A Fluid-based Simulation Study: The Effect of Loss Synchronization on Sizing Buffers over 10Gbps High Speed Networks

Suman Kumar, Mohammed Azad, Seung-Jong Park
Computer Science Department and Center for Computation and Technology
Louisiana State University
Baton Rouge, LA, USA
{sumank, sjpark}@csc.lsu.edu

***Abstract*—Router buffer size has been playing an important role in packet based networks. In particular, it has been argued that high speed optical networks of the order of 10Gbps demand for large buffer size according to rule of thumb. Because of inherent limitations (i.e., execution time and memory overhead) of a packet based simulator for high speed optical networks, there are few simulation studies available. Also, high speed network experimentation facilities (e.g., testbeds and real networks) are still scarce resources and therefore, a complete experimental study on this topic is not easy to perform. In this paper, a fluid-based model is introduced to accommodate the effect of packet loss synchronization on sizing intermediate buffers over 10Gbps optical networks. Our simulation approach is simple and easy to implement and above all provides some insight of the relation between synchronization level, buffer size and performance of different TCPs. Simulation results are presented to show the effect of loss synchronization/desynchronization on link utilization of popular transport protocols using different buffer sizes on 10Gbps optical networks. The proposed fluid-based method will promote principal understanding of the future high speed networks and accelerate protocol development process.**

I. Introduction

Backbone high-speed optical network is characterized with high capacity and low delay providing high-capacity telecommunication network based on fiber optics technologies. Increase in backbone network bandwidth is driven by growth in a wide range of applications such as scientific data transfer, distributed computing etc. At the heart of high speed backbone networks are the core IP routers with bandwidth of the order of 10Gbps. Sizing buffer of these high speed routers is an important issue. Rule-of-thumb suggests that a single TCP flow requires a buffer size which is equal to the bandwidth-delay product in order to prevent the link from going idle [1]. However, for a high bandwidth and long delay network, it is not feasible to have such large size of intermediate buffers due to the cost of memory, complexity of buffer management, and large queuing delay, etc.

Recent research [2] suggested that almost 100% link utilization can be achieved with the intermediate router buffer size of the order of $BDP/\sqrt{N}$, where BDP is the bandwidth delay product and N is the number of long lived TCP flows, as long as links carry a large number of TCP flows. Many long lived TCP flows are statistically multiplexed to provide high link utilization. In the same way several other studies show that buffer sizes lower than the rule-of-thumb can provide good link utilization if flows are de-synchronized [3], [4]. Previous works assume when there are large number of TCP flows which have a behavior of AIMD (additive increase and multiplicative decrease). However, most of recent applications and demonstrations launched at high speed 10Gbps optical networks have used a small number of flows of high speed TCP variants [5], [6] which greedily occupy high speed links.

It is observed that high speed TCP flows are aggressive and higher burstiness is attributed to high congestion window values [7]. In previous work, larger burstiness increase the probability of packet losses [8] and high-speed TCPs show some level of drop synchronization. We observe that de-synchronization of packet losses among high-speed TCP flows plays an important role in the design of buffer sizes to provide good link utilization. Although it is known that small number of high speed TCP flows also synchronized at different levels, to the best of our knowledge, no study is there to explore the role of different levels of drop synchronization on TCP performance for a given buffer size over 10Gbps networks. Therefore, our goal in this paper is to understand the impact of loss synchronization on sizing buffers and the counter effect that these two has on the performance of 10Gbps high speed networks with few number of TCP flows.

A. *Limitations of Current Work*

While the theory of stochastic multiplexing has its own limitations, [9] and [8] are two of the work, that are focused on high speed network on this topic. The first one presents an analytical model using M/M/1/K queuing model approximations, that is only valid for HSTCP. It is well known that queuing models has its own limitations, especially in heavy tail distributions. The second work is interested in finding the synchronization statistics in a high speed network environment via $NS2$ simulation. This work does not answer the question: How does loss-synchronization level affect the high speed TCP performance? or Is it same for all the environments? Also, both of these works do not address 10Gbps high speed network. While 10Gbps network is in deployment phase, it is very important for network managers/service providers to understand this relation for optimal and efficient network deployment. In this paper, we have a novel approach to analyze the impact of buffer size on the performance of popular high speed TCP variants in a 10Gbps network environments for different loss synchronization levels. Our result disclose some interesting behavior of high speed TCP variants and therefore, promote the understanding and possible development of ongoing protocol development work for high speed networks.

B. *Contribution and Paper Organization*

1) **A Loss-Synchronization Module for Fluid Simulation**

While network simulation is well accepted and widely used method for performance evaluation, it is also well known that packet-based simulators like NS2 and Opnet cannot be used in the case of high-speed network ($\tilde{1}$0Gbps) because of its inherent bottlenecks in terms of message overhead and cpu execution time [10]. Therefore, the model based simulation approach with the help of a set of coupled differential equations is preferred and proved to give satisfactory results [11]. In this paper, we show that normal fluid models do not capture the loss de-synchronization phenomena in the simulation. Through this paper, we propose a user

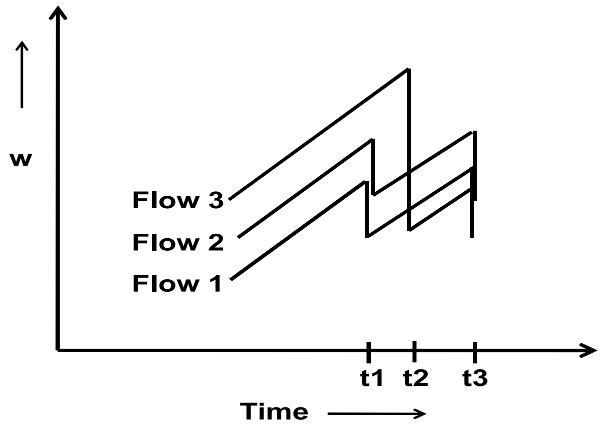


Fig. 1. Synchronized and De-Synchronized Losses on TCP flows

configurable loss synchronization module that can be attached to the bottleneck queue to break the all-flow loss synchronization in fluid simulation. Presented loss-synchronization module can be easily implemented the to study the performance of high speed networks. In previous research, queuing theory is used to obtain the load and packet loss rates which often requires complicated mathematics and are not in the same line of fluid model simulation approach. Whereas this paper's approach can be generalized to both drop-tail and AQM buffers. Therefore, the presented module can be highly useful for testing and further the development process of future protocol for high speed networks.

2) **Performance results of High Speed TCP variants for Different Buffer Sizes on 10Gbps Networks**
With the help of synchronization module, we perform an extensive study the impact of buffer sizes on the performance high speed TCP variants on 10Gbps networks for different levels of loss synchronization. In this paper, our performance metric is link-utilization. The presented study provides a reference point for the effect of loss synchronization on the performance of high speed protocols. This work further motivates the exploration of the relationship among synchronization behavior of high speed flows, buffer sizes and congestion control on high speed networks of the order of 10Gbps and beyond.

This paper is organized as follows: in section II, we introduce the concept of loss synchronization, and discuss the limitation of current fluid model simulation approaches. In the same section, we propose a loss-synchronization module for the fluid simulation and present the simulation set-up for high speed network simulation. Section III presents some basic simulation results using this model. Section IV gives a brief summary of research work on this topic. Section V summarizes our work and present conclusion and possible future research direction in this area.

II. Fluid Model Implementation for Loss-Synchronization

A. Loss Synchronization

In Figure 1, there are three flows numbered 1, 2, and 3. Flow 1 and 2 records packet loss at time t1 and flows 1, 2 and 3 records packet loss at t3. Often packet drops are not uniformly distributed among all the flows because different TCP flows have different sending rates and those packets arrive at the bottleneck link randomly rather uniformly.

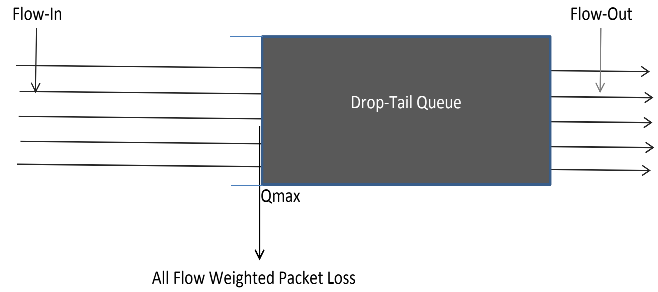


Fig. 2. Operation of a Drop-tail Queue under the Fluid Model Simulation

High speed TCP flows dramatically react to packet drops by reducing their congestion window sizes for each dropped packet. When a large number of flows experience packet drops around the same time, they synchronously react by reducing their congestion window sizes. This may lead to a significant reduction in instantaneous throughput of the system. Whereas if only a few flows record packet losses, rest of the flows keep the link busy by increasing their sending rates. Therefore, the change in aggregate congestion window is less significant in the case of lower level of synchronization whereas it is more when the synchronization level is high. Therefore, level of synchronization can eventually affect the performance of high speed flows and as we point out in this paper, synchronization level impact the required buffer size high speed networks.

B. A General Fluid Model and Its Limitations

For reference, we re-write the original fluid model equations as below:

F_i = A set of ordered queues traversed by i^{th} flow
W_i = Congestion Window for i^{th} flow
R_i = Round Trip Time for i^{th} flow
λ_i = Loss Indication Rate for i^{th} flow
q_l = Queue Size associated to l^{th} link
C_l = Service Capacity/Bandwidth for l^{th} link
p_l = Packet Drop Probability at l^{th} queue
q_l^{max} = Maximum queue size associated to l^{th} link
n_l = Denotes number of flows traversing l^{th} link
A_l^i=Arrival Rate of i^{th} flow at l^{th} link

In fluid model [12], [13], packet chunks are modeled as a fluid at a sender according to the following ordinary differential equations:

$$\frac{dW_i(t)}{dt} = \frac{a(t)}{R_i(t)} - W_i(t)b(t)\lambda_i(t) \tag{1}$$

When the packet fluid reaches the queue, the queue checks for the incoming rate and adjust the queue size according to equation below:
Queue Size:

$$\frac{q_l(t)}{dt} = -1(q_l(t) > 0)C_l + \sum_{i=1}^{n_l} A_l^i(t) \tag{2}$$

($q_l(t) > 0$ and can have only positive value. where $ARsum_l$ as sum of the arrival rates of all flows at queue l, where l is bottleneck queue in our case.

$$ARsum_l = \sum_{i=1}^{n_l} A_l^i(t) \tag{3}$$

Under the overload or over filling, the drop-tail queue generates the loss probability according to the equation below:

$$p_l(t) = \begin{cases} 0, & q_l(t) < q_l^{max} \\ max(\frac{ARsum_l - C_l}{ARsum_l}, 0), & q_l(t) = q_l^{max} \end{cases} \tag{4}$$

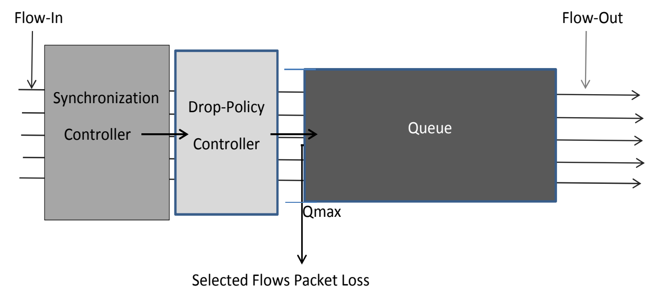


Fig. 3. Operation of the Loss-Synchronization Module on a Queue under Fluid Model Simulation

This loss probability is proportionally divided among all flows passing the queue as shown in Figure 2.

$$\lambda_i(t) = \sum_{l \in F_i} A_l^i(t) p_l(t) \tag{5}$$

In reality, TCP flows's burstiness induces a certain degree of de-synchronization among the dynamics of TCP flows sharing the same bottleneck buffer. In addition, during the time of congestion, losses are not evenly distributed among TCP flows and TCP flows with larger congestion windows are more likely to get affected. The model does not account for this behavior and therefore, it is not very useful in buffer size and performance studies in the presence of de-synchronized flows.

C. A Loss-Synchronization Fluid-Simulation Module

Our loss-synchronization simulation module consists of two parts as illustrated in figure 3.

1) Loss-Synchronization Controller: The controller controls the loss synchronization factor at the time of congestion. The loss synchronization factor can be user given or derived from a distribution or experimental data at any congestion event. The loss synchronization factor is an integer value and defines how many flows are to record packet losses. For the description below, loss synchronization factor is denoted as m_k at the k^{th} congestion. Suppose, there are N number of flows in the network and out of those N, m_k flows are experiencing packet losses. Therefore, following boundary value holds for any selected m_k

$$m_k \in (1, N) \tag{6}$$

2)Packet-Drop Policy Controller: Loss-synchronization controller pass the information to the packet-drop policy controller. The packet-drop policy controller selects m_k TCP flows to drop at the time of congestion by using a priority matrix that defines the order of flows to record packet losses and pass this information to the queue. Specifically, at the time of k^{th} congestion, the packet-drop policy controller determines the priority matrix $P^k = \left[D_1^k, D_2^k ...., D_N^k\right]$, where D_i^k defines some important time varying decision parameter for flow i at k^{th} congestion events. $D_i^k > D_j^k$ indicates that packets in flow i has higher drop probability than flow j. We define Pl_k as the set of m flows selected based on the priority matrix. We define Pl a policy through which these m flows get selected. Therefore, all the flows $i \in Pl$ satisfy following relationship:

$$\sum_{i \in Pl_k} \lambda_i(t) = ARsum_l - C \tag{7}$$

The above equation means that every loss is accounted and distributed among the flows (since burstiness is random and stochastic phenomena of TCP flows). In both the models above, we assume that congestion occurs when the buffer can not accept any more packets and total arrival rate exceeds the link capacity. Therefore, there is a duality between both the models in terms of loss rates at the queue.

D. High-Speed Network Simulation Set-up

1) **Loss Synchronization Module Set-up**

For high speed networks we make following assumptions:

(i) Congestion events occur when bottleneck buffer is full.

(ii) Highest rate flows are more prone to record packet losses.

(iii) High Speed TCP flows's burstiness induces higher level of synchronization.

The first assumption is obvious and states that buffer overflow causes congestion in the network. Second assumption relates to the fact that high speed TCP flows are aggressive and higher burstiness is attributed to high congestion window values, the assumption that highest sending rate flows have higher probability to record losses is heuristically justified(in [7], it is observed that larger burstiness increases the probability of packet losses) if not very accurate. To understand the third assumption, we refer to [8] that shows high speed TCP intent to show some level of drop synchronization. Fairness in packet drops (i.e. dropped packets are uniformly distributed among all the flows) can create synchronization among flows whereas unfair packet drops (i.e. only some of the flows record packet drops) can lead to reduced synchronization. Below, we outline the values for the loss synchronization simulation module for queue.

To select random m_k at any congestion event k, we define a parameter X which is defined to give the minimum level of synchronization (i.e. the ratio of synchronized flows (Flows experiencing packet losses) and total number of flows is no less than X):

$$m_k \in (XN, N) \tag{8}$$

It is observed that when congestion happens multiple number of flows record packet losses. Selection of X not only satisfies a least certain level of drop synchronization but also it does not avoid any degree of synchronization higher than X. Hence, the definition of X is reasonable for high-speed network case. It is to be noted that, the definition of X may not be suitable to all levels of statistical multiplexing for high speed TCPs but it presents a very simple reference point covering a wide range of synchronization behavior of high speed TCP flows.

In our study Pl_k is the set of m_k flows with highest sending rates that is denoted by priority matrix whose elements are arrival rate of flow i (=D_i^k) at the bottleneck queue. Intuitively, our fluid simulation model appears to be analogous to $M/D/1$ queuing system on a short simulation time scale and it will converge to that of $M/M/1$ system.

2) **Performance Matrix**

Our performance matrix is %link utilization denoted as U. To calculate link utilization, we sample the departure rate at the bottleneck link as defined below:

$$U = \frac{\sum_s \sum_{i=1}^{n_l} Dep_l^i(t_s)}{C_l \times \sum s} \times 100 \tag{9}$$

In the above equation, Dep_l^i is the departure rate of flow i at bottleneck link l, s denotes the sampling instances, and C_l is the capacity of the bottleneck link. Therefore, to present our result we calculate the link utilization by taking the average of sampled total departure rate at bottleneck link.

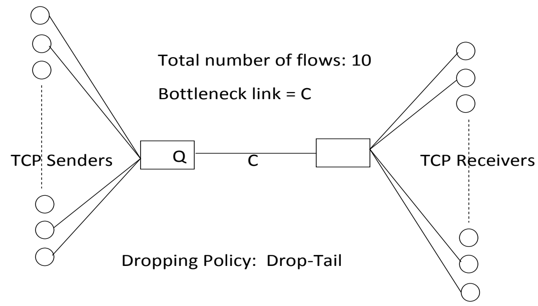


Fig. 4. Simulation topology with 10 nodes sharing a bottleneck

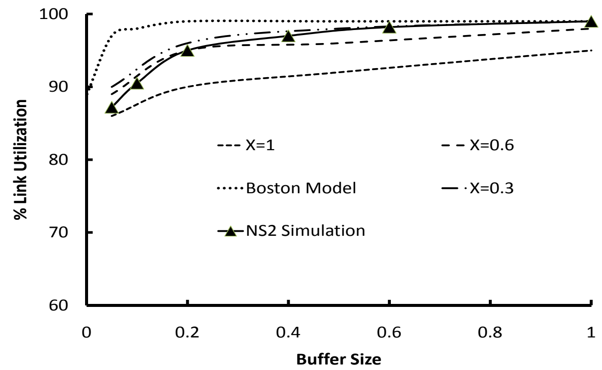


Fig. 5. Validation with previously published results(Average Link Utilization as function of Buffer size fraction (max 12500))

III. Simulation Results

Since our work is related to the development of future high-speed networks with competing high speed TCP flows, we consider 10 flows of high-speed TCP variants on 10Gbps link competing for bandwidth. Network in general consists of several queues. The behavior of TCP congestion control algorithm mainly depends on the congestion on the most congested link [2]. Therefore, in this work we consider the case of 10 persistent high speed TCP flows sharing the same bottleneck link on a dumbbell topology. The queuing discipline policy is drop-tail with loss-synchronization module. Mean field theory of buffer sizes [14] suggests that at least 60% flows records packet losses during a congestion event. Since high speed network is different than internet, we consider two cases of X, X=0.3 and X=0.6 to introduce lower and higher level of synchronization among high speed connections (X=1 refers to the case when all flows record packet losses). Values for configurable parameter m is drawn from normal distribution. We consider the effect of only the basic congestion control algorithms on bottleneck link and therefore, the mechanisms like time-out, slow start, and fast retransmit are not simulated. We perform fluid simulation by solving differential equations for high speed TCP variants. Since our approach involves random number generation, we ran the simulation 10 times each for 3000s and averaged them for each result presented in this section. It is to be noted that throughput results presented in this section is taken from the time of first congestion event (i.e. when the flows are stabilized, we record the first congestion since the start of simulation)

Before presenting our results, we verify our model with previously published results in [9]. Topology presented in figure 4 is used for all the results presented in this section.

A. Model Verification

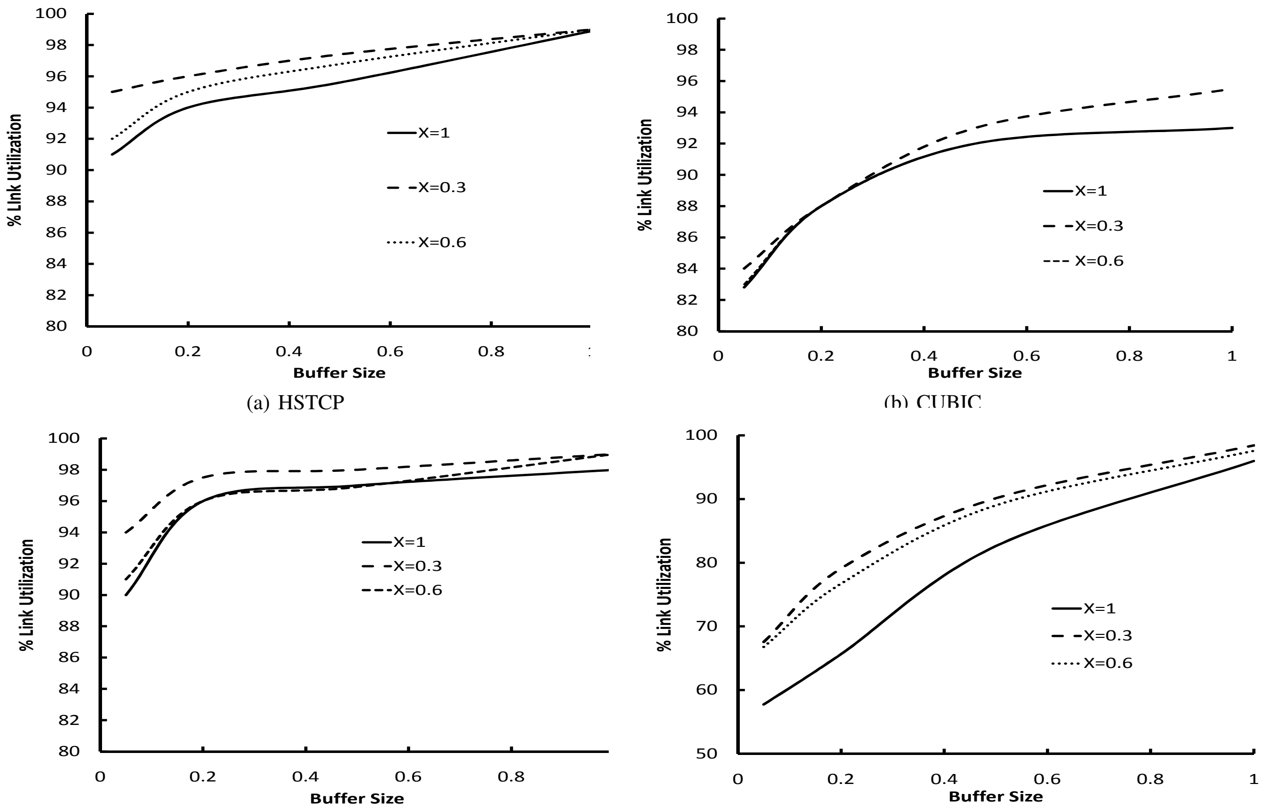
To verify our proposed model compared with the Boston model presented in [9], we used the NS2 simulator. For this validation, bottleneck link C is set to 1Gbps; packet size is 1KB; and RTT is ranging from 115.5ms to 124.5ms with average RTT of 120ms. The buffer size of the bottleneck buffer is varied as a fraction of 12500 packets corresponding to the BDP of largest RTT (= 124.5ms). We use the same parameter sets as suggested in [5]. Figure 5 shows that the Boston model has a large amount of difference compared with NS2 simulation results in case of low and moderate buffer sizes. However, the proposed model's results shown in figure 5 give a closer match with NS2 simulation results. We also observe that when all the flows are synchronized (X=1), fluid model does not match with the NS2 simulation result. As we decrease the synchronization (X=0.6 and X=0.3 case), the utilization improves and NS2 simulation result matches with fluid simulation results. We conclude that fluid simulation with synchronization module presented in this work gives more accurate and realistic results than the Boston model.

B. Link utilization as a Function of Buffer Size on 10Gbps Optical Network

In this section, we show simulation results for AIMD, HSTCP, CUBIC, and HTCP [15] for 10Gbps link (results for rest of the high speed TCP variants will be included in an extended paper). For the results presented in this subsection, we set C=10Gbps, link delay =10ms, RTTs of 10 flows are ranging from 80ms to 260ms with average RTT = 170ms. We use average RTT to calculate BDP which gives 141667 1500B packets maximum.

In figure 6(a), we show link utilization as a function of fraction of maximum buffer size for three different values of X. We observe that more than 90% link utilization can be achieved with buffer size fraction of 0.05. We also observe as buffer size increases link utilization improves. As shown in the previous results, de-synchronization improves the link utilization (with synchronization parameter X=0.3, we get higher link utilization than X=1 and X=0.6 case).

We perform the same simulation on CUBIC TCP. As shown in figure 6(b), the performance of CUBIC TCP is drastically affected by smaller buffer size. We observe less than 90% link utilization for all the three values of X. We also observe for buffer sizes greater than 20% of BDP, reduction in level of synchronization improves the performance however we observe higher synchronization (X=0.6) don't show any improvement than all synchronized flows (X=1). We also observe HSTCP performs better than CUBIC TCP with various buffer sizes. CUBIC TCP is designed to be more fair where as HSTCP has RTT fairness problem. Although CUBIC and HSTCP has some performance differences, we believe there has to be some tradeoff between fairness and link utilization. Fairness of the TCP flows is out of scope of this paper and is left for further exploration. AIMD result is presented in figure 6(c). We observe for the case of X=1, 10 AIMD flows behave like 10 parallel TCP flows. We observe we get more than 90% link utilization even for 5% of BDP. We observe that link utilization improves with the lower degree of synchronization. It is also observed that, 10 AIMD outperforms CUBIC and comparable to HSTCP for all buffer size cases. HSTCP and AIMD results are close to each other because congestion control algorithm of HSTCP emulates the parallel TCP flows. HTCP (figure 6(d)) performs poorly for small buffer sizes. We observed that frequent losses render the ability of synchronized HTCP flows to utilize the available bandwidth when 0.05BDP buffer size is used. However, de-synchronized flows are able to show better performance.



(a) HSTCP (b) CUBIC (c) AIMD (d) HTCP

Fig. 6. Average link utilization as function of fraction of buffer size on 10Gbps link (Max buffer = 141667 packets)

The main reason for the poor performance of CUBIC and HTCP as compared to AIMD and HSTCP is attributed to its improve fairness. In de-synchronized environment, both CUBIC and HTCP mark the last congestion event by setting the values of W_{max} and $time\ since\ last\ congestion\ event$ parameters respectively. However, if some flows miscalculate this congestion point if they do not record any packet loss. The inherent idea of these two TCP mechanisms is to be fair with other flows traversing the same bottleneck link. There is an intrinsic tradeoff between fairness and efficiency of these protocols. The exploration of this relationship is left for future work.

IV. Related Work

The rule-of-thumb, first stated in [16] and further studied in [1] is challenged in [2]. They assume that there are N long lived flows which are stochastically multiplexed at a router buffer requiring $BDP/\sqrt{N}$. On the similar hypothesis [3] proposes the buffer size is 0.63X$BDP/\sqrt{N}$. In [17], the sufficient buffer size of $BDP^2/32N^3$ can provide near 100% link utilization. The assumption that the number of flows in the network remains constant is further investigated in [18]. They conclude that depending on the core-to-access speed ratio, the buffer size and the number of flows are not independent from each other and therefore, these two quantities should not be treated independently. And O(1) buffer sizes are good enough for near 100% link utilization given the core-to-access ratio is large. [4] considers packet loss rate as an important metric and attempt to bound the loss rate to small value to achieve good link utilization with small buffers. Packet loss rate is proportional to N^2 where N is the number of flows and hence shown to be an important parameter while designing router buffers [19]. Some researchers also tried to attempt the problem from a different prospective e.g. in [20] input/output capacity ratio is considered to be an important metrics as far as end user is concerned. [9] presents an analytical model focused on effect of buffer size on HSTCP performance. Although the results shown in this work argue that a smaller buffer can be sufficient to give near 100% link utilization, they also assume that there are long lived flows in the network. Their study is solely focused on HSTCP and don't apply to other high-speed variants.

V. Summary and Conclusion

We focus on a simple yet critical setup to get a clear understanding of the underlying mechanisms behind role of synchronization level and congestion control on different buffer sizes in terms of link utilization. A loss synchronization module for fluid model simulation is proposed to show the effect of different degree of loss synchronization on high speed networks. We present the fluid simulation results showing the effect of different buffer sizes on bottleneck link utilization for high speed TCP flows. Simulation results for HSTCP, CUBIC and AIMD are presented to show the effect of different buffer sizes on link utilization. One can view the presented loss synchronization module as a black box, where loss synchronization data can be fed from real experiments or one can utilize some theoretical distribution models.

Because of space limitations, effect of synchronization on fairness properties of TCP flows is left for further exploration. Observed performance of CUBIC and HTCP requires further exploration of intrinsic tradeoff between fairness and link utilization. The summary of simulation results are presented below:

- Buffer size of less than 10% of BDP is sufficient to achieve more than 90% link utilization for both HSTCP and AIMD.

- Both CUBIC and HTCP require larger buffer size for better performance.
- Increase or decrease of loss synchronization levels does not show much improvement in the performance of de-synchronized HTCP and CUBIC flows. Whereas, Lower synchronization further improves the link utilization for HSTCP amd AIMD.

We observe that buffer size of bottleneck buffer plays an important role on performance of high speed networks. To develop next generation high speed networks, we need to study the relationship among congestion control algorithms, bottleneck buffers and queuing policies. While the high speed experimental facilities are still in development phase and packet based simulation method are bottlenecked by execution time and memory, the presented method can act as a tool for protocol developers. Although we presented a simple work, explorations of more complicated scenario can be expanded from this work.

REFERENCES

[1] Curtis Villamizar and Cheng Song, "High performance tcp in ansnet," *SIGCOMM Comput. Commun. Rev.*, vol. 24, no. 5, pp. 45–60, 1994.

[2] Guido Appenzeller, Isaac Keslassy, and Nick McKeown, "Sizing router buffers," in *ACM SIGCOMM*, Portland, August 2004.

[3] Damon Wischik and Nick McKeown, "Part i: buffer sizes for core routers," *SIGCOMM Comput. Commun. Rev.*, vol. 35, no. 3, pp. 75–78, 2005.

[4] Amogh Dhamdhere, Hao Jiang, and Constantine Dovrolis, "Buffer sizing for congested internet links," in *IEEE Infocom*, Miami, FL, March 2005.

[5] Sally Floyd, "Highspeed tcp for large congestion windows," Tech. Rep. RFC 3649, Dec 2003.

[6] Sangtae Ha, Injong Rhee, and Lisong Xu, "Cubic: A new tcp-friendly high-speed tcp variant," *ACM SIGOPS Operating System Review*, vol. 42, no. 5, pp. 64–74, July 2008.

[7] Ryosei Takano, Yetsu Kodama, Tomohiro Kudoh, Motohiko Matsuda, and Fumihoro Okazaki, "Realtime burstiness measurement," in *Proc. PFLDnet*, Nara, Japan, February 2006.

[8] Hassayoun S. and Ros D., "Loss synchronization and router buffer sizing with high-speed versions of tcp," in *IEEE INFOCOM Workshops*, Phoenix, AZ, USA, April 2008.

[9] Dhiman Barman, Georgios Smaragdakis, and Ibrahim Matta, "The effect of router buffer size on highspeed tcp performance," in *GLOBECOM*, 2004.

[10] Suman Kumar, Seung-Jong Park, S. Sitharama Iyengar, and Jung-Han Kimn, "Time-adaptive numerical simulation for high speed networks.," in *HPCNCS*. 2007, pp. 198–205, ISRST.

[11] Suman Kumar, Seung-Jong Park, and S. Sitharama Iyengar, "A loss-event driven scalable fluid simulation method for high-speed networks," *Computer Networks - Elsevier.*, vol. 54, no. 1, pp. 112–132, 2010.

[12] Vishal Misra and WeiBo Gong andDon Towsley, "A fluid-based analysis of a network of aqm routers supporting tcp flows with an application to red," in *ACM SIGCOMM*, Sept. 2000.

[13] Y. Liu, F. L. Presti, Y. Liu, F. L. Presti, V. Misra, D. Towsley, and Y. Gu, "Fluid models and solutions for large-scale ip networks," in *ACM SIGMETRICS*, 2003.

[14] Mei Wang, "Mean-field analysis of buffer sizing," in *Global Telecommunications Conference, 2007. GLOBECOM '07. IEEE*, Nov. 2007, pp. 2645–2649.

[15] D. Leith and R. Shorten, "H-tcp: Tcp for high-speed and long-distance networks," in *in Proc. PFLDnet, Argonne*, 2004.

[16] V. Jacobson, "Congestion avoidance and control," in *ACM SIGCOMM*, August 1988.

[17] Avrachenkov K., Ayesta U., and Piunovskiy A., "Optimal choice of the buffer size in the internet routers," in *IEEE Conference on Decision and Control*, Spain, 2005.

[18] A. Lakshmikantha, R. Srikant, and C. Beck, "Impact of file arrivals and departures on buffer sizing in core routers," in *IEEE Infocom*, 2008.

[19] R. Morris, "Tcp behavior with many flows," in *ICNP '97: Proceedings of the 1997 International Conference on Network Protocols (ICNP '97)*, Washington, DC, USA, 1997, p. 205, IEEE Computer Society.

[20] Ravi S. Prasad, Constantine Dovrolis, and Marina Thottan, "Router buffer sizing revisited: the role of the output/input capacity ratio," in *CoNEXT '07: Proceedings of the 2007 ACM CoNEXT conference*, New York, NY, USA, 2007, pp. 1–12, ACM.