either default value specified by manufacturer or by bandwidth delay product rule. They considered that buffer size should be equal to capacity of link multiplied by RTT (round trip time).On one side where small buffer can reduce queuing delay, full utilization of link and on other side it may cause high loss rate. so, they considered loss rate and queuing delay as important issue in the buffer sizing problem.

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In [7], authors described that Stanford scheme for buffer sizing focus only on link utilization and ignore the resulting loss rate. So, they introduced the formula to calculate minimum buffer requirement for drop tail link by applying constraint on minimum utilization, maximum loss rate and maximum queue delay. They conclude that to limit maximum loss rate, buffer size be proportional to number of flows that are bottleneck at that link.

In [8], authors discussed the problem of choosing buffer size of routers analytically as multi-criteria optimization problem. They used Lagrange function corresponds to a linear combination of the average sending rate and average delay in the queue. They conclude that minimum required buffer is smaller than bandwidth delay product and Stanford model.

## III. PROPOSED WORK

Wired networks provide users with plenty of security and the ability to move lots of data very quickly. Wired networks are faster than wireless networks, and they can be very affordable. To give better network service quality, throughput must be high. Less packet loss will result in more throughputs. The congestion can be main cause of data packet loss. Congestion is network state where a node or link carries the data more than capacity of link which results in poor network service quality, queuing delay, frame or data packet loss etc. In a congested network, response time slows with reduced network throughput. So, in this paper we will try to make wired network more reliable by increasing the throughput. We will identify the variation in packet loss on increasing buffer size. When a packet enters the router, it stores the packet into buffer for later processing. Environment setup as follows:

### *A. Environment Setup*

Consider a wired network in which sending packets will pass through many links and routers before reaching at final destination. A TCP destination agent receive data packet and send acknowledgement to sender. In simulation, wired network of 34 nodes are consider in which 30 nodes will engaged in data transfer and 4 nodes will act as intermediate routers. The packet size sent by TCP agent is 1024 bytes. Delay considered is 10ms and queue policy is Drop Tail. The bandwidth of all links is taken as 2Mb. We will increase router buffer's size from 10 (i.e. 10 data packets can stored at router) to 20 and so on till we get 0% packet loss. Window size at sender side will increase exponentially and when congestion avoidance phase starts, it increases linearly.



## IV. RESULTS AND DISCUSSIONS

In this section, we present simulation results for 2 scenarios.

Scenario 1: All nodes are engaged in data transfer having one to one correspondence (15 Source Destination Pairs or 30 nodes communicating at a time) i.e. 1 source node is communicating with only 1 destination node (worst case scenario) as shown in Fig 1.

Scenario 2: 20 nodes are involved in data transfer having one to one correspondence (10 Source Destination Pairs or 20 nodes communicating at a time) i.e. 1 source node is communicating with only 1 destination node as shown in Fig 2.

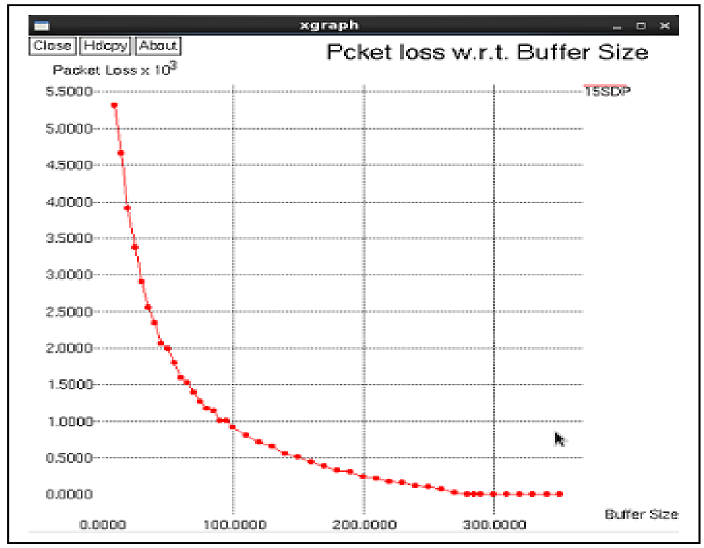


Fig. 1 Packet loss w.r.t. Buffer size for 15SDP

In Fig 1, on increasing buffer size of routers, packet loss decreases. Packet loss is shown as factor of $10^3$. When buffer size is taken 10 (i.e. 10 packets can store at a time in router's queue if more packets arrive and the packets will be discarded) then packet loss is 3.39%, at buffer size 20 packet loss is 2.5%, at buffer size 30 packet loss is 1.9%, at buffer size 40 packet loss is 1.5% and so on. So it can be seen that packet loss is continuously decreased as the buffer size is increased. At buffer value 280, packet loss is decreased to 0% i.e. no packets will be dropped. It is shown that if buffer value is increased from 280 to 300, it still remains 0%. So, from this simulation we conclude that when all nodes are communicating, packet loss is continuously decreasing as the router's buffer size is increased and reach to zero at some value and remain constant i.e. zero.

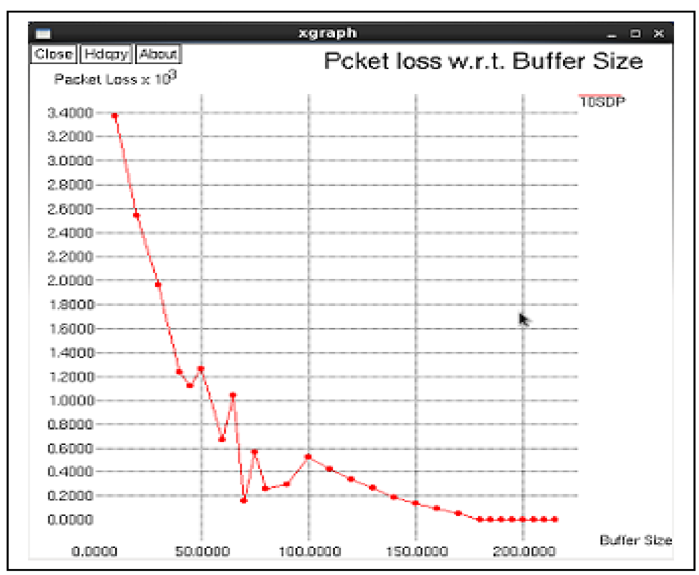


Fig. 2 Packet loss w.r.t. Buffer size for 10SDP

In Fig 2, on increasing buffer size of routers, packet loss decreases up to some value of queue size. Packet loss is shown in $10^3$. When buffer size is taken 10 then packet loss is 2.21%, at buffer size 20 packet loss is 1.68%, at buffer size 30 packet loss is 1.31%, but at buffer size 80 packet loss is 0.70% and at buffer size 90, packet loss decreases to 0.20% but at buffer size 100, packet loss increases to 0.35% and after that it starts decreasing with increase in buffer's size. This is due to increase in enqueue packets. If more packets are enqueued and router's Buffer size is not capable to handle packets then those packets are dropped. At buffer value 180, packet loss is decreased to 0 i.e. all packets are processed, It can be seen that if queue size is increased from 180 to 200, it still remains 0%. So, from this simulation we conclude that when average number of nodes are communicating then packet loss is constant i.e. 0% after some value of buffer's size.

## V. CONCLUSION AND FUTURE SCOPE

In this paper we have described affect of Router's buffer size on packet loss. We see that in worst case scenario i.e. where all nodes are communicating to others, increase in buffer size will surely decrease the packet loss. In case of less source destination pair the packet loss decreases to some value on increasing router's buffer size and shows some up and down for in between values due to increase in incoming packets in router's queue and after that it again starts decreasing and become zero after some particular value of queue size. So, we conclude that the router's buffer size must be three times greater than the total nodes in environment and by increasing router buffer size throughput will also increase. This work can be extended by estimating packet loss in case of random losses like link failure etc. which are another causes of packet losses. In case of link failure, packet can be loss due to time out.

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