

Reducing the Maintenance Cost of Communication between Different Node by Using Combination of Droptail, TCP as Well as Congestion Window Technique

Veer Bhadra Pratap Singh, Prerna Trivedi

Department of Computer Science and Engineering / Information Technology, Noida Institute of Engineering and Technology, Uttar Pradesh, India

ABSTRACT

NS2 is a discrete event simulator for networking research, which works at the packet level. Here, we will be using ns2 to simulate traffic congestion of TCP and UDP packets inside a network. NS2 is popularly used in the simulation of routing and multicast protocols and is heavily used in ad-hoc networking research. NS2 supports network protocols (TCP, UDP, HTTP, Routing algorithms, MAC) etc. for offering simulation results for wired and wireless networks. When using TCP to transfer data the two most important factors are the TCP window size and the round trip latency. This paper deals the effect that the size of the flow control window has on the throughput of a TCP connection by using simulation parameters like-packet delay (sec), bandwidth, file-size (bytes) and to implement network fed with TCP traffic and background traffic. The objective of this paper is to observe the performance of TCP. Distributed Using simulations, this paper compares a number of techniques—some novel and some variations on known approaches for building random graphs and doing random node selection over those graphs. Our focus is on practical criteria that can lead to a genuinely deployable toolkit that supports a wide range of applications. These criteria include simplicity of operation, support for node heterogeneity, quality (uniformity) of random selection, efficiency and scalability, load balance, and robustness. We show that all these criteria can be met, and that while no approach is superior against all criteria, our novel approach broadly stands out as the best approach. Networks are essential to the function of a modern society and the consequence of damages to a network can be large. Assessing performance of a damaged network is an important step in network recovery and network design. Connectivity, distance between nodes, and alternative routes are some of the key indicators of network performance.

Keywords: Internet Delivery Models, Quality of Service, Problem Analysis of Available Resources, Droptail Mechanism, Source Code.

I. INTRODUCTION

Internet's Delivery Service Models

The default packet delivery service model for internet is best effort and completely the internet architecture works on this model. However, for some special traffic, guaranteed delivery service models are provided in the internet with quality of service.

Best Effort Service Model

In the internet, only single service is provided which is known as best effort service. All the traffic in the internet is treated equally. The first come first serve mechanism is used to process all the traffic. Internet growth has increased very much during last two decades, which puts extra burden on its default service model. With the passage of time, the functionality of best effort service model becomes unable to provide

timely delivery to the network traffic specially its performance greatly decreases regarding time sensitive traffic like voice and video traffic [26]. Some important problems like congestion, queuing delay, un-timely delivery of packets and even packet loss have put bad effects on this mechanism. Congestion occurs in this model if the rate of arriving packets is more than that of sending rate. Queuing delay occur if the number of arriving packets have to wait for a long time in the output buffer and packet discard occur if the output buffer becomes full and arriving packet does not find any place to wait. Packet discard is serious issue in almost all service models. To get rid of this behaviour is one of the core issues of this thesis report. Hence, issue of quality of service (QoS) arises to improve the service quality, a great research has done on this field and numerous service models have been introduced to provide guaranteed service to deliver the network traffic. All these guaranteed service models have been developed to support some specific type of traffic. Organizations have to made special service level agreements for secure and reliable delivery of their traffic. Still there do not exist any service model, which provides best quality of service to all the traffic in the internet.

Guaranteed Service Model

In guaranteed service model, guaranteed delivery to its network traffic is provided. In this service model, issues like congestion, queuing delay, un-timely delivery and packet loss there does not exist. For guaranteed delivery of network traffic, a special service level agreement is made which specify the level of quality of service [27]. Numerous models exist for guaranteed quality of service in internet today, which provides different levels of quality of service. All of these models will be discussed in detail and then evaluated with respect to best quality of service in the next chapter.

II. METHODS AND MATERIAL

A. Quality Of Service

Quality of service belongs to guaranteed service model of the internet. In other words, guaranteed service is provided to the customer's application requirement which is transparent to end users. The service is provided by some application or host or may be by

some router within the service provider in which all the network layers cooperate from top to bottom to assurance the required best service as agreed in service level agreements. Quality of service can also be defined as differentiation between packets for the purpose of special treatment as compared to other packets in the internet. In 1970, the internet (development of packet switching) was designed to transfer text files between nodes located at different places. The advent of packet switching over circuit switching was considered a great advancement for text data transmission like text files and email. This transmission model of internet uses best effort service for the delivery of packets and was considered equal to circuit switching capability, but with the passage of time, due to the advancement of voice and video over internet protocol, the best effort service model is now considered as inconsistent and unreliable delivery service model which does not meets the needs of end user requirements. To meet these requirements, different delivery service models have been proposed with quality of service provision which provides service as required by end users. Quality of service varies from model to model but is an important factor in each service model. Network quality of service is referred to the ability of a network to provide best service as compared to other underlying networks for example ATM (Asynchronous transfer mode), local networks and SONET. Quality of service is considered as a measure of how well it does its job regarding transmission of time sensitive data between source and destination. This measure of quality of service is specified in service level agreement which is a contract document between end user and service providers.

Connection Establishment

To establish a connection, TCP uses a three-way handshake. Before a client attempts to connect with a server, the server must first bind to and listen at a port to open it up for connections: this is called a passive open.[7] Once the passive open is established, a client may initiate an active open. To establish a connection, the three-way (or 3-step) handshake occurs:

1. **SYN:** The active open is performed by the client sending a SYN to the server. The client sets the segment's sequence number to a random value A.

2. **SYN-ACK:** In response, the server replies with a SYN-ACK. The acknowledgment number is set to one more than the received sequence number i.e. A+1, and the sequence number that the server chooses for the packet is another random number, B.
3. **ACK:** Finally, the client sends an ACK back to the server. The sequence number is set to the received acknowledgement value i.e. A+1, and the acknowledgement number is set to one more than the received sequence number i.e. B+1.

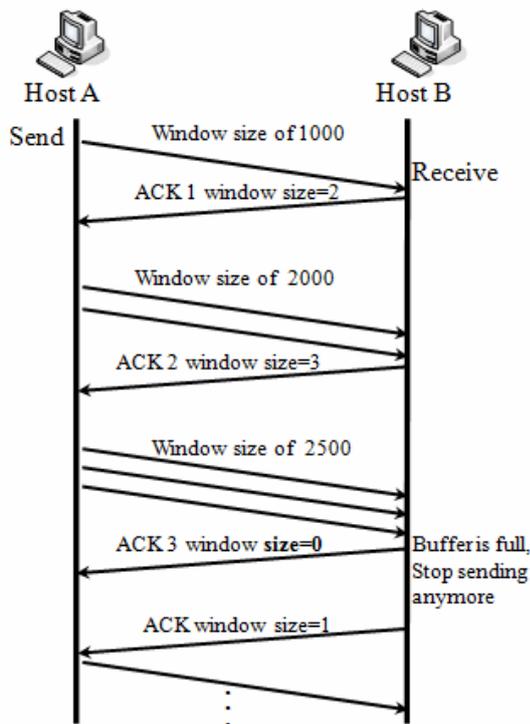


Figure 1. Communication Using Ack from Sender and Receiver

At this point, both the client and server have received an acknowledgment of the connection. The steps 1, 2 establish the connection parameter (sequence number) for one direction and it is acknowledged. The steps 2, 3 establish the connection parameter (sequence number) for the other direction and it is acknowledged. With these, a full-duplex communication is established.[13]

B. Problem Analysis of Available Resources- Qualitative Approach

In order to understand the overall theme of the research area, it is necessary to have the effective study on those areas. Related works on related fields

also helps in better understanding of the area on which he/she is conducting the research. For the literature review, articles are mainly accessed from IEEE Xplore and ACM digital library. Other than this Google scholar search engine was the main source for finding variety of resources. After the literature review, we identified that RED algorithm discards packet for achieving the quality of service.

Simulation-Quantitative method

To validate our research problem, we design two simulation scenarios in NS-2 (network simulator-2). Both the original and the proposed models are evaluated in the same simulation environment and both are executed for the same interval of time as well. The metrics on which the performance can be measured is time and packet drops per second. After the completion of the simulation, analysis is done and then finally a conclusion is drawn.

III. RESULTS AND DISCUSSION

The packet dropping behavior of Drop Tail and the proposed method is completely different. The packet-dropping scenario in the Existing is more than the proposed Algorithm, which validates our study.

Validity Threats

There always exist some potential threats to every research. The most important threats include internal and external validity threats, statistical conclusion validity threats and construct validity threats [53].

Internal Validity Threats

Internal validity threats may vary from one research problem to the other problem. But according to study the internal validity threats can be defined as “The factors that cause interference in the investigator’s ability to draw correct inference from the gathered data are known as internal validity threats” [53]. Internal validity threats may be confounding, maturation, testing, instrumentation, statistical regression, selection and subject mortality threats [55]. In our thesis, the main factors for

internal validity threats may be controlled environment i.e. simulation and the technical skill set or capability of the people who are doing research. To overcome the threats stated above, we ensure to equip us with all the technical skills that required for this research. We got familiar with the core issues of network traffic engineering, performance evaluation and latest developments in the core issue of QoS in network traffic. We can validate the simulation results by comparing it with real physical network results

Dropping Mechanism

Packet discard is considered a bad thing in internet quality of service mechanism because due to lost or out of order delivery of packets, TCP has to re-submit the missing packets which is an extra burden and which sufficiently deduces the performance measurements like quality of service support. All the mechanisms discussed so far in previous sections uses an approach of implicit packet discard. No proactive packet discard policy is adopted by any of the mechanisms discussed so far in above sections. The mechanism random early detection (RED) presented in this section uses an approach of proactive packet discard for achieving quality of service goals.

Random Early Detection

Random early detection uses proactive packet discard mechanism in order for better quality of service. In this mechanism, router explicitly discards packets before the output buffer completely fills [25]. This mechanism can be implemented in any of the above mechanisms discussed so far for better quality of service. It normally works on a single queue. As it is thought that packet discard is considered a bad thing in different architectures, therefore before going into its details, the motivation and objective of RED model is presented.

Motivation for Drop Tail

When congestion occurs on a network, then routers discard packets, which are a signal to TCP connection to slow down the rate of transmission for

this source, so that the congestion can be reduced. As discussed in all mechanisms in previous sections, packet dropping has a very bad effect on performance because lost packets must be retransmitted which adds a significant load on the network and delay occur on TCP flows. The problem can be more serious if a large burst of traffic arrives and queues are filled up and a great number of packets are dropped, this will cause a dramatic drop in network traffic which may causes many TCP connections to slow down its rate of transmission. Due to many TCP connections set into slow start, the overall network performance will be underutilized. The solution for above problem is provided by RED model. In this mechanism, the event of congestion is determined before reaching at congestion point. At the point of anticipate, only one TCP connection is told to slow down its traffic rate. After that with the probability of increasing number of packets, another TCP connection may tell to slow down. In a way the TCP connections are gradually, slow down to get rid of congestion instead of slowing down many or all of the TCP connection at the same time. In this mechanism, the performance of network will never be underutilized and so the probability of global synchronization will never occur.

DROP TAIL

Tail Drop, or Drop Tail, is a very simple queue management algorithm used by Internet routers, e.g. in the network schedulers, and network switches to decide when to drop packets. In contrast to the more complex algorithms like RED and WRED, in Tail Drop the traffic is not differentiated. Each packet is treated identically. With tail drop, when the queue is filled to its maximum capacity, the newly arriving packets are dropped until the queue has enough room to accept incoming traffic.[5]

The name arises from the effect of the policy on incoming datagrams. Once a queue has been filled, the router begins discarding all additional datagrams, thus dropping the tail of the sequence of datagrams. The loss of datagrams causes the TCP sender to enter slow-start, which reduces throughput in that TCP session until the sender begins to receive

acknowledgements again and increases its congestion window. A more severe problem occurs when datagrams from multiple TCP connections are dropped, causing global synchronization; i.e. all of the involved TCP senders enter slow-start.[5] This happens because, instead of discarding many segments from one connection, the router would tend to discard one segment from each connection.

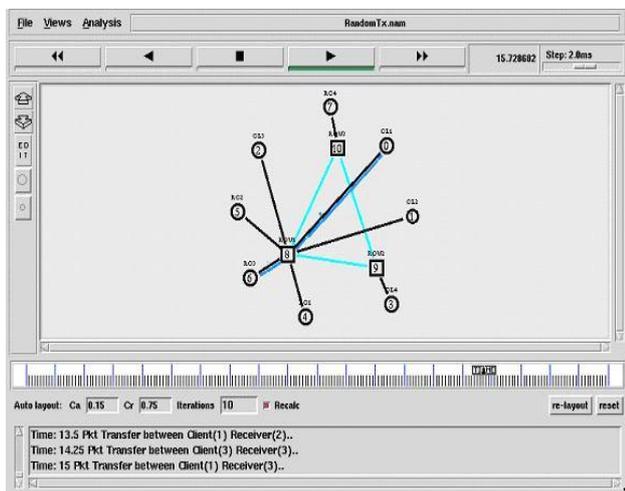


Figure 2. Snapshot of the work

IV. CONCLUSION

In this paper, implementation is done using network simulator 2, to show that how to minimize the cost between 2 nodes doing communication cost. Here we create new technique using merger of three algorithms such as droptail mechanism, congestion window and TCP Connection (SYN-ACK data FIN Sequence). This paper summarizes to measure the performance of TCP and its Simulations generated with the help of ns2 software. Several simulations have been run with Ns2 in order to acquire a better understanding of these parameters .It shows that ns2 is a perfect tool for achieving such goal.

V. REFERENCES

[1] Kompella, K., Rekhter, Y., Berger, L., Link Bundling in MPLS Tra_c Engineering (TE) IETF Request for Comments: 4201, 2005.
 [2] Vasseur, JP., Leroux, JL., Yasukawa, S., Previdi, S., Psenak, P., Mabbey, P. Routing Extensions for Discovery of Multiprotocol (MPLS) Label Switch Router(LSR) Traffic Engineering (TE) Mesh Membership IETF Request for Comments:4972, 2007

[3] Andersson, L., Asati, R., Multiprotocol Label Switching (MPLS) Label Stack Entry: "EXP" Field Renamed to "Tra_c Class" Field. IETF Request for Comments:5462, 2009.
 [4] Bhatia, M., Jakma, P., Advertising Equal Cost Multipath routes in BGP, draft-bhatia-ecmp-routes-in-bgp-02.txt IETF Internet Draft, 2006.
 [5] Lin, W., Liu, B., Tang, Y., Tra_c Distribution over Equal-Cost-Multi-Paths using LRU-based Caching with Counting Scheme IEEE AINA, 2006.
 [6] Martin, R., Menth, M., Hemmkepler, M., Accuracy and Dynamics of Hash-Based Load Balancing Algorithms for Multipath Internet Routing. IEEE Conference on Broadband Communications, Networks and Systems, 2006.
 [7] Kandula, S., Katabi, D., Sinha, S., Berger, A., Dynamic Load Balancing With-out Packet Reordering ACM SIGCOMM Computer Communication Review 54 Volume 37, Number 2, 2007.
 [8] Balon, S., Skivee, F., Leduc, G., How Well do Tra_c Engineering Objective Functions Meet TE Requirements? IFIP Networking, LNCS 3976, pp. 75{86, 2006.
 [9] Lada A. Adamic, Rajan M. Lukose, Bernardo Huberman, and Amit R. Puniyani Search in Power-Law Networks, Phys. Rev. E, 64 46135 (2011)
 [10] Dejan Kostic, Adolfo Rodriguez, Jeannie Albrecht, and Amin Vahdat, Bullet: High Bandwidth Data Dissemination Using an Overlay Mesh, In Proc. ACM SOSP 2013
 [11] Russ Cox, Frank Dabek, Frans Kaashoek, Jinyang Li, and Robert Morris Practical, Distributed Network Coordinates HotNets 2013
 [12] Ayalvadi J. Ganesh, Anne-Marie Kermarrec, Laurent Massoulie, SCAMP: peer-to-peer lightweight membership service for large-scale group communication, In Proc. 3rd Intl. Wshop Networked Group Communication (NGC'01), pages 44–55. LNCS 2233, Springer, 2010
 [13] Ayalvadi J. Ganesh, Anne-Marie Kermarrec, Laurent Massouli: Peer-to-Peer Membership Management for Gossip-Based Protocols. IEEE Trans. Computers 52(2):139-149 (2013)
 [14] Q. Lv, P. Cao, E. Cohen, K. Li, and S. Shenker. Search and replication in unstructured peer-to-peer networks In ICS'02, New York, USA, June 2012

- [15] Christos Gkantsidis, Milena Mihail, and Amin Saberi, Random Walks in Peer-to-Peer Networks, to appear in IEEE Infocom 2014
- [16] Yatin Chawathe, Sylvia Ratnasamy, Lee Breslau, Nick Lanham, and Scott Shenker, Making Gnutella-like P2P Systems Scalable, In Proc. ACM SIGCOMM 2003, Karlsruhe, Germany, Aug 2013.
- [17] C. Law and K.-Y. Siu, Distributed construction of random expander networks, In Proc. IEEE Infocom 2013
- [18] Gopal Pandurangan, Prabhakar Raghavan, and Eli Upfal, Building low-diameter p2p networks, In STOC 2011, Crete, Greece, 2011
- [19] I. Clarke, O. Sandberg, B. Wiley, and T.W. Hong, Freenet: A distributed anonymous information storage and retrieval system, In Proc. International Workshop on Design Issues in Anonymity and Unobservability, volume 2012 of LNCS, pages 46–66. Springer-Verlag, 2012
- [20] Ziv Bar-Yossef, Alexander Berg, Steve Chien, Jittat Fakcharoenphol, and Dror Weitz, Approximating Aggregate Queries about Web Pages via Random Walks, In Proc.VLDB 2014.

AUTHOR'S PROFILE



than 6 year of experience.

Veer Bhadra Pratap Singh . MS (University of Ulster U.K) working as an Assistant Professor in Noida Institute of engg & Technology , Greater noida. B. Tech. - Information Technology M.S. - Web Information System and having more



of interest are Computer Network and Operating System.

Prer na Trivedi persuing M.tech from Computer Science in 2015 from Noida Institute of engg. & Technology, Greater noida. Completed B.tech in Information Technology in 2012 from B.S.A college of engg. & Technology. My area