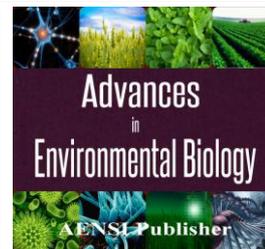




AENSI Journals

Advances in Environmental Biology

ISSN-1995-0756 EISSN-1998-1066

Journal home page: <http://www.aensiweb.com/AEB/>

Examining Methods to Control Network Congestion

¹Sajjad Baghernezhad and ²Ali Hosseini

¹Department of Computer, Darab Branch, Islamic Azad University, Darab, Iran

²Young Researchers and Elite Club, Beyza Branch, Islamic Azad University, Beyza, Iran

ARTICLE INFO

Article history:

Received 19 June 2014

Received in revised form

19 September 2014

Accepted 29 September 2014

Available online 10 November 2014

Keywords:

Congestion control, throughput, TCP, TCP Reno, network, routing.

ABSTRACT

Background: Internet success depends considerably on its protocol's ability and strength. If we look at internet, we find that almost all applications current in internet depend on TCP/IP protocol. **Objective:** The users' demands and applications grow increasingly due to daily internet growth; the TCP/IP's disadvantages have become clear increasingly due to such challenges. **Results:** The packets are lost more and network output decrease so the users have encountered with essential problems. **Conclusion:** This article emphasizes on important challenge related to internet and simulates congestion control techniques such as TCP and TCP Reno by NS2 software and examine network throughput mean parameter to present mechanisms increasing network efficiency.

© 2014 AENSI Publisher All rights reserved.

To Cite This Article: Sajjad Baghernezhad and Ali Hosseini., Examining Methods to Control Network Congestion. *Adv. Environ. Biol.*, 8(12), 1436-1440, 2014

INTRODUCTION

A user considers a network with congestion when its share from the network sources decreases due to network load increase. If the network output remains stable for a special user even at presence of a high load, the user does not consider the network encountered with congestion, but if the network efficiency and output are with changes harmful to other users, they will find congestion in the network[1].

In addition to above definition if total demands for a source go beyond the capacity available to it for each time interval, the source in that time interval is considered with congestion; this definition uses the relation between demand and available data to identify congestion periods in the network. The demands include data delivery from a point to another one and meet the user's limits such as permitted delays and reliability, but the available data include network sources such as buffer space, lines bandwidth and processor speed. If all the demands and needs are met, the network is not considered as congested[2]. Considering increasing users and new applications it is necessary to enable internet to manage new challenges; in line with this in addition to network sources increase to meet the demands, it is necessary to have a sound strategy to minimize the congestion effects, too.

2 – Study history:

Internet success depends considerably on its protocols' ability and strength. If we look at internet, we find that most of applications current in internet depends on TCP/IP protocol. In previous decades the essential success of TCP/IP is its ability to serve in high traffic times.

The main reason of above potential is related to TCP congestion control. It is believed about TCP congestion control that the network load is controlled by transmitter nodes which regulate their transmission speed by virtue of the congestion rate in the network.

TCP follows a saw-like paradigm window in traditional and usual form in a way that the window increases continuously to become a lost packet; although it allows TCP to search for additional bandwidth such behavior leads to lose the packet (Figure 1)[3,4].

If a TCP source receives three repetitive feedbacks for each sent packet, it finds a lost packet and the window size decreases to one and threshold to half to prevent the congestion and another packet be lost. If a TCP source finds all its packets have been delivered, it increases slowly the packets sending to use optimally the speed network capacity.

Corresponding Author: Sajjad Baghernezhad, Department of Computer, Darab Branch, Islamic Azad University, Darab, Iran.

Also if TCP Reno version receives three repetitive feedbacks for each sent packet, a packet has been lost; but the window size decreases to half to prevent congestion and another packet be lost (Figure 2)[5].

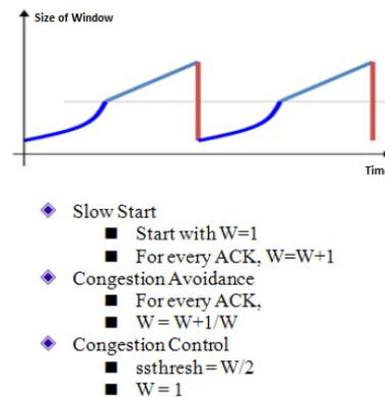


Fig. 1: TCP Algorithm.

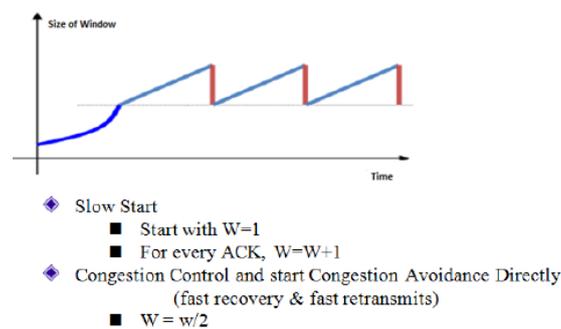


Fig. 2: TCP Reno Algorithm.

By virtue of these methods TCP could minimize effectively the lost packets and benefit from the maximal network capacity, but the users' demands and applications grow increasingly due to daily internet growth so the TCP/IP's disadvantages have become clear increasingly; more increased lost packets and network output decrease have created essential problems for the users; besides, inability to support new services of vast development has delayed the users. We will assess the two mechanisms as follows:

3 – Assessment:

Two protocols were simulated to assess TCP and TCP Reno operations. All the simulations were done by the NS2[6] assmilator software.

3 – 1 – Light congestion occurrence:

These assmilators use the supposed network (Figure 3). The throughput parameter was to assess the mechanisms in a way that the TCP transmitter nodes throughput mean using FTP are the measure for comparison. The transmitter nodes were ten.

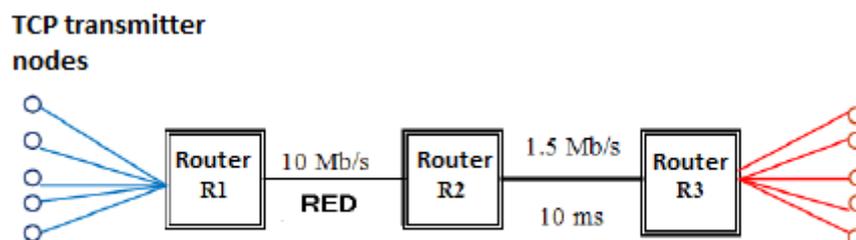


Fig. 3: The network supposed to assess.

All the nodes use 1,000 byte packets to enable the network drawn in Figure 3 to include the structure related to different types of networks.

The line delay between R1 and R2 routers is 25 – 500 ms and includes low delays for the routers near each other and high delays for the routers far from each other. The assimilation lasts 200 second and the buffer space is 25 for the routers. Also the parameters related to RED have been ranged as follows[7,8]:

1 – Minimum threshold = 50 percent of the buffer space.

2 – Maximum threshold = 75 percent of buffer space.

Figure 4 shows the throughput mean gained for ten transmitter nodes of TCP traffic and Figure 5 shows the throughput when used in TCP Reno traffic in the transmitters.

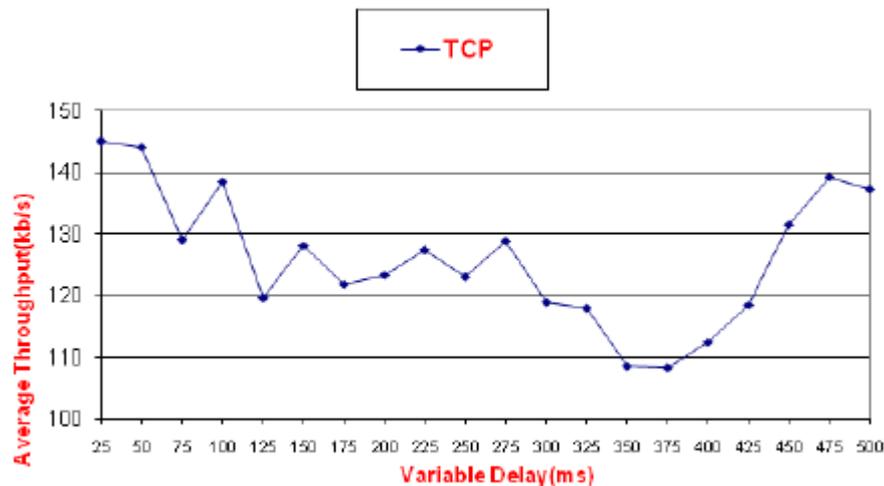


Fig. 4: Throughput mean gained when TCP used.

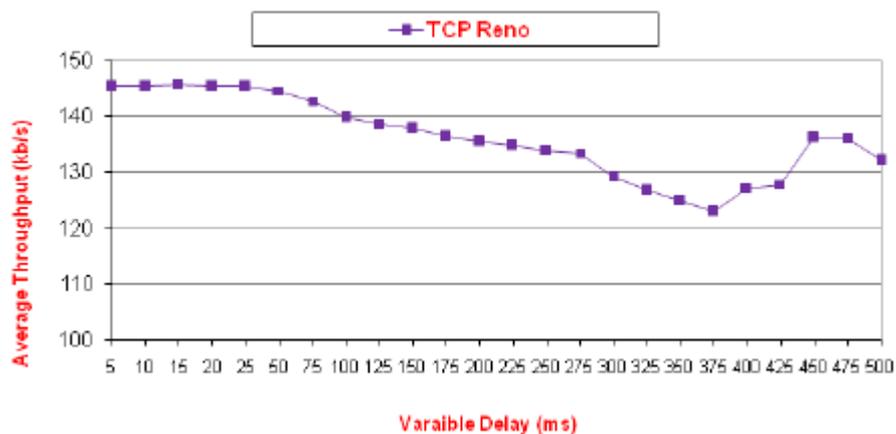


Fig. 5: Throughput mean gained when TCP Reno used.

Table 1 Comparing two mechanisms: TCP and TCP Reno for some amounts of the delay.

Table 1: Transmitter's nodes throughput mean.

Delay(ms)	TCP (kb/s)	TCP Reno (kb/s)
25	144.902	145.4289
50	140.5859	144.6075
75	129.0598	142.7694
100	138.5540	139.8633
250	119.8016	133.8246
375	122.4458	122.9385
450	131.5145	136.1913
500	137.3784	132.1749

Table 1 and the throughput mean show the TCP Reno mechanism improves the network efficiency compared to TCP. Also it was shown that TCP Reno has a relative stable behavior in low delays where the packets reach the destination sooner and congestion is more probable while the TCP protocol has created many fluctuations, but TCP Reno use shows no special advantage in very high delays (More than 400 ms).

2 – 3 – High congestion occurrence:

The simulations occurred in this part use the supposed network, too but the UDP traffic is done to the network; the diagram related to new network is shown in Figure 6.

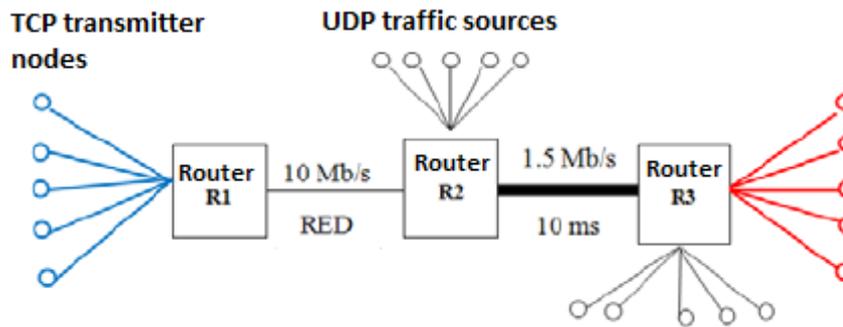


Fig. 6: The network supposed to assess.

UDP traffic sources have exponential distribution with 2.50 s mean and send their packets with 100 Kb/s speed namely each source sends packets with 100 Kb/s per 2.50 s and sends no packet in following 2.50 s. the UDP nodes are 7 and the TCP nodes are ten. These simulations are to assess the mechanisms proposed at presence of UDP traffic without any reaction to congestion[9].

Figure 7 shows the throughput mean for ten transmitter nodes of TCP traffic and Figure 8 shows the throughput mean when TCP Reno traffic is used.

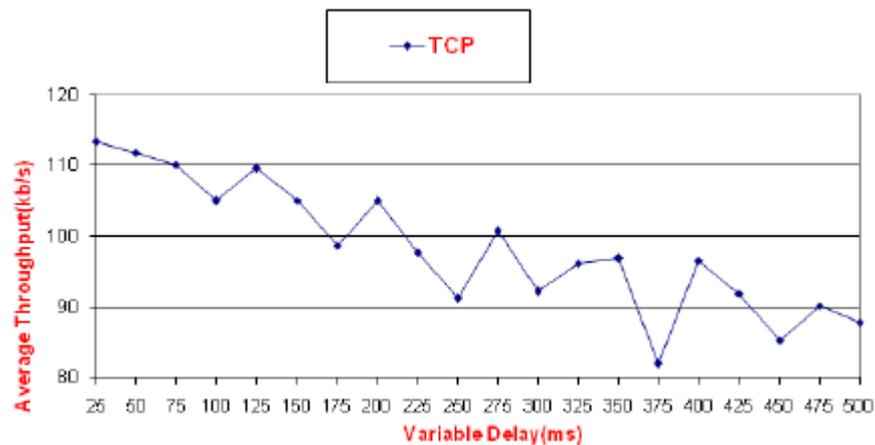


Fig. 7: Throughput mean gained when TCP used.

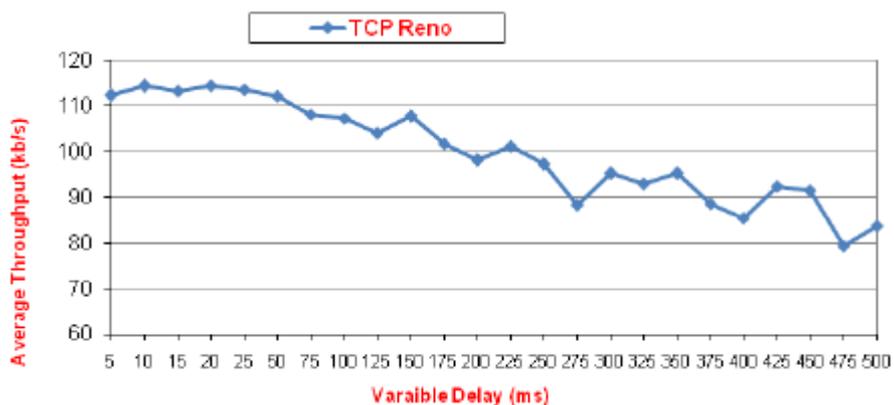


Fig. 8: Throughput mean gained when TCP Reno used.

Table 2 shows the comparison between TCP and TCP Reno mechanisms for some amounts of delays.

Table 2: Transmitter's nodes throughput mean.

Delay(ms)	TCP (kb/s)	TCP Reno (kb/s)
25	113.2978	113.6567
50	111.8859	112.0966
75	110.1008	108.1863
100	105.0292	107.3364
250	91.2531	97.2674
375	81.9883	88.6794
450	85.3879	91.5373
500	87.82	83.7013

As expected the UDP traffic operations decrease the network output for TCP and TCP Reno traffic transmitters, but Tables 2, 7 and 8 show the TCP Reno use even when network encountering more congestion increases the network efficiency but has not considerable benefit in very high delays (More than 400 ms).

4 – Conclusion:

This article analyzed TCP to show its disadvantages in congestion control and indicates the main problem concerning congestion namely related to intervals between congestion and its distinction by final nodes exists even by using better algorithms. Also it examines TCP Reno to manage congestion to increase the presented network efficiency by different simulators; the simulation findings based on NS2 indicate although TCP Reno benefits from the lost packets rate's decrease potential, but its function way is not clear when the congestion is durable and chronic; besides, it does not guarantee fair sharing between the sharer's communications in the same line. With two essential challenges related to internet namely increasing users and new applications IETE hopes to optimize queue management algorithms efficiency by understanding how internet traffic changes in limited times and control the lost packets' rate in internet by creating AQM (Active queue management) mechanisms, but in addition to AQM mechanisms in routers the congestion control mechanisms in the final nodes are effective in congestion management[10].

This article is based on unicasting process so this research development to support multicasting will improve the network efficiency to encounter congestion due to multicasting traffics[11].

REFERENCES

- [1] Floyd, S. and K. Fall, 2009. "Promoting the Use of End-to-End Congestion Control in the Internet". IEEE/ACM Transactions on Networking, 7(4): 458-472.
- [2] Jacobson, V., 2008. "Congestion Avoidance and Control," Proc. SIGCOMM Symposium on Communications Architectures and Protocols, pp. 3 Active Queue Management 314-329, ACM SIGCOMM, Stanford, CA.
- [3] Widmer, J., R. Denda and M. Mauve, 2009. "A Survey on TCP-Friendly Congestion Control (Extended Version)," Tech. rep. TR-2001-002, Dept. of Math. and Comp. Sci., Univ. of Mannheim.
- [4] Floyd, S., 2004. "TCP and Explicit Congestion Notification," ACM Computer Communication Review, 24(5): 10-23, ACM.
- [5] Padhye, J., *et al.*, 2010. "Modeling TCP Reno Performance: A Simple Model and Its Empirical Validation," IEEE/ACM Trans. Net., 8(2): 133-45.
- [6] NS2, "Network Simulator -- ns version 2", online software:www.isi.edu/nsnam/ns.
- [7] Floyd, S. and V. Jacobson, 2013. "Random Early Detection gateways for Congestion Avoidance", IEEE/ACM Transactions on Networking, 1(4): 397-413.
- [8] Lin, D., R. Morris, 2007. "Dynamics of Random Early Detection", Proceedings of ACM SIGCOMM.
- [9] Postel, J., 2000. "User Datagram Protocol", IETF, RFC 768.
- [10] May, M., C. Diot, B. Lyles, J. Bolot, 2012. "Aggregate Traffic Performance with Active queue Management", Technical Report.
- [11] Li-wei Lehman, J. Stephen, Garland and L. David. Tennenhouse, "Active Reliable Multicast", IEEE INFOCOM.