Congestion and Flow Control in Homogeneous and Heterogeneous Networks: Discrete Event Simulation Model

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Abstract

This Paper provides a study and review of existing congestion control algorithms in various types of homogenous and heterogeneous networks. To test the new methodology for the congestion and flow control we have used the two different discrete event simulation tools. The OPNET simulation tool is used to find out performance of the TCP reliable protocol for built-in congestion control mechanism. The second part is for simulation of existing congestion control algorithm (AIMD-Additive Increase Multiplicative Decrease) and new suggested congestion and flow control algorithm using OMNeT++ discrete event simulation.

Keywords: AIMD, discrete event simulation, advertise window, congestion window.

1. INTRODUCTION

Congestion control is the efforts made by network nodes to prevent or respond to overload conditions. We use concept of 'fairness' i.e. try to share pain among the all users, rather than causing great pain to a few. Many congestion control mechanism having built in notion of resource allocation. Flow control is keeping fast sender form overrunning a slow receiver. Congestion control keeps a set of senders form sending too much data into network because of lack of resources at some point. In other words the bandwidth is also known as throughput. The term throughput is used to measure the performance of the system.

Congestion control and resource allocation involves both host and network elements such as routers. Congestion and resource allocation are two sides of same coin involves host and network elements such as routers. The queuing discipline can segregate traffic. As congestion and resource allocation is not isolated to single level of protocol hierarchy. Resource allocation is process by which network elements try to meet the competing demands that application have for network resources. Resources are link bandwidth and buffer space in routers or switches. The problem is for refusing the users.

The congestion controlling algorithms are categorized into "box is black", "box is gray" and "box is green". This type of categorization is done on the basis of knowledge of it's state. In case "box is black" category no knowledge of it's state , other than binary feedback upon congestion. The algorithms AIMD-FC, Binomial mechanism, SIMD, HIGHSPEED-TCP, BIC-TCP and other generations of AIMD are included in this first category. The "Box is Gray" category use measurements to estimate available bandwidth, level of contention or even the temporary characteristics of congestion. Due to possibility of wrong estimations and measurements, the network is considered a gray box. The TCP-VEGAS, FAST-TCP, TCP-REAL, TCP-WESTWOOD [13][16], TFRC, TCP-JERSEY are some the algorithms. The "Box is Green" consist of bimodal congestion control, which calculates explicitly the fair-share, as well as the network assisted control, where the network communicates it's state to the transport layer. The 'green' category includes VCP, XCP and JETMAX approaches to control the congestion [15].

TCP is always under research for the checking the performance of the network under satellite networks the work is done by performance analysis is done for "TCP Spoofing" using ns-2. The simulation is to analyze spoofing over a large range of file sizes and under various congested conditions, while prior work on this topic has primarily focused on bulk transfers with no congestion.[3] The investigation regarding the fundamental problem of achieving the system optimal rates, which maximize the total user utility, in a distributed network environment using only the information available at the end hosts. This is done by decomposing the overall system problem into sub problems for the network and for the individual users and introducing an incentive-compatible pricing scheme. This is done using a window based algorithm to provide an algorithm for the network to adjust its prices and the users to adjust their window sizes such that at an equilibrium the system optimum is achieved. [4]. The work regarding uses emulations to explore the benefits of adding selective acknowledgments (SACK) and selective repeat to TCP. Comparison of Tahoe and Reno TCP, the two most common reference implementations for TCP, with two modified versions of Reno TCP[5]. The variable structure congestion control(VCP) protocol is developed from basic TCP[6]. TCP New Jersey, which is capable of distinguishing wireless packet losses from congestion packet losses, and reacting accordingly. TCP New Jersey consists of two key components, the timestamp-based available bandwidth estimation (TABE) algorithm and the congestion warning (CW) router configuration. TABE is a TCP-sender-side algorithm that continuously estimates the bandwidth available to the connection and guides the sender to adjust its transmission rate when the network becomes congested.[7]

The first implementation of the MaxNet TCP network congestion control protocol. MaxNet uses explicit multi-bit signaling from routers to achieve high throughput and low latency over networks of arbitrary capacity and topology, and virtually any delay. The MaxNet algorithm is extended in this paper to give both provable stability and rate fairness. The implementation is based on the Linux Traffic Control framework. The system consists of a sender and receiver TCP algorithm as well as a router module[8]. In heterogeneous networks, TCP connections that incorporate a terrestrial or satellite radio link are greatly disadvantaged with respect to entirely wired connections, because of their longer round trip times (RTTs). To cope with this problem, a new TCP proposal, the TCP Hybla, is presented and discussed in the paper[9]. The work on retransmission ambiguity is done in some papers for RTT.[10]. The research related with wireless frame traffic is studied in various ways [12]. The physical flow-based congestion management allocation mechanism for multiple transaction networks is given to characterize the transmission congestion [17] . The transient behaviors of TCP friendly Congestion control protocol is analyzed with analytically as well as the simulated environments given in GAIMD, TFRC and TEAR[18].

The sender-based approach for multicast congestion control targeted towards reliable bulk data transfer is done by using ns-2 simulator for finding dynamically worst congested path in multicast tree in TCP friendly environment[19]. Determine near-optimal policies when the available bandwidth is unchanging, and near-optimal competitive policies when the available bandwidth is changing in a restricted manner under the control of an adversary. The focus on regulating the rate of a single uni-cast flow when the bandwidth available to it is unknown and may change over time.[20]. The approach in maintaining Quality of Service(QoS) adopting equation-based congestion control (EBRC) for differentiated services instead of traditional TCP congestion control for best-effort service. The congestion proactive Sender functionality, packet loss control mechanism in routers and receiver functionality are differentiated and implemented in some approaches[21]. The uniform solution accommodating both responsive and unresponsive traffic with trivial overhead by ignorance of rate adaptation is found in a novel RED-based hop-by-hop congestion control mechanism which is based on the coordination of the routers and the hosts. No per-flow state information is maintained in the routers [22].The theory of window based unicast congestion control and use of fairness and efficiency found in the GAIMD and TCP [23].

The various approaches of the congestion control and flow control in the network environment are done as work of the applying different counter measures across the various criteria. TCP implements a highly tuned congestion control mechanism. The OPNET is major tool used for the network performance optimization [11]. The counter measures applied are making stop mechanism to the senders so that avoid the overloading of network. The available bandwidth estimation is again one of the factor for the detecting how many packets it can safely transit. It maintains the state variable for each connection, called the congestion window, which is used by the source to limit how much data it is allowed to have in transit at a given time.

The various rising numbers are indication of the congestion is more in the network. So if we are able to reduce the statistics of these factors, we are able to control the congestion. Some of the factors such as percentage of all packets discarded for lack of buffer space,[1] average queue length, number of packets that time out and retransmitted, average packet delay, and standard deviation of packet delay. The existing algorithms are made to reduce this statistics across the various levels.

2. BACKGROUND

TCP interprets timeouts as sign of congestion. It maintains state variable for each connection, called the congestion window, which is used by the source to limit how much data it is allowed to have in transit at a given time. TCP uses mechanism, called Additive Increase Multiplicative Decrease (AIMD), that decreases the congestion window when the level of congestion goes up and increase the congestion window when the level of congestion goes down. Each time a timeout occurs, the source sets the congestion window to half of the previous value. This halving corresponds to the multiplicative decrease part of the mechanism.[2]

The congestion window is not allowed to fall below the size of the packet i.e. the TCP maximum segment size (MSS). This halving corresponds to the multiplicative decrease part of the mechanism. Each time the source successfully sends a congestion window's worth of packets, it adds the equivalent of one packet to the congestion window; this is additive increase part of the mechanism. TCP uses a mechanism called slow start to increase the congestion window "rapidly" from a cold start in TCP connections. It increases the congestion window exponentially, rather than linearly. Another one mechanism TCP utilizes is fast retransmit and fast recovery. Fast retransmit is heuristic that sometimes triggers the transmission of dropped packet sooner than regular timeout mechanism. The end-to end transmission protocol is used analyze the size to the congestion window with different mechanism. The inbuilt TCP have a TCP congestion control is for each source to determine how much capacity is available in the network, so that it knows how many packets it can safety have in transit.

3. METHODOLOGY

Discrete-event-simulation concerns the modeling of system as it evolves over time by representation in which the state variables change instantaneously at separate points in time. In mathematical terms, we might say that the system can change at only a countable number of points in time. These points in time are the ones at which an event occurs, where an event defined as an instantaneous occurrences that may change the state of the system. Although discrete event simulation could conceptually be done by hand calculations, the amount of data that must be stored and manipulated for most real-world systems dictates that discrete event simulation can be done by digital computers[14]. The same theme of DES is represented in figure 3 The AIMD algorithms with basic formulae are represented in this section. Effective Resource allocation is to maximize power given by.

Power= Throughput / Delay ……………………………………………………………….(1)

Delay= (Depart Time – Arrival time) +Transmission Time + Latency……………………(2)

Throughput = Packets per second X Bits per Packets……………………………………(3)

Latency =Propagation + Transmit + queue………………………………………………..(4)

Propagation = Distance / Speed of Light…………………………………………………..(5)

Transmit = Size/ Bandwidth…………………………………………………………………(6)

Throughput=Transfer size/ Transfer Time…………………………………………………(7)

Transfer Time= RTT + (1 / Bandwidth) X Transfer Size……………………………………..(8)

Latency is the time required for a packet to traverse the network from source to destination. Our evaluations of latency are on zero-load latency of the network. This ignores latency due to contention with other packets over shared resources. If simulation includes contention latency, latency becomes a function of offered traffic and it becomes instructive to plot this function. Bandwidth is measure of width of frequency band. It is measured in Hertz.(range of signals that can be accommodated. If we talk about the width of a communication link, we refer to the number of bits per second that can be transmitted on the link. In TCP congestion control mechanism each source is determines how much capacity is available in the network, to transmit the number of packets. The Additive Increase and Multiplicative Decrease (AIMD) decreases window size when congestion level goes to high.
The simulation model here developed in OMNeT++. The model developed for calculating the "advertise window" for flow control and Congestion Window for AIMD. The various factors with small modification in the basic algorithm are implemented for further calculation of throughput and delay (latency).

4. EXPERIMENTAL ANALYSIS AND RESULTS

The simple model for flow control is created by using OMNeT++ discrete event simulation modeling. The events are processing with respect to the packet or message arrived at any module. The idea is to increase the Advertise window size (by some factor) if acknowledgement of the packet is received. The sink sends the Advertise window size to the source. The model is as show in figure 1. The same Advertise Window is considered as the main factor in Congestion control to increase it for the multiple sources environment. The new term introduced is "Congestion Window".

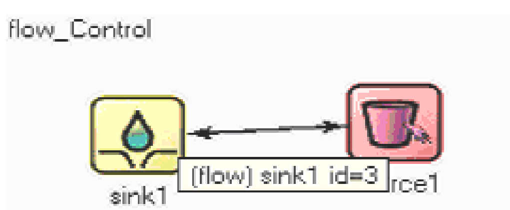

FIGURE 1: Simple Model for Flow Control Using OMNeT++

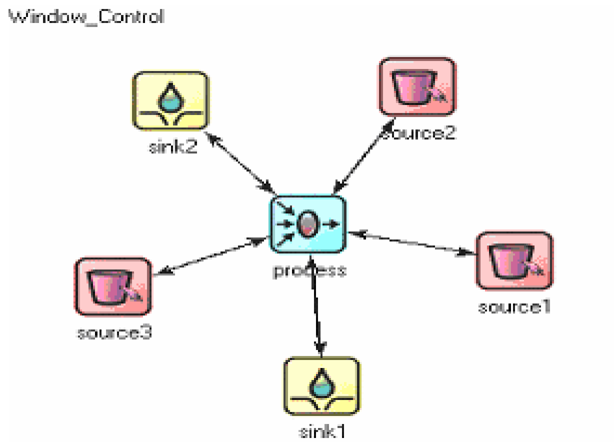

FIGURE 2: Simple Model for AIMD Congestion Control Using OMNeT++

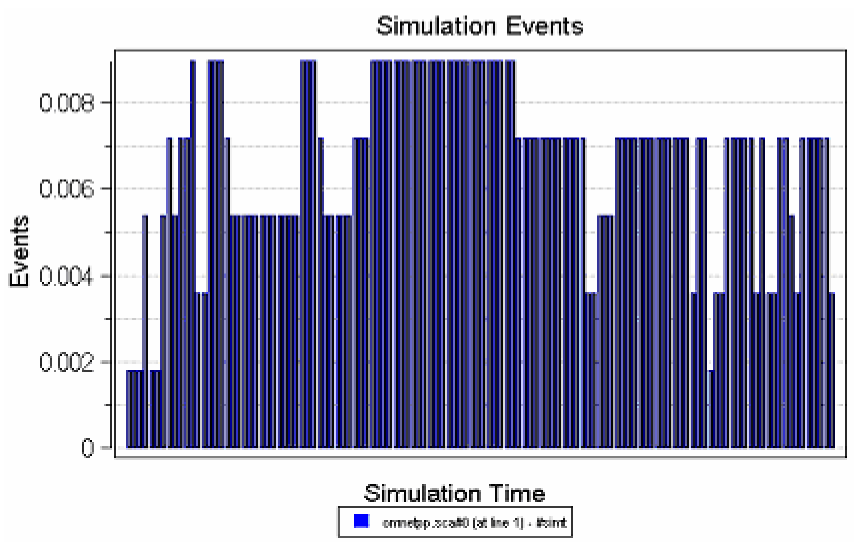

Figure 3: Simulation Time in Model with Three Sources and Two sinks

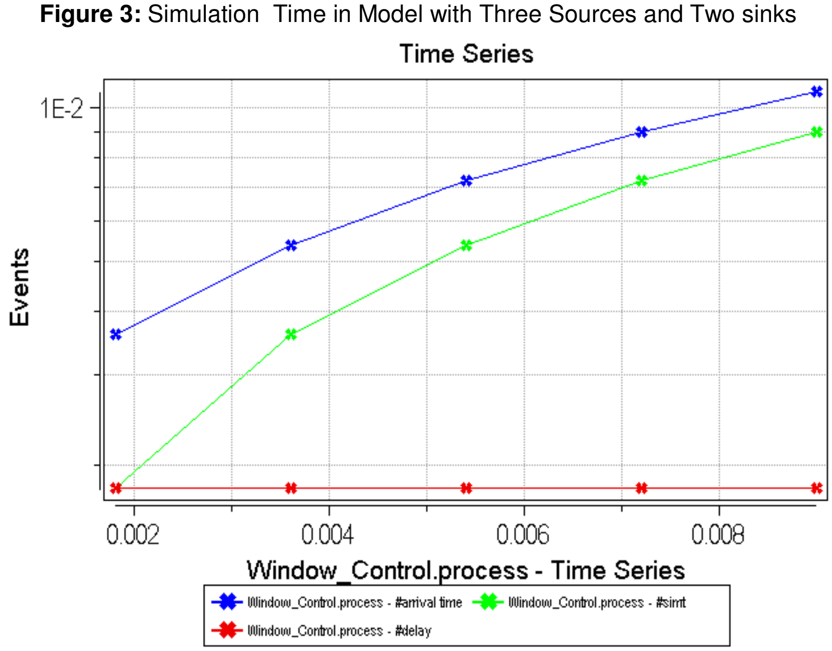

Figure 4: The time series for “Process” module.

The figure 3 explains the simulation events happened in the timeline chart. The Y-axis shows the increasing events with respective to the x axis as the events times. The results we observed are show in graph in figure 3 and figure 4. The OMNeT++ is simulation programming tool. We tested the results in OPNET. The results for the TCP congestion window size is shown in figure 5, 6, and 7 . The traffic is considered as the multi-scale integrated traffic. The nature of algorithm is tested and graphs plotted to check across the results with OMNeT++ Simulator.

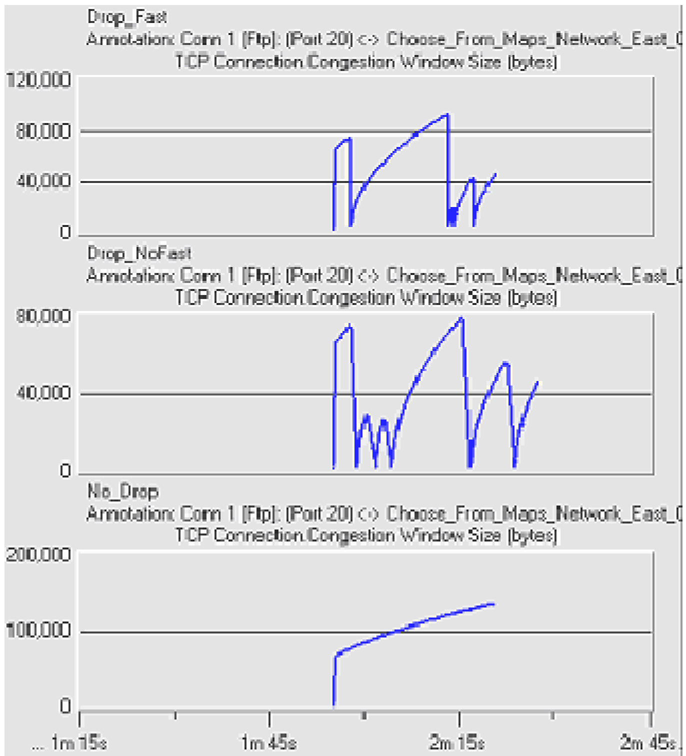

Figure5: Size of window with Drop fast, drop no fast and no drop scenario

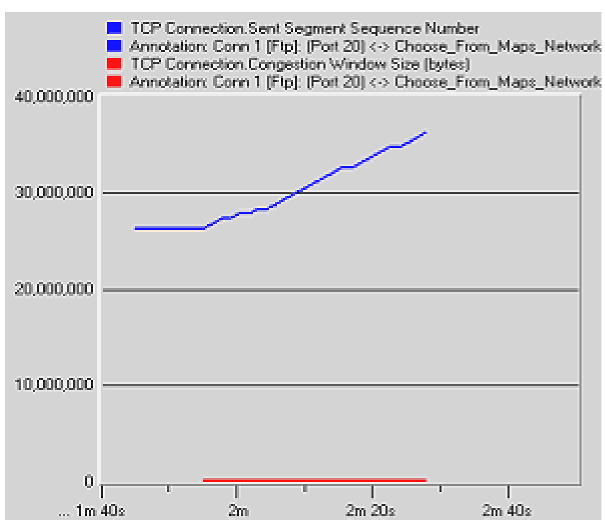

FIGURE 6: Sent Segment Sequence Number in TCP.

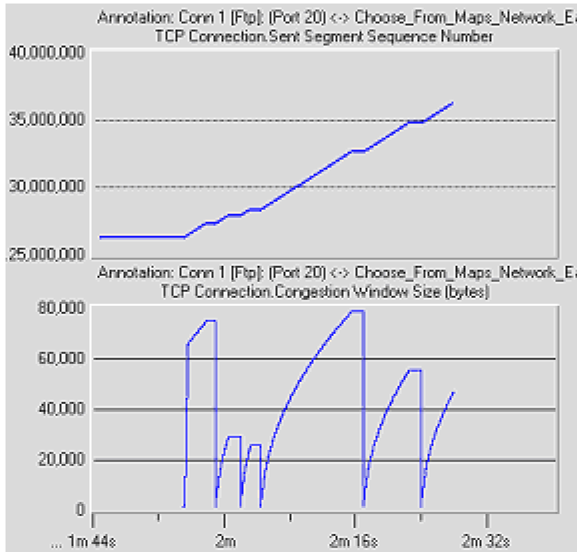

FIGURE 7: Sent Segment Sequence Number and Congestion Window Size in TCP Detailed View

5. CONCLUSION & FUTURE WORK

The AIMD is increases the window size as the acknowledgement is received for every packet sent by the source. But missed packet acknowledgement may reduces the Congestion Window size and performance decreases. The mechanism to raise the congestion window for slow raising is good idea to improve the performance of the AIMD. Plotting graphs of segment sequence number remain unchanged with every drop in the congestion window. The drop-no-fast scenario have slowest growth in sequence numbers as compared with no-drop and drop-fast scenario. We have given the comparison of the simulations across two different simulating environment is cross checking mechanism. As shown in figure 4 the OMNeT++ simulation with our own idea of slowly increase can gives resulting scene of the throughput over the figure number 6 and 7 of three different scenarios of the TCP.

Future Work: Calculating a fairness index for set of outputs to find out resources allocation in the network. Fairness index can be calculated as. Given a set of flow throughputs (x1,x2,x3…xn) (measured in consistent units such as bits/second).[paterson]

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