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Rate Based Pacing with Various TCP Variants

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ABSTRACT: TCP congestion control has been intended to ensure the internet constancy along with impartial and an efficient allocation of the network bandwidth. To reduce the congestion collapse, they are several congestion control algorithms, namely TCP Forward acknowledgement (TCP Fack), TCP Selective acknowledgement (TCP Sack), and Rate Based Pacing are discussed here. The main purpose of this paper is to compare and analyze the various congestion control algorithms which are available in TCP/IP protocols suite. In this paper, we compare and analyze the performance of different TCP congestion control algorithms by considering the performance metrics, Throughput, end to end delay, Jitter, Packet Delivery Ratio and packet loss. A table has also specified which show the comparison results of the TCP variants. This analysis will be helpful in determining the best variants among the various TCP congestion control algorithms to ensure the efficient data transfer, reliability, and speed.

Keywords*:* Transmission control protocol (TCP), delay, TCP Sack, TCP Fack, Rate based pacing.

I. INTRODUCTION

TCP is a trust worthy means reliable connection-oriented stream protocol [2]. A connection is just like telephone connection which is virtually between computers. So, to maintain virtual connection, we need to maintain current status of the information which means last byte sent. TCP a connection-oriented because of its 3-way hand shake protocol and for every connection state information is maintained [7]. TCP is also a stream based protocol. Even though the underlying Layer (IP) offers an unreliable data delivery but TCP guarantees to deliver data to the other end. It means TCP is responsible to deliver an error-free data. This reliability can be achieved by identifying the sequence number that each byte of data is to be transmitted [13]. Along with it positive acknowledgments (ACKs) is sent back to the data sender. This acknowledgment specifies that the next byte of the data predictable by the receiver. The packets are re-assembled by the TCP receiver and transmit to next layer protocol. It is necessary to send an acknowledgement for every incoming packet [4]. This acknowledgement specifies to the source whether the packet is reached or not. Packets can be traced and also re-transmitted in this manner if required.

II. CONGESTION AVOIDANCE

Whenever so many packets are try to get access the same router's buffer it leads to congestion, which results in the dropping of packets [15] .Both transmission protocols and network routers are necessary to take action to avoid the congestion. Indeed, during a congestion collapse, only a minimal of the existing bandwidth is to be used by traffic that reaches the receiver finally. Congestion is considered as [3], in general, as a disastrous event. However, the congestion itself is linked with different properties, depending upon the features of the original networks, the mechanisms of the communication protocols, the traffic features of the satisfying flows, the level off low dispute, and specification of the functionality in network routers.

Slow Start algorithm is used in the initial phase of the TCP connection. During the Slow Start phase, the network is enforced to drop one or more segments due to overload or congestion [8] [11]. If this occurs, congestion avoidance is used to reduce the transmission rate. However, the Slow Start is combining with Congestion Avoidance as the earnings to get the data transmission going over so it does not slow (goes) down.

In the Congestion Avoidance algorithm [4] the re-transmission timer failing or the response of duplicate ACKs can tacitly signal the sender that the network congestion situation is happening. The sender directly sets its congestion window to one and half of the current congestion window (the minimum of the sender's congestion window size and the receiver advertised window) but using of at-least two segments. If congestion was specified by a timeout, the congestion window is again set to one packet, which inevitably puts the sender's in to the Slow Start state. If congestion was specified by duplicate acknowledgement (ACK), the Fast Retransmit and Fast Recovery algorithms are appealed [8]. As data received during the Congestion Avoidance mode, the congestion window is increased. However, the Slow Start mechanism is only applied to the intermediate point where congestion originally happened. This intermediate point was recorded prior as the new transmission window. After this intermediate point, the congestion window size is incremented by one segment for each of the segments in the transmission window that are to be acknowledged. This approach will force the sender to more slowly produce its transmission rate, as it will approach to the point where congestion has previously been detected [12].

III. FEW VARIANTS OF TCP

A. TCP SACK

The congestion control algorithms performed in our TCP Sack are the conservative extension of the TCP Reno's congestion control mechanism [9], in that it uses the same algorithm for increasing or decreasing the cwnd and make minor changes in the different congestion control mechanisms. Adding the Sack to the TCP does not change the basic principle of congestion control algorithms. The TCP Sack implementation reserves the properties of TCP Tahoe and TCP Reno are processed out-of-ordered packets, and uses re-transmit timeouts as the recovery method of last possibility. The main difference between the TCP Sack implementation and the TCP Reno implementation is that the behavior when multiple segments are dropped from single window of data. As in TCP Reno, the TCP Sack implementation enters into Fast Recovery state when the data sender receives TCP re-transmits duplicate acknowledgments. The sender re-transmits the packet and reduces the congestion window by half.

During Fast Recovery [6], TCP Sack maintains the variable called pipe that represents the predictable number of packets unsettled in the specific path. (It varies from the TCP Reno implementation mechanism). The sender sends only new or re-transmitted data when the assessed number of segments in the route is lesser than the congestion window. The pipe variable is increased by one when the sender either transmits a new segment or re-transmits the old packet. It is decreased by one segment whenever the sender received a duplicate ACK packet with a TCP Sack option reporting that new data is received at the receiver side. Using of the pipe variable decouples the choice of when to send the packet from the decision of which the packets to trannsfer. The sender manages a data structure (DS), the scoreboard that remembers the acknowledgments from previous TCP Sack options. Whenever the sender is able to allow sending a packet, it re-transmits whenever a packet is loosed at the receiver. If no such packets and the receiver advertised window size is satisfactorily large, the sender (source) then sends a new packet. When a re-transmitted packet is itself dropped, the TCP Sack implementation detects the drop with a re-transmit timeout, re-transmitting the dropped segment and then performs slow start. The sender enters in to the Fast Recovery whenever the recovery ACK is established acknowledging all the data that was outstanding.

Problems- The main problem with TCP Sack is that it has the selective acknowledgments that are not providing by the receiver. To appliance TCP Sack we will need to provide selective acknowledgments. It is very difficult task.

B. TCP FACK

Forward Acknowledgments (FACK) also aims at best recovery from the multiple packet losses [3]. The term "Forward ACK" derives from the fact that the procedure keeps track of the properly received data by the largest sequence number. The term "forward ACK" originates from the fact that the procedure keeps track of the appropriately received the data by the largest sequence number [3]. In TCP Fack, TCP maintains two extra variables. TCP Fack, which signifies the forward maximum segments that have been acknowledged through the receiver over the TCP Sack option. Using these two TCP variants, the amount of owing data during recovery can be assessed as forward-most of the data. TCP Fack controls this value (the amount of uncertain data in the network) to be with in the single segment of congestion window (cwnd), and it remains constant in the fast recovery mode. The TCP Fack variable is also used to generate the Fast Retransmits more promptly [14].

C. RATE BASED PACING (RBP)

In the prior work has suggested that the "slow-start restart" problem is the provider to low performance of P-HTTP over the TCP. The solution to the problem is to send the segments at a certain pace until we may get the ACK clock running again [6]. This rate should be based on the fraction of prior estimates of the data transfer rate, since that is the nearby estimate of accessible bandwidth that we have to believes that the modification referred as Rate Based Pacing (RBP). It will give better performance for the situations mentioned in the problem.

RBP Implementation – It requires the below changes to the TCP:
1. Estimation of the bandwidth.
2. Calculate- the window we expect to send in RBP and also the time between the packets in that window.
3. The mechanism that clocks the packets sent in the RBP.

IV. SIMULATION AND RESULTS

The Network simulator NS2 used for simulation. The topology we are using is a wired topology with 10 nodes represented in the figure. Here the nodes are represented as node 0, 1, 2. These are connected through node 3 using a duplex-link and node 3 is linked with node 4 and 5 with duplex-link and node 4 and 5 are also linked with the node 6 via duplex-link, and the node 6 is linked with node 7 and this node is linked with node 8 and 9 with duplex- link. Various TCP agents are attached with node n0 and send the File Transfer Protocol (FTP) data to the matching destination node 8 and node 2 that is in the source node for the node 9 and then sends the packets to the destination node. We discuss the consequences of the simulated scenario of various TCP variants, and distinguish these results. After gaining the results a graph is to be specified and measure the performance of various TCP variants.

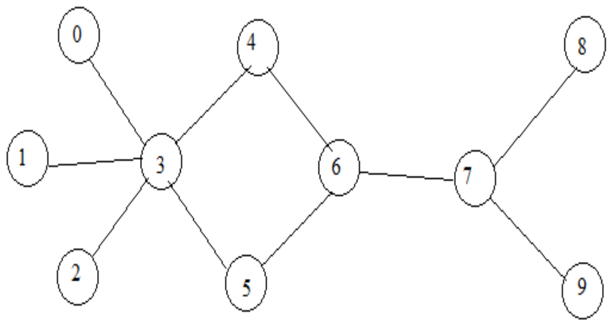

Figure 1: Topology for implementation of various TCP variants.

Figure 1, we use various performance metrics like Throughput, Packet Delivery Ratio, No of packet loss, Average End to End delay and the Jitter applied on TCP variants. Above topology is used to find the performance of various TCP variants. On the source of those parameters which comprised in the practical experiment, the values are generated in the experiment that is shown in table given below. From those data various graphs are to be drawn.

Table 1: Performance metrics of various TCP variants

	TCP Fack	TCP Sack	RBP
Throughput	4003.355	4350.885	4679.154
Packet Losses	50	72	0
Packet Delivery Ratio	98.629	99.761	99.767
End to End Delay	2327.5	2256.773	1469.464
Jitter	1515.61	3566.29	405.1

A. Throughput: Throughput is calculated as the node throughputs of data transmission to the total number of nodes.

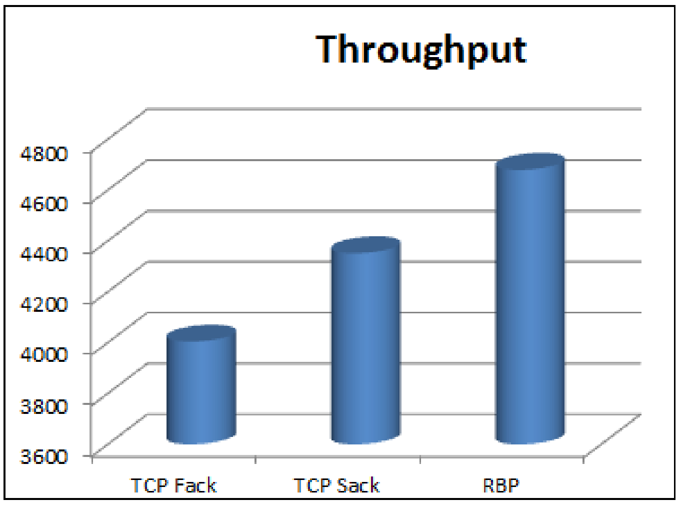

Figure 2: Throughput comparison of TCP Fack, TCP Sack, RBP

Figure2, Shows the throughput in the form of bytes. It shows, RBP gives the better performance in terms of throughput compared to various TCP variants.

B. Number of packet Loss: Packet loss is calculated based on the number of packets sent and packets received. Number of Packet loss= Number of packets sent – Number of packets received..

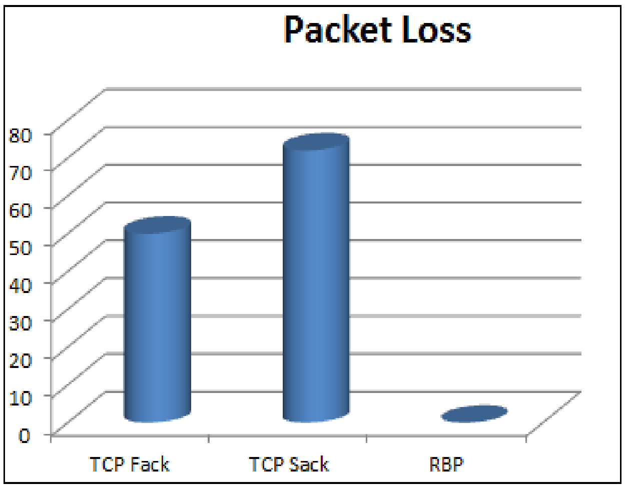

Figure 3: Packet loss comparison of TCP Fack, TCP Sack, RBP

Figure 3, shows the packet loss in the form of bytes. It shows that RBP gives better performance in terms of packet losses compares to TCP Sack and TCP Fack.

C. Packet Delivery Ratio: Packet delivery ratio is performed using the number of packets successfully delivered to the receiver. It is to performed as the total number of packets successfully sent to be number of packets that are generated.

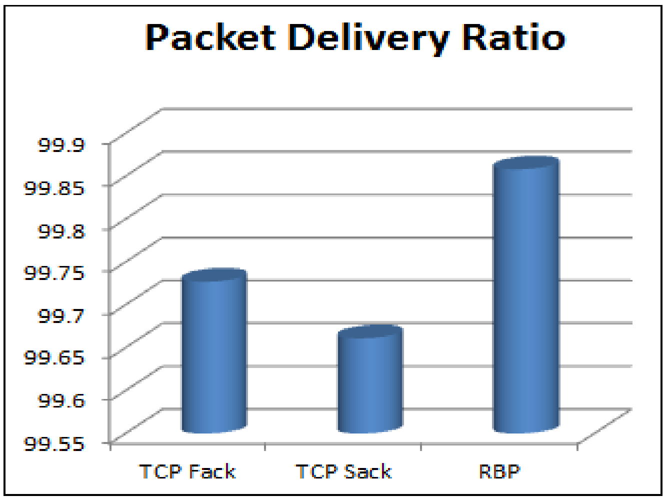

Figure 4: Packet Delivery Ratio comparison of TCP Fack, TCP Sack, RBP

From Figure 4, packet delivery ratio of various TCP variants is represented in the form of bytes. It shows that RBP has better performance in terms of packet delivery ratio than different TCP variants.

D. Jitter: Jitter is a time variation of between the packets leaves from one to another.

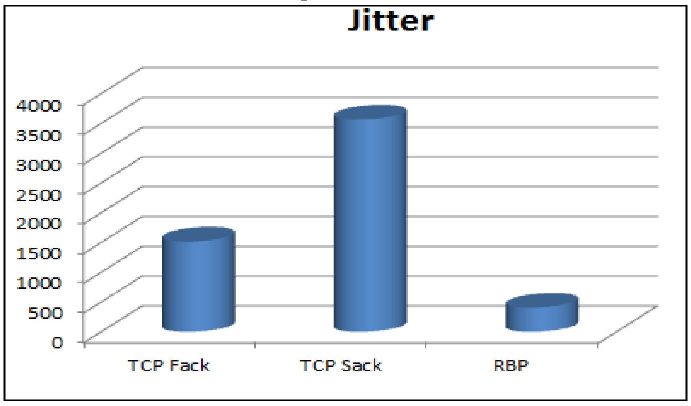

Figure 5: Jitter of TCP Fack, TCP Sack, RBP

From Figure 5, Jitter of various TCP variants are compare with RBP. It shows that RBP has the better performance compares to TCP Fack and TCP Sack.

E. Average Delay: An average end to end delay is calculated using the delay of each packet delivered between the sender and the receiver.

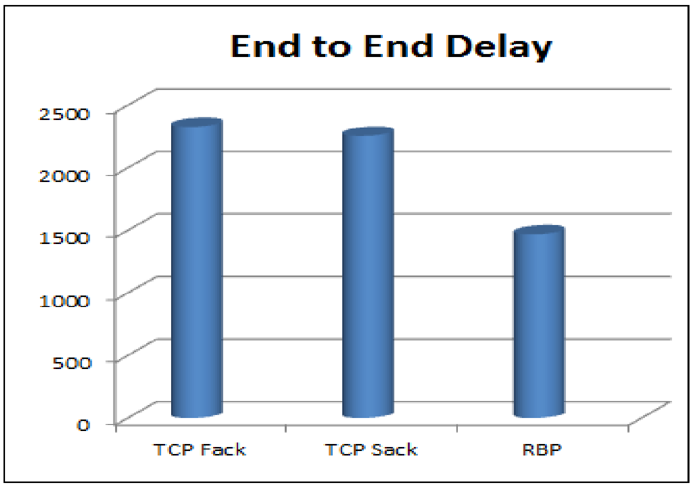

Figure 6: End to End Delay of TCP Fack, TCP Sack, RBP

From figure 6, end to end delay of various TCP variants are compared. From the above graph, RBP has better performance compares to the TCP Fack and TCP Sack.

V. CONCLUSION

In this paper, performance of RBP compares with various TCP variants like TCP Sack and TCP Fack using NS2 simulator using different performance metrics like Throughput, End to end delay, Jitter, Packet delivery Ratio and packet loss. The results shows that the overall performance of various TCP Variants. From the above specified graphs, it shows that RBP has the highest performance and efficiency in the simulation topology. It has been represented in table 1 and also various figures from figure2 to figure 6.

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