7th International Conference on Recent Innovations in Science, Engineering and Management
The Institutions of Engineers, Delhi State Center (India)
16th September 2016, www.conferenceworld.in
RISEM - 16
ISBN : 978-93-86171-07-8

PERFORMANCE EVALUATION OF CONGESTION CONTROL PROTOCOLS TCP-RENO, VEGAS, LP, WESTWOOD IN WIRELESS NETWORK

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ABSTRACT

"Performance evaluation of congestion control protocols (TCP-Reno, Vegas, LP, Westwood) in wireless network" in that the wireless communication Transmission Control Protocol (TCP) plays a vital role in developing communication systems which provides higher and reliable communication capabilities in most styles of networking atmosphere. This paper concentrates on comparative study of the various congestion management protocols supported some performance metrics (Packet Delivery Ratio, Throughput, End2End delay) with some basic scenarios like- RTT(Round Trip Time) and Error Probability. The purpose of this paper is to control congestion and improve performance of congestion control protocols which is implemented in Network Simulator (NS-2). Fairness is a vital and knowledge domain topic used in several fields. This text conjointly discusses fairness index of TCP protocols in wireless networks.

I. INTRODUCTION

Wireless network could be a network established by victimization radio emission frequency to speak among computers and alternative network devices. Generally it's additionally brought up as Wi-Fi network or LAN. This network is obtaining standard these days attributable to straightforward to setup feature and no cabling concerned. You'll be able to connect computers anyplace in your home while not the requirement for wires. In networks, the packet loss will occur as a result of transmission errors, however most often owing to congestion. TCP's congestion management mechanism reacts to packet loss by dropping the amount of unacknowledged information segments allowed within the network.TCP flows with similar round-trip times (RTTs) that shares a common bottleneck to reduce their rates so that the accessible bandwidth will be constantly, distributed equally among them.[1]

1.1 Congestion Control in Wireless Network

The congestion control in remote systems has been generally explored throughout the years and a number of plans and techniques are created, all with the purpose of enhancing execution in remote system. With the speedy growth and implementation of wireless technology it's essential that the congestion management drawback be resolved. The Transmission management Protocol (TCP) has been wide employed in today's net.

ISBN : 978-93-86171-07-8

The protocol supports reliable information transport by establishing an affiliation between the transmissions and receiving ends. The transmitter starts a timeout mechanism once causation a packet to the receiver. The transmitter perpetually tracks the round-trip times (RTTs) for its packets as a method to work out the acceptable timeout amount. At the receiver, every received packet is acknowledged implicitly or expressly to the transmitter. If the transmitter doesn't receive associate in Nursing acknowledgment for a given packet once the corresponding timeout amount expires, the packet is deemed to be lost and subject to retransmission. A congestion window with dynamically adjusted size is employed by the protocol to control the traffic be due the transmitter to the receiver. Though communications protocol was at first designed and optimized for wired networks, the growing quality of wireless information applications has lead third generation wireless networks like CDMA2000 and UMTS networks to increase communications protocol to wireless communications still. The initial objective of communications protocol was to expeditiously use the obtainable information measure within the network and to avoid overloading the network (and the ensuing packet losses) by suitably strangling the senders' transmission rates. Network congestion is deemed to be the underlying reason for packet losses. Consequently, communications protocol performance is commonly off once employed in wireless networks and needs numerous improvement techniques. A key issue inflicting the off performance is that the communication system quality in wireless networks will fluctuate greatly in time thanks to channel weakening and user quality.

1.2 Transmission Control Protocols (TCP)

TCP is a connection-oriented protocol, which means a connection is established and maintained until the application programs at each end have finished exchanging messages. TCP, easily the most widely used protocol in the transport layer on the Internet (e.g. HTTP, TELNET, and SMTP), plays an integral role in determining overall network performance. The TCP congestion-avoidance algorithm is the primary basis for congestion control in the Internet. There are some congestion control protocols of TCP which are TCP-Reno, TCP-Vegas, TCP-LP, and TCP-Westwood. These transmission control protocols are given below:

1.3 TCP-Reno

TCP Reno implements an algorithm called Fast recovery. A fast retransmit is sent, half of the current CWND is saved as SS Thresh and as new CWND, thus skipping slow start and going directly to Congestion Avoidance algorithm. Slow start assumes that unacknowledged segments are due to network congestion. While this can be a suitable assumption for several networks, segments could also be lost for alternative reasons, like poor link layer transmission quality. Thus, slow begin will perform poorly in things with poor reception, like wireless networks.

1.4 TCP-Vegas

TCP Vegas is a TCP congestion avoidance algorithm that emphasizes packet delay, rather than packet loss, as a signal to help determine the rate at which to send packets. It was developed at the University of Arizona by Lawrence Brakmo and Larry L Peterson. TCP Vegas detects congestion at an incipient stage based on increasing Round-Trip Time (RTT) values of the packets in the connection unlike other flavors such

ISBN : 978-93-86171-07-8

as Reno, New Reno, etc., which detect congestion only after it has actually happened via packet loss. The algorithm depends heavily on accurate calculation of the Base RTT value. If it is too small then throughput of the connection will be less than the bandwidth available while if the value is too large then it will overrun the connection. [2]

II. TCP-LP

Service prioritization among completely different traffic categories is a crucial goal for the web. Typical approaches to resolution this downside think about the present best-effort category because the low-priority category and plan to develop mechanisms that give "better-than-best-effort" service. a brand new distributed formula to appreciate a low-priority service (as compared to the present best effort) from the network endpoints. TCP Low Priority (TCP-LP), a distributed formula whose goal is to utilize solely the surplus network information measure as compared to the "fair share" of information measure as targeted by TCP. The key mechanisms distinctive to TCP-LP congestion management square measure the utilization of unidirectional packet delays for early congestion indications and a TCP-transparent congestion shunning policy

2.1 TCP-Westwood

TCP Westwood (TCPW) could be a sender-side-only modification to communications protocol new reno that's supposed to raised handle massive bandwidth-delay product ways (large pipes), with potential packet loss attributable to transmission or different errors (leaky pipes), and with dynamic load (dynamic pipes). Transmission control protocol Westwood depends on mining the ACK stream for data to assist it higher set the congestion control parameters: Slow begin Threshold (ssthresh), and Congestion Window (cwin). In communications protocol Westwood, an "Eligible Rate" is calculable and utilized by the sender to update ssthresh and cwin upon loss indication, or throughout its "Agile Probing" section, a projected modification to the well-known slow begin section. Additionally, a theme referred to as Persistent Non Congestion Detection (PNCD) has been devised to find persistent lack of congestion and induce an agile inquiring section to efficiently utilize massive dynamic information measure.

III. LITERATURE REVIEW

Performance Analysis of Linux-Based TCP Congestion Control Algorithms in VANET Environment [1] was represented by Kire Jakimoski (2016), tells about the transport layer protocol is used by the majority of the applications in wire and wireless networks. In addition, there are a lot of challenges in testing and applying this protocol in highly dynamic topological Vehicular Ad Hoc Networks (VANET). Quality of Service and reliability of the Transmission Control Protocol in vehicular wireless network is not assured due to frequent link failure, short duration of session, packet drop due to congestion, multipath propagation delay etc. Performances of the Transmission Control Protocol (TCP) are highly depended on congestion control algorithms that limit the amount of data to be transmitted based on the estimated network capacity and receiver's TCP window. Goal of the researcher is to analyze and evaluate TCP congestion control algorithms in VANET environment and suggest the most appropriate of them

Performance Analysis of TCP Congestion Control Algorithms [2] was represented by authors Habibullah Jamal and Kiran Sultan in 2008. The author presents the fast transfer of large volume of data, and the deployment of the network infrastructures is ever increasing. However, the dominant transport protocol of today, TCP, does not meet this because it favors reliability over timeliness and fails to fully utilize the network capacity due to limitations of its conservative congestion control algorithm. The slow response of TCP in fast long distance networks leaves sizeable unused bandwidth in such networks. A large variety of TCP variants have been proposed to improve the connection’s throughput by adopting more aggressive congestion control algorithms. Some of the flavors of TCP congestion control are loss-based, high-speed TCP congestion control algorithms that uses packet losses as an indication of congestion; delay-based TCP congestion control that emphasizes packet delay rather than packet loss as a signal to determine the rate at which to send packets

TCP-LP: Fair and Friendly Congestion Control Approach [3] was purposed by Deepak Mehta, Internet users always seek service prioritization. This service can be defined as “Give importance to important network traffic over unimportant network traffic”. Conventional methods can be categorization of traffic by considering the existing traffic as “best-effort” class can be named as low-priority (LP) class, and keen to develop mechanisms which will give “better-than-best-effort” service. It is worth mentioning that this paper is going to mention the ides of developing a Low Priority distributed algorithm whose objective is to utilize only the rest bandwidth and give priority to other delay sensitive traffic generated by interactive applications or media streaming application and interested in devising a efficient approach or algorithm called novel distributed algorithm to implement a LP service which work as against existing best effort traffic service from the communication endpoints.

Comparative study of congestion control techniques[4] was represented by author Shakeel Ahmad was represented this paper in 2009. Congestion in network occurs due to exceed in aggregate demand as compared to the accessible capacity of the resources. Network congestion will increase as network speed increases and new effective congestion control methods are needed, especially to handle "bursty" traffic of today’s very high speed networks. Since late 90’s numerous schemes etc. have been proposed. The author concentrates on comparative study of the different congestion control schemes based on some key performance metrics. An effort has been made to judge the performance of Maximum Entropy (ME) based solution for a steady state GE/GE/1/N censored queues with partial buffer sharing scheme against these key performance metrics.

Unified Framework for Modeling TCP-Vegas, TCP-SACK, and TCP-Reno [5] was represented by author Adam Wierman in 2003, the author represent a general analytical framework for the modeling and analysis of TCP variations. The framework is quite comprehensive and allows the modeling of multiple variations of TCP, i.e. TCP-Vegas, TCP-SACK, and TCP-Reno, under very general network situations. In particular, the framework allows us to propose the first analytical model of TCP-Vegas under on-off traffic - all existing analytical models of TCP-Vegas assume bulk transfer only. All TCP models are validated against event driven simulations (ns-2) and existing state-of-the-art analytical models. Finally, the analysis provided by framework

ISBN : 978-93-86171-07-8

leads to many interesting observations with respect to both the behavior of bottleneck links that are shared by TCP sources and the effectiveness of the design decisions in TCP-SACK and TCP-Vegas.

3.1 Simulation Tool

NS (from network simulator) is a name for a series of discrete event network simulators, specifically ns-1, ns-2and ns-3. All of them are discrete-event computer network simulators, primarily used in research and teaching. Ns-3 is free software, publicly available under the GNU GPLv2 license for research, development. Ns-2 simulator tool is implemented for evaluating the results of the entire network.

3.2 Ns-2

Ns started as a variation of the essential system machine in 1989 and has advanced well in the course of recent years. In 1995 ns improvement was upheld by government organization through the VINT venture at LBL, Xerox PARC, UCB, and USC/ISI. without further ado ns improvement is backing through government organization with albizia and through National Science Foundation with CONSER, each untidily with various analysts together with ACIRI. Ns has consistently encased significant commitments from various researchers, together with remote code from the UCB Daedelus and CMU Monarch comes and Sun Microsystems. For documentation on late changes, see the variant a couple of revision log.

IV. PERFORMANCE METRICS

There are basically three types of performance metrics used for congestion control and there is a fairness finding index which is used to find the fairness.

4.1 PDR (Packet Delivery Ratio)

Packet delivery quantitative relation is outlined because the quantitative relation of knowledge packets received by the destinations to those generated by the sources. Mathematically, it is outlined as: PDR= S1÷ S2 wherever, S1 is that the total of knowledge packets received by the every destination and S2 is that the total of knowledge packets generated by the every supply. Graphs show the fraction of knowledge packets that square measure with success delivered throughout simulations time versus the quantity of nodes. Performance of the DSDV is reducing often whereas the PDR is increasing within the case of DSR and AODV. AODV is best among the 3 protocols.

4.2 Throughput

It is outlined because the total range of packets delivered over the whole simulation time. The output comparison shows that the 3 algorithms performance margins area unit terribly shut beneath traffic load of fifty and one hundred nodes in MANET state of affairs and has giant margins once range of nodes will increase to two hundred. Mathematically, it is outlined as: Throughput= N/1000 wherever N is that the range of bits received with success by all destinations.

ISBN : 978-93-86171-07-8

4.3 End2End Delay

The normal time it takes a data bundle to understand the goal. This incorporates all potential deferrals brought on by buffering all through course disclosure dormancy, lining at the interface line. This metric is figured by subtracting time at that underlying parcel was transmitted by give from time at that underlying data bundle touched base to goal. Scientifically, it's printed as: Avg. EED=S/N where S is that the add of the time spent to convey parcels for every goal, and N is that the change of bundles got by the all goal hubs

4.4 Fairness Index

TCP fairness requires that a new protocol receive no larger share of the network than a comparable TCP flow. This is important as TCP is the dominant transport protocol on the Internet, and if new protocols acquire unfair capacity they tend to cause problems such as congestion collapse.

4.5 Jain's Fairness Index

$f(x_1,x_2,x_3,\ldots,x_n) = (\sum x_i)^2/n \sum x_i^2$

- Where i goes from 1 to n.
- A definition for fairness:
- 0 <= f () <= 1, given flow throughputs x
- where x_i is the normalized throughput
- n is the number of connections

Locally equal partitioning of bandwidth achieves index of 1. If only k of n flows receive equal BW (and others get none), index is k/n. [6]

V. RESULTS

The results can be formulated in the form of tables and graph plots. These results have been obtained by studying three different scenarios in which the different parameter among the different performance metrics can be evaluated.

5.1 Result for Scenario (various RTT)

In this section the tables and graphs of different performance metrics with different values of RTT (round trip time) and constant Error Probability are given:

5.1.1 Packet Delivery Ratio

Packet delivery ratio is defined as the ratio of data packets received by the destinations to those generated by the sources. The graph of PDR is given below:

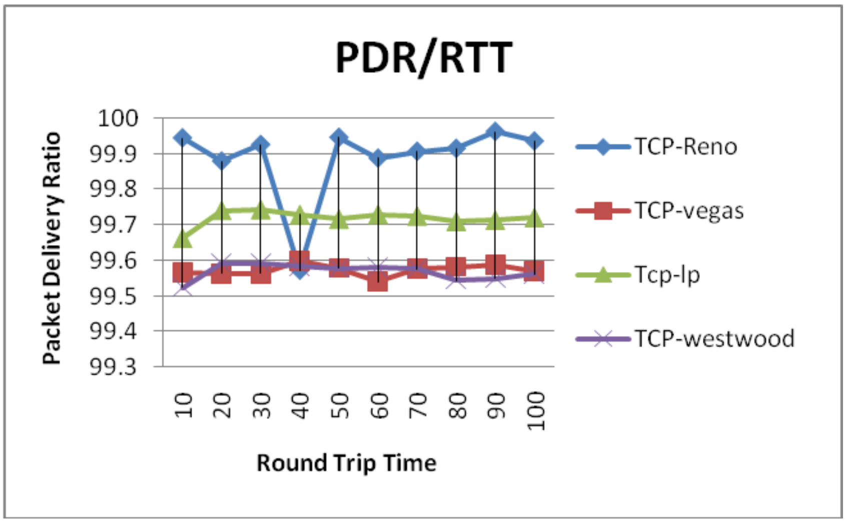

Figure 1.1

This Figure 1.1 shows that RTT values ranges from {10, 20, 30, 40, 50, 60, 70, 80, 90, 100} in which the TCP-Westwood and TCP-Vegas is approximately in the lowest position according to others. TCP-LP in this case is quite increase the performance of network but the result of TCP-Reno is approximately up to 100 which is completely improved the performance of network.

5.1.2 Throughput

It is defined as the total number of packets delivered over the total simulation time. The and graph of throughput is given below:

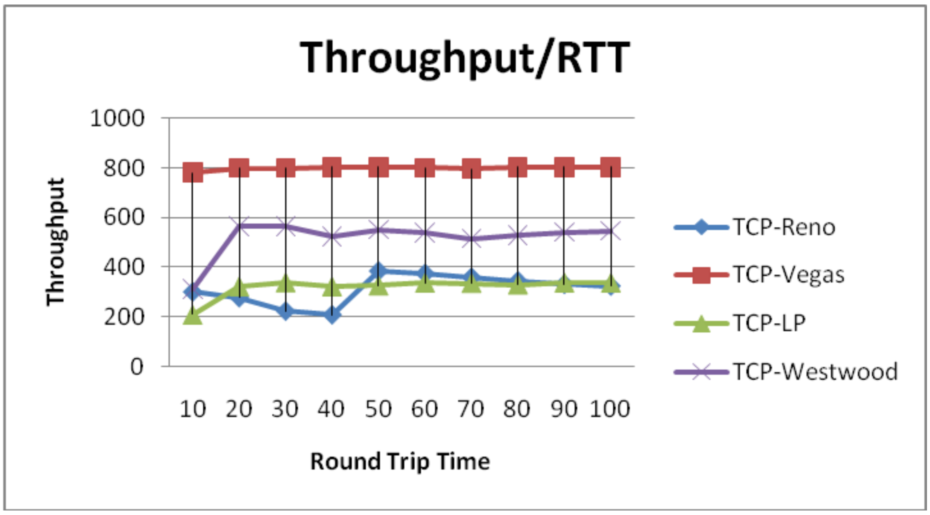

Figure 1.2

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5.1.3 End2End Delay

The average time it takes a data packet to reach the destination. This includes all possible delays caused by buffering during route discovery latency, queuing at the interface queue. The graph of End2End Delay is given below:

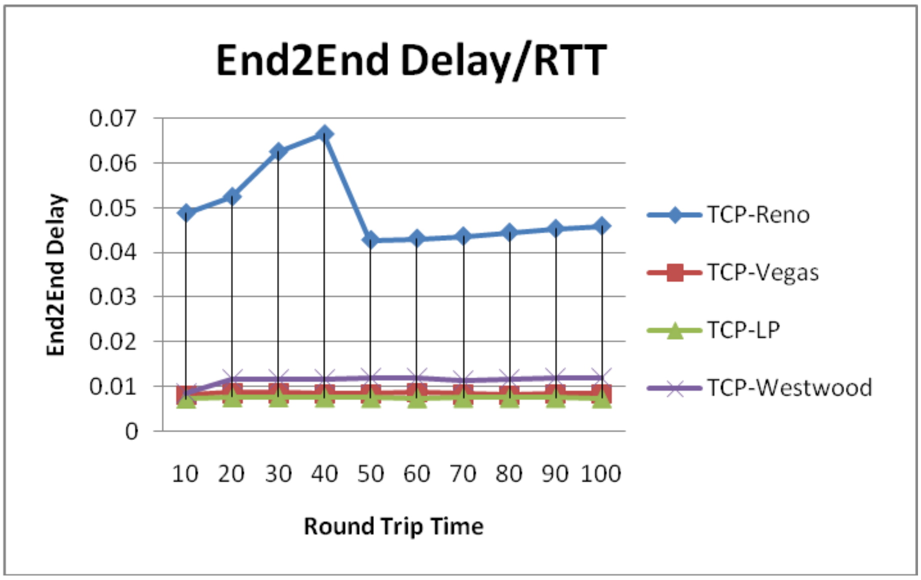

Figure 1.3

In this figure 1.3 shows that result of TCP-Vegas, TCP-LP, TCP-Westwood are constantly increase the performance of the network by decreasing the delay of packets. In which the TCP-Reno result shows that it decrease the performance by increase the value of End2End delay.

5.2 Results for Fairness Index

Fairness index is used to find the fairness among all the protocols of the network. This graph shows the fairness results of all protocol in which the result are evaluated according to the Jain's Fairness Index. This graph is given below:

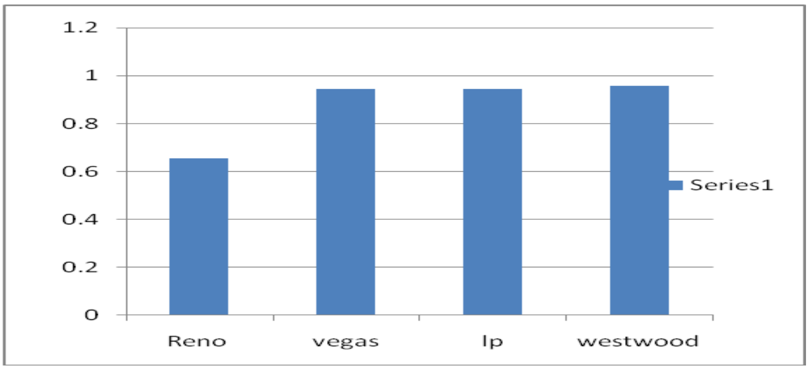

Figure 1.3

In this figure 1.3 it shows that fairness among the protocols in which the fairness of the TCP-Westwood is highly increased whether the fairness of the TCP-Reno is almost in decreasing order and the fairness of TCP-Vegas and TCP-LP is approximately similar but less than TCP-Reno.

VI. CONCLUSION

In this work, a comparative study on the performances of Congestion Control Protocols like TCP-Reno, TCP-Vegas, TCP-LP, and TCP-Westwood with various performance metrics can be evaluated. The performance of network can be improved by using different congestion control techniques in which some parameters are used that is RTT and Error Probability. In which there are three scenarios can be evaluated. In First scenario, various RTT (round trip time) is used to evaluate the result in which the performance of TCP-Vegas is approximately high. In Second scenario, various values of Error Probability are used to evaluate the result in which again the performance of TCP-Vegas is high. In Third scenario, Fairness can be evaluated among the protocols in which the performance of TCP-Westwood is high as compare to others. So overall conclusion is that the first two scenarios evaluated that the performance of TCP-Vegas is good and the last scenario evaluated that performance of TCP-Westwood is high in comparison.

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