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NEWTCP A REFINED NEW MODEL FOR EFFICIENT EXPLICIT CONGESTION REDUCTION IN HIGH TRAFFIC HIGH SPEED NETWORKS THROUGH AUTOMATED RATE CONTROLLING

K. Satyanarayan Reddy[1] & Lokanatha C. Reddy[2]

The conventional TCP suffers from poor performance on high bandwidth delay product links meant for supporting very high data transmission rates to the tune of multi Gigabits per seconds (Gbps). This is mainly due to the fact that during congestion, the TCP's congestion control algorithm reduces the congestion window cwnd to ½ and enters additive increase mode, which can be slow in taking advantage of large amounts of available bandwidth. In this paper an effort has been made to overcome the drawbacks of the TCP protocol through newTCP, a refined new model suggested for controlling the congestion and study its behavior based on various parameters viz., Throughput, Fairness, Scalability, Performance and Bandwidth Utilization for supporting data transmission across the High Speed Networks.

Keywords: Congestion Control, newTCP, High Speed Networks.

1. Introduction

TCP has been the extensively used transport protocol for the Internet for more than two decades. The scale of the Internet and its usage has increased manifolds. The nature of applications has changed significantly. Some of the assumptions made during the early design process of TCP are no longer valid for such applications. And yet, TCP remains the main protocol of the TCP/IP protocol stack based on which the Internet runs. The reason for this importance is that it constantly evolves to keep up with the changing network demands [1], [2], [12].

However as the application needs changed, newer rate control schemes were proposed [2], [3], [4], [6], [8], [9], [10] and [12]. As a result, the Internet operates on a variety of congestion control schemes, though TCP remains the most widely used transport protocol. In [3], [9], [10] the authors have argued that these new congestion control schemes can lead to a new congestion collapse situations and pose the problem of congestion response conformance (wherein selfish/non-behaving sources get an unfavorable share of bandwidth in comparison to TCP).

TCP resides in layer 4 of the 7-layer OSI network model. It provides a connection-oriented, reliable, byte-stream service that is both flow and congestion controlled to the upper layers (application layer), while assuming or expecting little from the lower layers (IP layer and below). This is accomplished by a complicated set of algorithms.

The congestion control functionality of TCP is provided by four main algorithms namely slowstart, congestion avoidance, fast retransmit and fast recovery in conjunction with several different timers. Slowstart uses exponential window increase to quickly bring a newly starting flow to speed. In steady state, the flow mostly uses congestion avoidance in conjunction with fast retransmit/recovery.

2. The Refined New Model for NewTCP and Its Equations

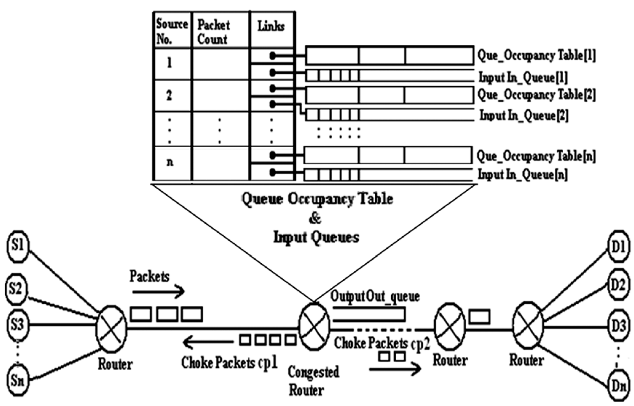

Fig. 1: A New Refined Model for NewTCP [19]

2.1. Model Equations

Where the model given in Figure 1 above is a modified version of model presented in [17], [18], [19] and

a. S1, S2… Sn are sending sources and D1, D2… Dn are the destination nodes.

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b. The choke packet cp1 is meant for informing the sending source about its new sending rate and choke packet cp2 is meant for informing the router (in the direction of Destination node) next to current Congested router to drop packets of misbehaving source indicated in it.

c. Total current In_queue capacity = ΣPacketCount[i] for i = 1, 2… n. (1)

d. Actual capacity of 'n' In_queue's = Σ In_Queue[i] for i = 1, 2… n. (2)

e. from equation (1) and (2) above, the total percentage of In_queue occupancy can be found as follows:

$$\text{Total \% ueue\_occupancy} = \frac{\Sigma\,\text{PacketCount[i]}}{\Sigma\,\text{In\_Queue[i]}} \tag{3}$$

f. Individual percentage of i^{th} In_queue occupancy factor 'β' is given as follows:

$$\beta = \text{percent\_occu} = \frac{\text{PacketCount[i]}}{\text{In\_Queue[i]}} \tag{4}$$

g. Packet Count is meant for containing the count of no. of packets present currently in the In-Queue [i]

for each i = 1, 2, …, n.

The WaitTime for i^{th} source denoted as 'WaitTime^i' is directly proportional to the Round Trip Time (RTT) for i^{th} source denoted as RTT_i i.e.

$$\text{WaitTime}^i \propto RTT_i$$

$$\text{WaitTime}^i = k * RTT_i \tag{5}$$

Where 'k' is the constant of proportionality which is is calculated as follows:

h. The Lee-Time factor 'α1'

$$\alpha 1 = 1 - \beta \tag{6}$$

i. The Extra-Time 'α' is calculated based on the Round Trip Time (RTT) of the i^{th} source RTT_i and is given by

$$\alpha = RTT_i * \alpha 1 \tag{7}$$

j. The WaitTime 'T' is calculated as follows:

$$\text{WaitTime}^i = T = RTT_i + \alpha \tag{8}$$

Putting value of á form equation (7) into equation (8) we get

$$\text{WaitTime}^i = T = RTT_i + RTT_i * \alpha 1$$

$$\text{WaitTime}^i = T = RTT_i * (1 + \alpha 1) \tag{9}$$

So from equation (5) we can infer

$$k = 1 + \alpha 1 \tag{10}$$

The cp1 and cp2 are the choke packets towards downlink (sending source) and uplink (receiving node) respectively and cp1 and cp2 have following formats:

cp1 Format:

Table 1

SourceIP Address	DestinationIP Address	New Sending Rate	Extra Time "α"	Congested Router's Time Stamp

cp1: Indicates to the sending source (with SourceIP Address) that current rate of data transmission is to be set to New Sending Rate i.e. currentRate ⟵ New Sending Rate.

Also the sending source is informed about the extraTime (besides Sending Source's Time & Congested Router's Time Stamp) it has.

cp2 Format:

Table 2

SourceIP Address	DestinationIP Address	DropFlag = 1

cp2: This is an indication to the router (in the direction of Destination node) next to current Congested router that if DropFlag = 1 then all the remaining packets (if any) from the misbehaving source with "SourceIP Address" and meant for "DestinationIP Address" of cp2 should be dropped from its In_Queue and the links for Que_OccupacyTable and In_Queue be freed.

2.2. Equations for Bandwidth Management

We propose to manage the network bandwidth using the Dynamic Programming Algorithm [14], [19] assuming that Network bandwidth is to be allocated amongst 'n' number of hosts, which are willing to connect (to communicate with the other nodes) to the network.

Let B1, B2, …, Bn be the bandwidth requirements of the 'n' hosts respectively and let the total throughput function "T" (The Objective Function) be expressed as sum of the individual throughputs of the 'n' hosts as follows:

$$\text{Maximize } T(B_1, B_2, \ldots, B_n) = t_1 \times B_1 + t_2 \times B_2 + \ldots + t_n \times B; \tag{11}$$

The constraint on bandwidth can be defined as: $B_1 + B_2 + \ldots + B_n \leq B$; (12)

where $B_i \geq 0$ for i = 1,2,…,n; and B is the total link Bandwidth and

Case i. $t_1 + t_2 + \ldots + t_n \geq 0.85$; (13)

Where "t_i" is the Throughput of the link 'i'

Case ii. $t_1 + t_2 + ... + t_n \geq 0.95$; (14)

Where "t_i" is the Throughput of the link 'i' $t_i \leq B_i$; for all i = 1, 2, 3, ….., n.

We define the free or un-allocated bandwidth given by the formula:

$$F(B) = B - (B_1 + B_2 + \ldots + B_n); \quad (15)$$

We define TAB: Total Available Bandwidth is given by the formula

$$TAB = F(B) + \Sigma B_k; \quad (16)$$

for all k = 1, 2, …., m; where B_k is the bandwidth that became available when k^{th} misbehaving connection was dropped; and

Let BRJ: Bandwidth requested by j^{th} node during scale-up is given by: = B_j bandwidth requirement of new link 'j'.

We now define the Total Available Bandwidth (TAB_S) after Scaling (i.e. new connections request for Bandwidth) as follows:

$$TAB_S = TAB - \Sigma B_j; \quad (17)$$

for all j; where j is the node requesting Bandwidth during scale-up.

3. THE DESIGN ISSUES

We have used the simulation environment NS 2.28 [11] supporting Highspeed data rate for transmitting the data across the source nodes S_i and the destination node D_i for i = 1, 2, …., n through intermediate bottleneck routers.

3.1. The Model for Simulating NewTCP

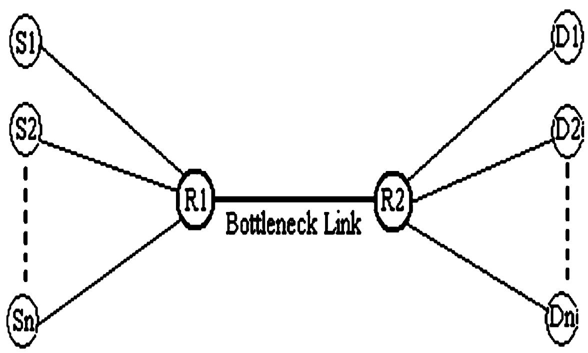


Fig. 2: Model for Simulating NewTCP

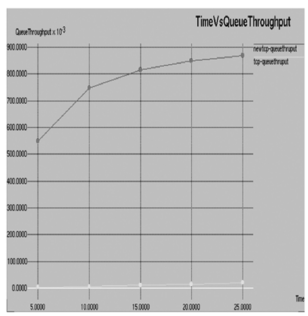
Where S1, S2,…., Sn are the sending sources, of which some of which may be behaving sources and some may be misbehaving sources. The link between the routers R1 and R2 is called the bottleneck link. D1, D2, ….., Dn are the destination nodes. The full algorithm for simulating newTCP against the conventional TCP has been taken from [18], [19].

4. RESULTS AND INTERPRETATIONS

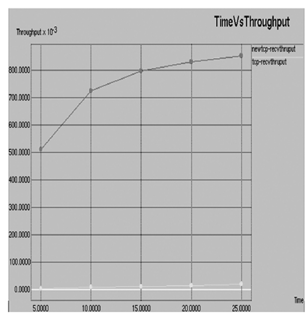
4.1. The Throughput

Case 1: In this case, two transmitting sources have been considered, when both the sending sources are behaving sources. In slide1 and slide2 (as shown below) the plots are indicating Time Vs. QueueThroughput and Time Vs. Throughput respectively.

The slide 1 shown below, clearly indicates the throughput measured at the queue level for the TCP and the newTCP, it is observed that TCP's throughput suffers at queue level i.e. after a duration of 25 seconds of simulation the queue throughput remains to 15% to 20% which is quite low, whereas the throughput of newTCP starts with a throughput of 55% after a simulation run of 5 seconds and it gradually increases to 88% after a simulation run of 25 seconds. Thus the newTCP algorithm is achieving the expected results at high transmission rates on High Traffic High Speed Networks.



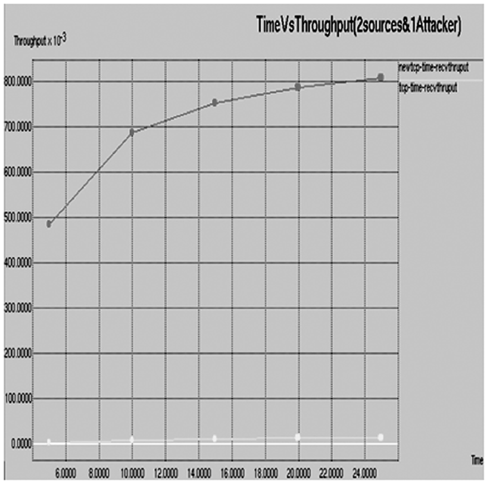
Slide 1: Queue Throughput



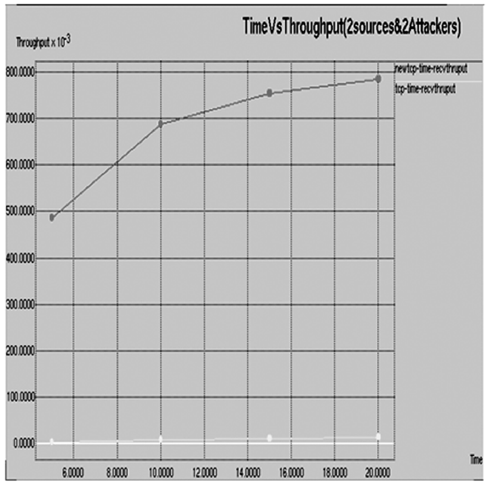
Slide 2: Throughput at the Destination Node

The slide 2 shown above, clearly indicates the throughput measured at the destination node for the TCP and the newTCP, it is observed that TCP's throughput suffers at destination node level i.e. after a duration of 25 seconds of simulation the throughput remains to 10% to 15% which is quite low, whereas the throughput of newTCP starts with a throughput of 52% after a simulation run of 5 seconds and it gradually increases to 85% after a simulation run of 25 seconds. Thus the newTCP algorithm is achieving the expected results at high transmission rates on High Traffic High Speed Networks.

Case 2: In this case, three transmitting sources have been considered, in which two of the sending sources are behaving sources and one is misbehaving source (Attacker).



Slide 3: Throughput at the Destination Node



Slide 4: Throughput at the Destination Node

The slide 3 shown above, clearly indicates the throughput measured at the receiving source (destination node) for the TCP and the newTCP, it is observed that the TCP's throughput suffers at the destination node level i.e. after a duration of 24 seconds of simulation the throughput remains to 2% to 3% which is quite low, whereas the throughput of newTCP starts with a throughput of 48% after a simulation run of 5 seconds and it gradually increases to 82% after a simulation run of 24 seconds. Thus the newTCP algorithm is achieving the expected results at high transmission rates on High Traffic High Speed Networks.

Case 3: In this case, four transmitting sources have been considered, in which two of the sending sources are behaving sources and two are misbehaving source (Attacker).

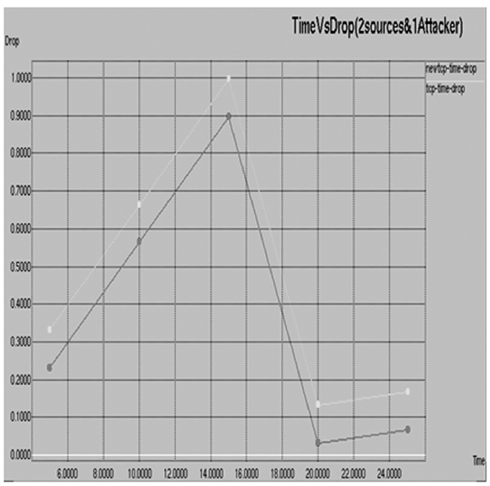
The slide 4 shown above, clearly indicates the throughput measured at the destination node for the TCP and the newTCP, we observe that TCP's throughput suffers at the destination node level i.e. after a duration of 24 seconds of simulation the throughput remains to 2% to 3% which is quite low, whereas the throughput of newTCP starts with a throughput of 48% after a simulation run of 5 seconds and it gradually increases to 78% after a simulation run of 20 seconds.

Thus the newTCP algorithm is achieving the expected results at high transmission rates on High Traffic High Speed Networks.

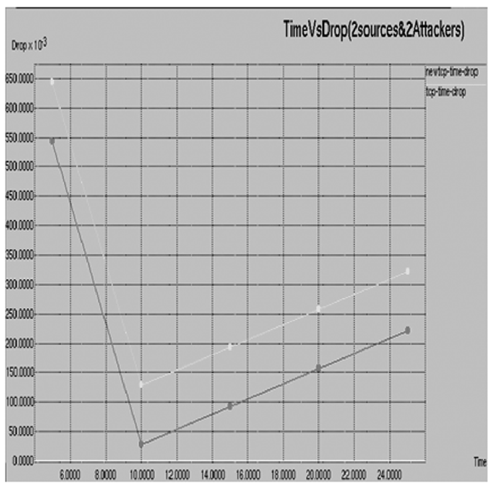
4.2. The Fairness of NewTCP Algorithm

Case 4: In this case, three transmitting sources have been considered, in which two of the sending sources are behaving sources and one is misbehaving source (Attacker).

The slide 5 shown below, clearly indicates the packet drops measured at the input In_Queue for the TCP and the newTCP, it is observed that TCP's throughput suffers more packet losses at the input In_Queue level when compared to the newTCP i.e. after a duration of 24 seconds of simulation the packet drop reduces considerably for newTCP when compared to the TCP is quite low. Thus newTCP exhibits fair behavior when compared to the TCP flow (as shown below in slide 5).



Slide 5: Packet drops for TCP and newTCP

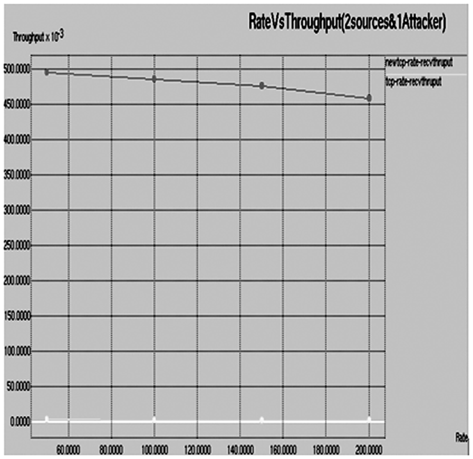


Slide 6: Packet drops for TCP and newTCP

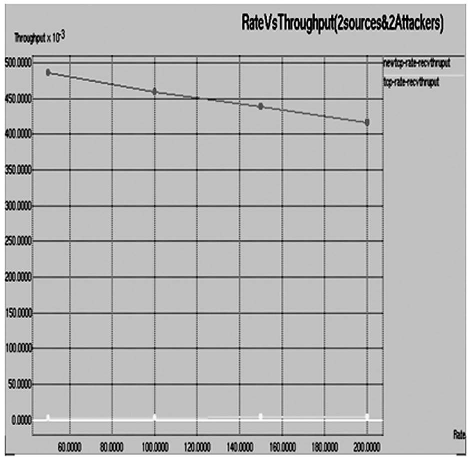
Case 5: In this case, four transmitting sources have been considered, in which two of the sending sources are behaving sources and two are misbehaving source (Attacker).

The slide 6 shown above, clearly indicates the packet drops measured at the input In_Queue for the TCP and the newTCP, it is observed that TCP's throughput suffers more packet losses at the input In_Queue level when compared to the newTCP i.e. after a duration of 24 seconds of simulation the packet drop reduces considerably for newTCP when compared to the TCP is quite low. Thus newTCP exhibits fair behavior when compared to the TCP flow (as shown below in slide 6).

Case 6: In this case, three transmitting sources have been considered, in which two of the sending sources are behaving sources and one is misbehaving source (Attacker).



Slide 7: Reduction in Throughputs for TCP and NewTCP



Slide 8: Reduction in Throughputs for TCP and NewTCP

The slide 7 shown above, clearly indicates the packet drops at the input In_Queue for the TCP and the newTCP, it is observed that TCP's throughput suffers drastically when compared to the newTCP i.e. after a certain duration of simulation when the packet drops, the throughput of TCP suffers drastically whereas for newTCP reduction in throughput is considerably quite low (from 49% to 46%) and achieves data rate of 200 mbps soon. Thus newTCP exhibits fair behavior when compared to the TCP flow (as shown in slide 7 above).

Case 7: In this case, four transmitting sources have been considered, in which two of the sending sources are behaving sources and two are misbehaving source (Attacker).

The slide 8 shown above, clearly indicates the packet drops at the input In_Queue for the TCP and the newTCP, it is observed that TCP's throughput suffers drastically when compared to the newTCP i.e. after a certain duration of simulation when the packet drops, the throughput of TCP suffers drastically whereas for newTCP reduction in throughput is considerably quite low (from 48% to 43%) and achieves data rate of 200 mbps soon. Thus newTCP exhibits fair behavior when compared to the TCP flow (as shown in slide 8 above).

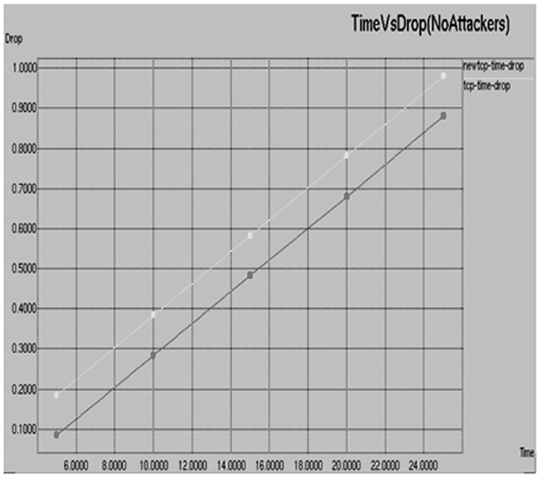
4.3. The Bandwidth Utilization

From the slide1, slide2, slide3 and slide4 as shown above it is observed that in case of High Speed Networks supporting high data transmission rates, the bandwidth utilization for newTCP is considerably very high (on an average, utilization of above 80%) when compared to that of the TCP (an average of around 2% to 3%) in both the cases when there are no misbehaving sources also when there are misbehaving sources.

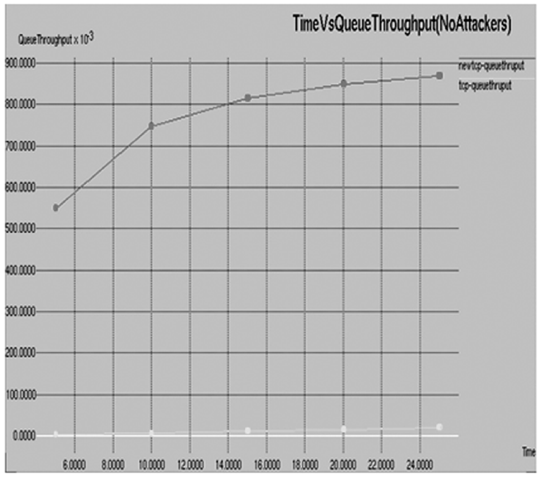
4.4. The Performance

Case 8: When all the sources are behaving & are transmitting data at a rate of 200 mbps to 300mbps and the number of transmitting sources are 2, 4, 6 and 8 respectively then the packet drops observed in case of conventional TCP is considerably very large when compared to the packet drops in case of newTCP for a simulation duration of 24 seconds is as shown below in Slide 9.

Case 9: When all the sources are behaving & are transmitting data at a rate of 200 mbps to 300mbps and the number of transmitting sources are 2, 4, 6 and 8 respectively then the Queue throughput at input In_Que that is observed in case of conventional TCP is considerably very low when compared to the Queue throughput of newTCP (88%), for a simulation duration of 24 seconds is as shown below in Slide 10.

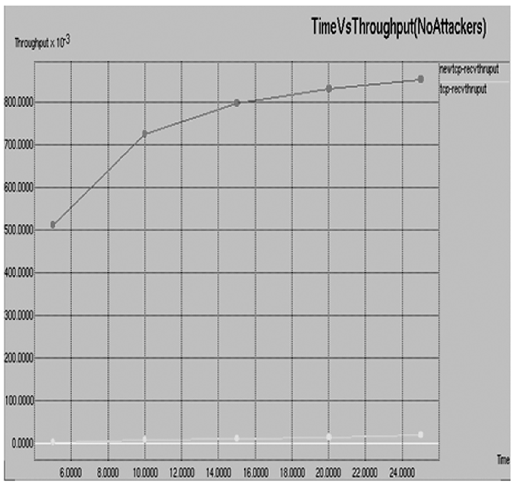


Slide 9: Observed Packet Drops for TCP and NewTCP

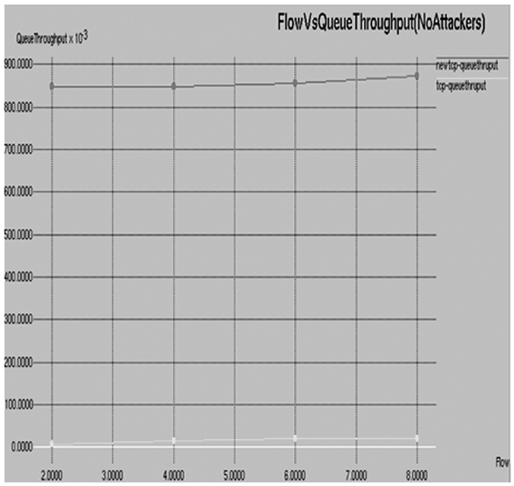


Slide 10: Observed Queue Throughput for TCP and NewTCP

Case 10: When all the sources are behaving & are transmitting data at a rate of 200 mbps to 300mbps and the number of transmitting sources are 2, 4, 6 and 8 respectively then the throughput at the destination nodes that is observed in case of newTCP is considerably very high (85%) when compared to the throughput of conventional TCP, for a simulation duration of 24 seconds is as shown below in Slide 11.



Slide 11: Observed QueueThroughput at In_Que for TCP and NewTCP



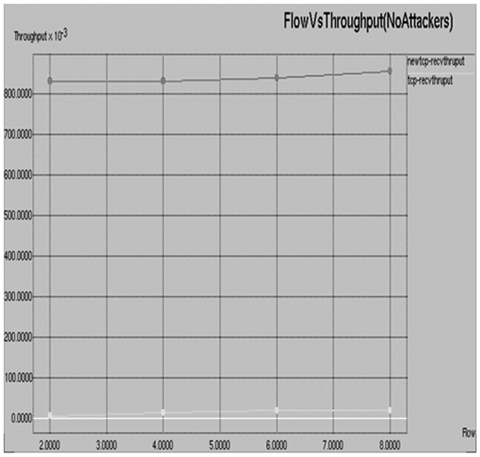
Slide 12: Observed QueueThroughput for Various Flows at In_Que for TCP and NewTCP

Case 11: When all the sources are behaving & are transmitting data at a rate of 200 mbps to 300mbps and the number of transmitting sources are 2, 3, 4, 5, 6, 7 and 8 respectively. The QueueThroughput at the input In_Que that is observed in case of newTCP is progressively increasing from 85% to 88% when compared to the throughput of conventional TCP (which is very low) thus indicating the Scalability of the newTCP algorithm, for a simulation duration of 24 seconds is as shown above in Slide 12.

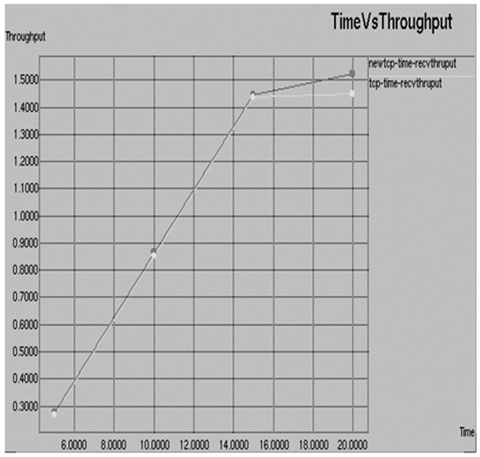
Case 12: When all the sources are behaving & are transmitting data at a rate of 200 mbps to 300mbps and the number of transmitting sources are 2, 3, 4, 5, 6, 7 and 8 respectively then the Throughput at the destination nodes that is observed in case of newTCP is progressively increasing from 85% to 88% when compared to the throughput of conventional TCP thus indicating the Scalability of the newTCP algorithm, for a simulation duration of 24 seconds is as shown below in Slide 13.

Case 13: When the flows comprise combination of both TCP as well as newTCP. The slide 14 as depicted above shows the plot of throughput of both the flows against time.

From the above slide it is evident that in presence of both the flows viz. TCP and newTCP, the throughput is almost same till first 15 seconds and then there is marked difference in the throughputs of the flows leading to lowering of throughput of TCP flows.



Slide 13: Observed Throughput for Various Flows at Destination nodes for TCP and newTCP



Slide 14: Observed Throughput for Both flows against Time for TCP and newTCP

5. CONCLUSION

The newTCP [19] a Refined New Model, achieves fairness through the fact that the sources which are sending packets indiscriminately are penalized with drastic cut in their transmission rates (maximum to ½ the current rate of transmission, like TCP) and behaving sources may have to reduce their sending rates to a low or moderate levels (but not all the sources are required to reduce their current rate of transmission to ½).

Thus the proposed model newTCP based on [8], [9], [10], [17], [18] and [19] is:

a. Utilizing the Bandwidth optimally using Dynamic Programming concept of Operation Research and it makes the bandwidth available to the Behaving sources under Congestion situation and also when there is No Congestion.

b. Maximizing the Throughput for the Behaving sources under Congestion situation and also when there is No Congestion.

c. Meeting the QoS demands of the Network Traffic during Congestion situation and also when there is No Congestion.

d. Dropping all the packets from the Non-behaving sources during the congestion, and packets from the behaving sources are accepted and accommodated in the queue for onwards transmission.

e. Allowing the scaling up i.e. allocating Bandwidth to new host which agrees to behave by sending packets as per QoS agreement.

f. Achieving Fairness for the Behaving sources.

Thus overall performance of newTCP when all the transmitting sources are behaving is reasonably good when compared to the conventional TCP.

6. ACKNOWLEDGEMENTS

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