

Study on the Behavior of Packet Loss Based on CBR Packet Size and Intervals in Wired Networks

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Abstract

In the current age of high speed networks, it is increasingly important to improve the congestion control keeping in mind the buffer size and the number of packets to be sent. Since the CBR (constant bit rate) is the maximum bit rate that matters, not the average, so CBR would be used to take advantage of all of the capacity. This paper presents a simulation based analysis of the behaviour of packet loss based on CBR packet size and intervals in wired networks and comparison in the number of packet loss when both the factors are changed.

Keyword

Tcp, Udp, Cbr, Red Queue, Packet-Switched Computer.

I. Introduction

A. Wired Networks vs. Wireless Networks

In computing terminology, the term “wired” is used to differentiate between wireless connections and those that involve cables. While wireless devices communicate over the air, a wired setup uses physical cables to transfer data between different devices and computer systems.

A wired network is a common type of wired configuration[1]. Most wired networks use Ethernet cables to transfer data between connected PCs. In a small wired network, a single router may be used to connect all the computers. Larger networks often involve multiple routers or switches that connect to each other. One of these devices typically connects to a cable modem, T1 line, or other type of Internet connection that provides Internet access to all devices connected to the network.

Wired may refer to peripheral devices as well. Since many keyboards and mice are now wireless, “wired” is often used to describe input devices that connect to a USB port. Peripherals such as monitors and external hard drives also use cables, but they are rarely called wired devices since wireless options are generally not available.

While many peripherals are now wireless, some users still prefer wired devices, since they have a few benefits over their wireless counterparts. For example, an Ethernet connection is not prone to signal interference that can slow down Wi-Fi connections. Additionally, wired network connections are generally faster than wireless ones, which allows for faster data transfer rates.

B. Congestive Collapse

Congestive collapse (or congestion collapse) is a condition that a packet-switched computer network can reach, when little or no useful communication is happening due to congestion. Congestion collapse generally occurs at “choke points” in the network, where the total incoming traffic to a node exceeds the outgoing bandwidth[2]. Connection points between a local area network and a wide area network are the most likely choke points. When a network is in such a condition, it has settled (under overload) into a stable state where traffic demand is high but little useful throughput is available, and there are high levels of packet delay and loss (caused by routers discarding packets because their output queues are too full) and general quality of service is extremely poor.

When more packets were sent than could be handled by intermediate routers, the intermediate routers discarded many

packets, expecting the end points of the network to retransmit the information. However, early TCP implementations had very bad retransmission behavior. When this packet loss occurred, the end points sent extra packets that repeated the information lost, doubling the data rate sent, exactly the opposite of what should be done during congestion. This pushed the entire network into a ‘congestion collapse’ where most packets were lost and the resultant throughput was negligible.

C. Congestion Control

Congestion control concerns controlling traffic entry into a telecommunications network, so as to avoid congestive collapse by attempting to avoid oversubscription of any of the processing or link capabilities of the intermediate nodes and networks and taking resource reducing steps, such as reducing the rate of sending packets. It should not be confused with flow control, which prevents the sender from overwhelming the receiver.

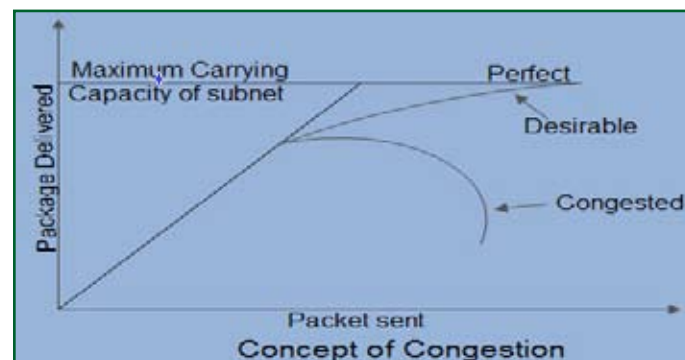


Fig. 1: Concept of Congestion Control

D. Causes of Congestion

The various causes of congestion in a subnet are[3]:

- The input traffic rate exceeds the capacity of the output lines. If suddenly, a stream of packet start arriving on three or four input lines and all need the same output line. In this case, a queue will be built up. If there is insufficient memory to hold all the packets, the packet will be lost. Increasing the memory to unlimited size does not solve the problem. This is because, by the time packets reach front of the queue, they have already timed out (as they waited the queue). When timer goes off source transmits duplicate packet that are also added to the queue. Thus same packets are added again and again, increasing the load all the way to the destination.
- The routers are too slow to perform bookkeeping tasks

(queuing buffers, updating tables, etc.).

- The router's buffer is too limited.
- Congestion in a subnet can occur if the processors are slow. Slow speed CPU at routers will perform the routine tasks such as queuing buffers, updating table slowly. As a result of this, queues are built up even though there is excess line capacity.
- Congestion is also caused by slow links. This problem will be solved when high speed links are used. But it is not always the case. Sometimes increase in link bandwidth can further deteriorate the congestion problem as higher speed links may make the network more unbalanced. Congestion can make itself worse. If a route does not have free buffers, it start ignoring/discarding the newly arriving packets. When these packets are discarded, the sender may retransmit them after the timer goes off. Such packets are transmitted by the sender again and again until the source gets the acknowledgement of these packets. Therefore multiple transmissions of packets will force the congestion to take place at the sending end.

II. Implementation Details

A. Performance evaluation metrics

The scope of this project is to simulate the performance of wired networks based on changing the CBR packet size and intervals. Different performance evaluation Metrics are considered to compare the performance of TCP and UDP such as[4]:

- Throughput
- Packet Loss Rate
- End to End Delay
- Bandwidth Utilization

The productivity of the network is based on its throughput. Network throughput is the average rate of successful message delivery over a communication channel and the amount of traffic that a network can handle. The throughput is measured in bits per second (bps). The efficiency of an algorithm is measured in terms of the packets lost. Packet loss rate is measured as the percentage of the total number of dropped packets by the total packets transmitted over the link. The responsiveness of the network is limited by its delay. End-to-end delay refers to the time taken for a packet. Constant Bit Rate(CBR) is a term used in telecommunications, relating to the quality of service. When referring to codecs, constant bit rate encoding means that the rate at which a codec's output data should be consumed is constant. CBR is useful for streaming multimedia content on limited capacity channels since it is the maximum bit rate that matters, not the average, so CBR would be used to take advantage of all of the capacity.

B. Techniques for Performance Evaluation

There are three techniques for performance evaluation which are analytical modeling, simulation and measurement. Simulation had being chosen because it is the most suitable technique to get more details that can be incorporated and less assumption is required compared to analytical modeling. Accuracy, times available for evaluation and cost allocated for the thesis are also another reason why simulation is chosen. By using simulation, researchers should be allowed to study a system in well-known conditions, repeatedly if necessary in order to understand events[5].

C. Network Simulation

Network simulation is a technique where a program models

the behaviour of a network either by calculating the interaction between the different network entities (hosts/packets, etc.) using mathematical formulas, or actually capturing and playing back observations from a production network[6]. The behavior of the network and the various applications and services it supports can then be observed in a test lab; various attributes of the environment can also be modified in a controlled manner to assess how the network would behave under different conditions.

III. Simulation Setup

The simulations in this project are implemented on the Network Simulator tool (NS-2.35) in the Ubuntu 14.04 LTS platform. The simulated network consists of clients that are connected to a server through 2 routers. The clients are connected to the routers via a 1Mbps duplex link incorporating a delay of 10ms. The router and server are linked with a 1Mbps duplex link having a 10ms delay. The RED queue algorithm has been used for this project. The packet size and the CBR rate is changed by discrete amounts to observe the changes in the packet loss.

IV. Simulation Results

Based on the simulation on the above architecture, the packet loss is measured by changing the packet size and CBR rate intervals. The observation is noted in the table below:

Table 1: Comparison Of Packet Size And Cbr Intervals To Measure The Rate Of Packet Loss In Wired Network.

Packet size (in MB)	CBR interval (Sec)	Total Packet loss KB
1800	0.03	6960
1800	0.04	5328
1800	0.1	1725
2000	0.03	7242
2000	0.04	5244
2000	0.1	1641
1200	0.03	5550
1200	0.04	3513
1200	0.1	639

From the above table, it can be observed that the packet loss can be measured in terms of CBR intervals and packet size. As the packet size increases, the probability of packet loss also increases for a certain fixed CBR interval. But if the CBR interval is also increased, then the total packet loss decreases.

V. Conclusion

Here, we have compared both the packet size and CBR interval by discrete amounts to observe all the possible behaviour of the packet loss for a wired network. From the table, we can draw the conclusion that if CBR intervals are increased, the congestion can be reduced for large packet sizes as the packet sizes imply serious consequences on the congestion control for wired networks.

References

- [1]. The Tech Terms website. [Online]. Available: <http://techterms.com/definition/wired>
- [2]. The Wikipedia website. [Online]. Available: https://en.wikipedia.org/wiki/Network_congestion
- [3]. The Computer Notes website. [Online]. Available: <http://ecomputernotes.com/computernetworkingnotes/>

communication-networks/what-is-congestion-control-describe-the-congestion-control-algorithm-commonly-used

- [4]. ManasPratimSarma, "Performance Measurement of TCP and UDP Using Different Queuing Algorithm in High Speed Local Area Network", *International Journal of Future Computer and Communication*, Vol. 2, No. 6, December 2013.
- [5]. Vijayalakshmi M., Avinash Patel, "Qos Parameter Analysis on AODV and DSDV Protocols in a Wireless Network", *Vijayalakshmi M. et. al. / Indian Journal of Computer Science and Engineering Vol. 1 No. 4* 283-294
- [6]. Nitesh Kumar Singh, "Simulation of Network Traffic Based on Queuing Theory using Opnet", *Master of Engineering thesis, Thapar University, 2010*