International Journal of Basic Sciences and Applied Computing (IJBSAC)
ISSN: 2394-367X, Volume-1 Issue-6, May 2015

Expedite Flow Completion on High Speed Network Through Protocols

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Abstract- It has been proved by a lot of researchers that the present operation of TCP which is the main internet control protocol will suffer poor performance in future high speed networks. It has also been established that performance issues are very crucial in computer networks, for example when many computers are interconnected, complex interactions arise with unforeseen consequences. This complexity leads to degradation of performance if the system is not managed properly. Yet research on congestion control focuses almost entirely on maximizing link throughput, utilization and fairness, which matter more to the operator than the user. To arrest the situation, various factors which affect network performance were examined. Characteristics of congestion Control Protocols were described. Congestion Control Protocols like Transmission Control Protocol (TCP) and Explicit Congestion Protocol (XCP)were evaluated. The proposed congestion control protocol, Rate Congestion Protocol (RCP) was also evaluated. Then NS2 simulator was used under different scenarios to evaluate the performance of RCP and the aforementioned protocols to prove that RCP outperforms them in terms of expediting flows.

Keywords: Rate Control Protocol (RCP); Explicit Control Protocol (XCP); Processor Sharing (PS); Network Simulator 2(NS2); Transmission Control Protocol (TCP).

I. INTRODUCTION

This study is meant to address congestion and also increase the rate of flow of traffic in computer networks which leads to performance related issues in most organisations in general and the Internet in particular. Currently, the Transmission Control Protocol, or TCP, is the most widely used congestion control mechanism. TCP fulfills two significant functions. The first entails a reliable and in order delivery of bytes to the higher application layer. It builds on the unreliable, connectionless IP service, providing a service that is reliable by transmitting lost or corrupted data until the data is successfully received at the destination. It also delivers bytes in order (reorders out-of-order data and eliminates duplicates before delivering to the application process), multiplexes and de-multiplexes traffic from different processes on an end-host, and performs flow control (prevents a sender from overwhelming a receiver by specifying a limit on the amount of data that can be sent). TCP’s second function is to perform congestion control and protect the network from a congestive collapse. We briefly describe TCP’s congestion control mechanisms below. TCP uses adaptive congestion control mechanisms that react to congestion events (such as packet loss or delay) by limiting the sender’s transmission rate.

Manuscript Received on May 07, 2015.

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These mechanisms allow TCP to adapt to heterogeneous network environments and varying traffic conditions, and keep the Internet from severe congestion events. TCP congestion control works on an end-to-end basis, where each connection, before starting, begins with a question: At what rate should the data be sent for the current network path? It does not receive an explicit answer for this question, but each connection determines the sending rate by probing the network path and modulating its rate based on perceived congestion, through packet-loss and delay. The connection rate is proportional to TCP’s sliding window (swnd is the limit on the amount of outstanding data in flight), which is set as the minimum of the receiver advertised window (rwnd) and of the congestion window (cwnd changes dynamically based on feedback of network conditions).

To arrest the situation, Flow and network level properties were examined and implemented.

1. Processor Sharing: We would like to share the bottleneck link equally among competing flows. Emulating Processor Sharing (PS) is a simple way to do this. Processor Sharing is a worthwhile goal to achieve: Even if its mean flow completion time is not quite the minimum achievable, it comes reasonably close to the minimum, and so flows complete quickly, often an order of magnitude quicker than in TCP for typical Internet size flows. Furthermore, its mean completion time is invariant of flow size distribution for a single bottleneck. Even when flow completion time does not make sense (e.g., in long-lived bulk transfers), Processor Sharing results in flows getting high throughput and fair sharing of the bottleneck bandwidth.
2. Stability: Networks occasionally experience sudden large traffic surges (e.g., flash-crowds). We want the network to come back to a stable operating behavior quickly after such disruptions.
3. Queuing delay and packet losses: Ideally, we want close to zero buffer occupancy at all times. Queued up packets in buffers mean extra latency for every packet. This is a problem for short flow performance because queuing delay in buffers can be a significant portion of flows’ completion time. If possible, we would like to achieve close to zero buffer occupancy or a loss-free network.
4. Efficiency: Naturally, at the same time we do not want to sacrifice the efficiency of high bandwidth-delay links such as the long haul fiber-optic links. These links, which often go through difficult terrains, are expensive, and service providers like to minimize unused capacity when the sources have traffic to send.
5. Differential bandwidth sharing: When need be, we would like to achieve some kind of differential bandwidth sharing among flows—for example, if we would like to give an important file transfer temporarily ten times the bandwidth share we give to a less urgent background movie download, congestion control should be able to achieve that.

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6. Network and traffic conditions: We want to achieve the above under any network conditions such as different round-trip times, bandwidth-delay products—including challenging conditions like wireless links with long delays, and high loss rates—and similarly under any traffic conditions such as short flows, long flows, or any mix of flows, flash crowds and so on. These issues are more relevant to the users. Users concern are more of quick flow completing time as most transactions on the internet demand that. This type of transactions seems to dominate in the future high speed network.

A. General and Specific Objective

The general objective of the research is to contribute to the general body of knowledge in the area of computer network performance to enhance productivity at workplaces in general and internet in particular.

To achieve the general objectives, the research addressed the following specific objectives:

- To reduce round-trip time of packet flow in the network.
- To enforce congestion control and fairness inside the network.
- To ensure efficient and fair bandwidth allocation on high bandwidth delay product networks while maintaining low queues and near-zero packet drop rate.
- Propose an efficient and effective window based control protocol which uses a feedback mechanism and allows explicit exchange of information between the end user and the network.

B. Problem Statement

The Internet is a global infrastructure for information exchange that has transformed the social, economic, and political aspects of our lives. One of the most crucial building blocks of the Internet is a mechanism for resource sharing and controlling congestion on the Internet. When end-hosts access a certain resource (such as a webpage from CNN, a video on YouTube, etc.,) on the Internet, it is important to ensure that they do not overwhelm network elements (such as routers), are able to efficiently utilize network resources, and achieve fairness in some agreed-upon sense. Today, congestion control for most of the traffic is provided by the Transmission Control Protocol (TCP)(Jacobson,1988). However, TCP is now showing significant performance limitations and the need for new transport protocol designs has become increasingly important (Alizadeh, et al 2010).This need has arisen from TCP's inability to meet the challenges brought about by the -pcapacities, latencies, and Bit-Error Rates (BER) as well as due to increased diversity in applications and their requirements.

C. Significance of Study

The results of this research study will categorically benefit all stakeholders of internet facility. Users will have their downloads and uploads times reduce drastically. Queue build ups on links are going to be reduced to nearly zero. Productivity at most organisations will increase as more organisations deploy their commercial activities on the internet. Social network activities on the internet will be enhance. More software vendors will go into designing delay sensitive applications. The future high speed network envisage by all to manage triple play look bright. With RCP more packets can be managed on the network without congestion.

D. Limitation

All studies have inherent limitations and delimitations. Limitations refer to limiting conditions or restrictive weaknesses. The research uses primary data for it analysis, this call for generation of the data. In studies of computer networks, it is highly expensive if not impossible to deploy real devices for experiments. To reduce the cost considerably and avoid damage of devices, simulation models are used. Despite the advantages of simulators, like most tools, do have their drawbacks. Many of these problems can be attributed to the computationally intensive processing required by some simulators. As a consequence, the results of the simulation may not be readily available after the simulation has started -- an event that may occur instantaneously in the real world may actually take hours to mimic in a simulated environment. The delays may be due to an exceedingly large number of entities being simulated or due to the complex interactions that occur between the entities within the system being simulated.

E. Delimitation

There are several means of handling congestion in computer network. Some of these are, over provisioning which is increasing capacities of devices attached to the network. This means is very expensive. Another means of ensuring that the network is not congested is to employed security approach. This method can also limit the availability of the network. The study is delimited to the control theory. The control theory approach offer an efficient means of handling congestion in computer networks. The study will not use secondary data because the researcher will generate data for analysis through simulation. Real devices like nodes and links will not be used because they are expensive and can get damage. These devices could be generated artificially through simulation. Data collection tools like questionnaire and interview will not be employed because simulation is employed to gather data needed for analysis.

II. METHODOLOGY

F. Research Design and Method

The researcher used Descriptive method. According to Glass and Hopkins (1984), Descriptive research can be either quantitative or qualitative. It can involve collections of quantitative information that can be tabulated along a continuum in numerical form, such as scores on a test or the number of times a person chooses to use a-certain feature of a multimedia program, or it can describe categories of information such as gender or patterns of interaction when using technology in a group situation. Descriptive research involves gathering data that describe events and then organizes, tabulates, depicts, and describes the data collection. It often uses visual aids such as graphs and charts to aid the reader in understanding the data distribution. The researcher in this case used NS2 simulator to generate data which is depicted in a graphical form.

G. Research Format

The researcher adopted Causal process. The rationale behind the choice being that Causality (also referred to as causation) is the relationship between an event (the cause) and a second event (the effect), where the second event is understood as a consequence of the first. In common usage, causality is also the relationship between a set of factors (causes) and a phenomenon (the effect). Anything that affects an effect is a factor of that effect. A direct factor is a factor that affects an effect directly, that is, without any intervening factors (Intervening factors are sometimes called "intermediate factors"). The connection between a cause(s) and an effect in this way can also be referred to as a causal nexus (Pear, 2009).The outcome of the simulated values of the protocols are evaluated against standard indicators like flow size, flow completing time, average flow completing time, maximum flow completing time under different traffic loads.

H. Key Assumptions

The main assumption is that packet drop within the network indicates that the network is congested. This work is also based on the assumption that simulation could be used to mimic real network topology with nodes representing host like routers, links representing transmission medium like copper, fiber or air and agents representing protocols like TCP, RCP, XCP etc.

I. Research Technique

The researcher adopted simulation to come out with the results of the research. The rationale for the choice is as follow: Network simulators provide a variety of needs. Judging against the time involved and the cost in creation of an entire test bed having multiple networked data links, routers and computers, network simulators are relatively inexpensive and fast. Network simulators permit engineers to test settings or scenarios that might be expensive or difficult to emulate employing real hardware. Simulators can aid in design of hierarchical networks employing various types of nodes like routers, computers, bridges, hubs, multicast routers, mobile units etc. We chose to use NS2 for this research, among other simulators, based on the fact it is the best-supported simulator, open source and includes a research community that consists of more than two hundred institutions worldwide (Breslau et. al. 2000). NS2 offers an attractive software platform in terms of its research interest for the study of congestion control algorithm. One part of the ns-allinone package is 'xgraph', a plotting program which can be used to create graphic representations of simulation results

J. Solution Strategic/Approach

For one to setup and model a network using NS2 simulation there is a need to write an OTCL script which will facilitate the procedure. The crucial stage of modeling a network is to define its topology. In NS2, the topology is defined by the use of three primitive blocks, which are agents, links and nodes. Nodes represent end hosts, that could be wired or wireless, that allow packets to be exchanged between other nodes. Links on the other hand are the physical transmission medium, either by air or wire which interconnects the nodes. Agents act as transport process that runs on the hosts. Once there is a definition of the topology, agents are then attached to the nodes and the traffic sources and sinks attached to the agents to send data. The traffic source nodes are where data emanates and sink nodes are where data is received. NS2 uses C++ to implement it as C++ is fast to run but slow to modify, thus making it appropriate for detailed protocol implementation. It makes it easier for reduction of packet size and event processing time. The Tcl for TCP, XCP and RCP are written and run. When the outputs are created, they can be visualised using either graphical representation called xgraph or a network animator known as nam (http://www.isi.edu/nsnam/ns/).

III. RESULT

In this section we present a simulation result that briefly depict how we increase the flow size against the average completion time and against the maximum flow completing time.

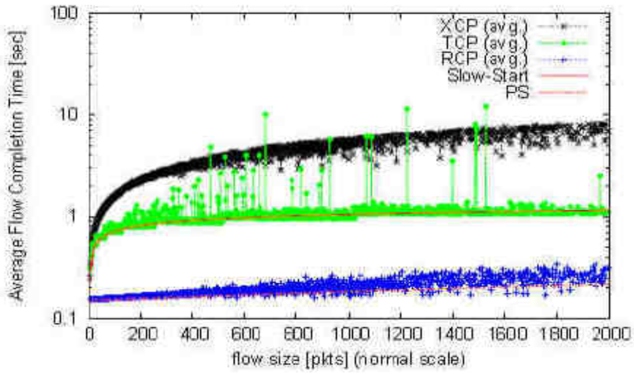

Fig. 3.1. Flow Duration (Secs) Versus Flow Size

Source: simulation

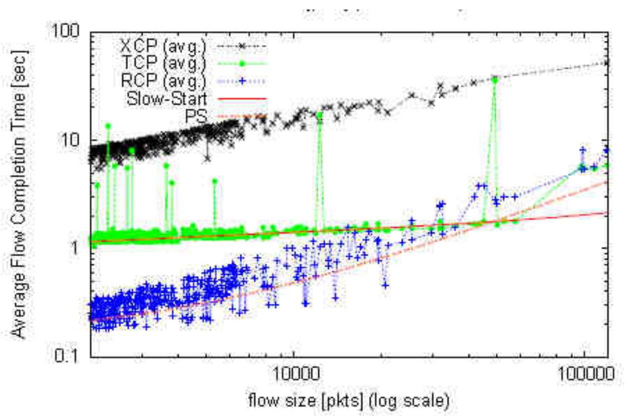

Fig. 3.2. Flow Duration (Secs) Versus Flow Size

Source: simulation

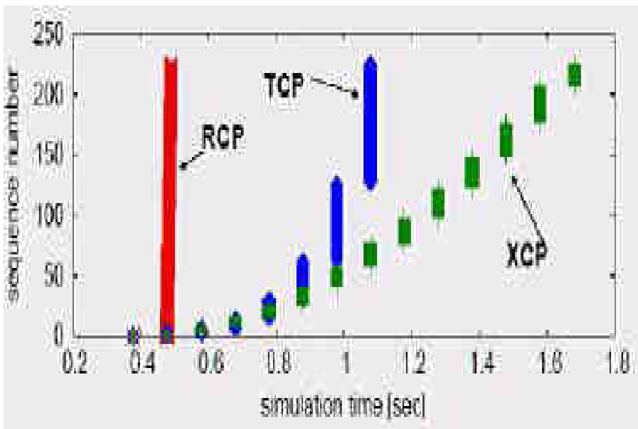

Fig. 3.3. RCP VRS TCP VRS XCP

Source: simulation

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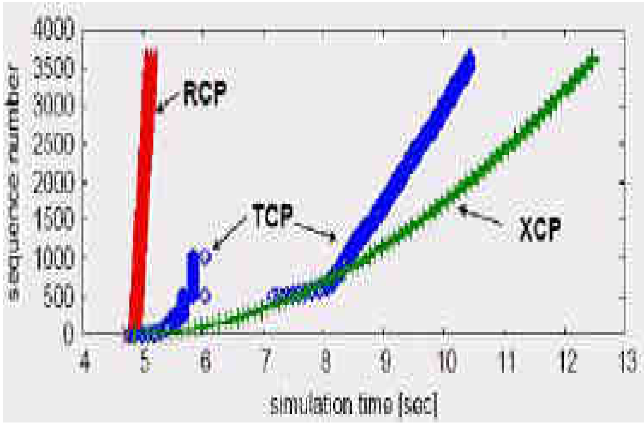

Source: simulation

IV. DISCUSSION

A well-known and simple method that comes close to minimizing flow completing time is for each router to use processor-sharing. In processing sharing, a router divides outgoing link bandwidth equally among all the flows for which it currently has queued packets. If all packets are equal-sized, the router can maintain a queue for each flow, and simply round-robin among the non-empty queues, serving one packet at a time. If packets are not equal sized, the router can use packetized processor sharing or fair queuing. In view of this, the above simulation is used to compare how close the protocols are able to minimize flow completion time with respect to flow size as compared to routers processor sharing. TCP flows start too slowly and are therefore artificially stretched over multiple round-trip times. Fig 3.1 depicts that the average flow completing time of TCP is about 10 times the flow completing time for PS average completing time. TCP also shows instability as the packet flow size increases in Fig 3.2. XCP is even more conservative in giving bandwidth to flows – particularly to new flows – which is why there are always more active, incomplete flows. It gradually reduces the window sizes of existing flows and increases the window sizes of the new flows, making sure there is no bandwidth oversubscription. Even though XCP depicts some form of stability as the flow size increases in both Fig 3.1 and 3.2, the average flow completing time is about 30 times that of PS average flow completing time. Thus making it unnecessary too long. Rate Control Protocol (RCP) greatly reduces flow completing times for a broad range for network and traffic characteristics. RCP achieves this by explicitly emulating PS at each router. It is depicted in both Fig 3.1 and 3.2 that RCP average flow completing time is almost equal to the PS of the routers. In RCP, a router assigns a single rate, R(t), to all flows that pass through it; i.e. unlike XCP, it does not maintain and give a different rate to each flow. RCP is an adaptive algorithm that updates the rate assigned to the flows, to approximate processor sharing in the presence of feedback delay, without any knowledge of the number of ongoing flows. It has three main characteristics that make it simple and practical:

1) The flow rate, R(t), is picked by the routers based on very little information (the current queue occupancy and the aggregate input traffic rate).

2) Each router assigns a single rate for all flows passing through it.

3) The router requires no per-flow state or per-packet calculations.

The basic RCP algorithm operates as follows.

1) Every router maintains a single fair-share rate, R(t), that it offers to all flows. It updates R(t) approximately once per RTT.

2) Every packet header carries a rate field, Rp. When transmitted by the source, Rp = 1. When a router receives a packet, if R(t) at the router is smaller than Rp, then Rp R(t); otherwise it is unchanged. The destination copies Rp into the acknowledgment packets, so as to notify the source. The packet header also carries an RTT field, RT Tp, where RT Tp is the source's current estimate of the RTT for the flow. When a router receives a packet it uses RT Tp to update its moving average of the RTT of flows passing through it.

3) The source transmits at rate Rp, which corresponds to the smallest offered rate along the path.

Fig 3.3 shows that load increase does not affect the completion time of RCP. TCP becomes unstable when the load increase exceeds 500, packets drop, establish equilibrium and starts increasing sharply. XCP on the other hand increases steadily with load increase.

V. CONCLUSION

The research confirm through extensive simulation that it is possible to use protocol to speed up transmission of packets on the network than always the demand for increase in bandwidth which is very expensive and does not give the needed solution. It was observed that as more delay sensitive traffic are injected into the network, the main transmission control protocol (TCP) becomes suspect and the feedback mechanism employed is implicit thus using packet drop as a means to manage congestion. In attempt to resend the drop packets cause delay and further congestion. Explicit Control Protocol (XCP) which manages the congestion better also take longer time to do so. It is inferred from the simulation that Rate Control Protocol (RCP) expedite flow ten times faster than TCP and thirty times faster XCP. This make RCP a preferred choice as we move into triple play more delay sensitive applications are been developed and users want to get their downloads faster.

ACKNOWLEDGEMENTS

It is an undisputable fact that a research of this nature cannot be carried out single-handedly. In line with this, I wish to thank almighty God for bring me to this far and express my appreciation and indebtedness to all who contributed in diverse ways to make this research successful.My first gratitude goes to Professor Reynolds Okai, Rector of Koforidua Polytechnic for giving me opportunity to serve as seminar co-ordinator for School of Applied Science. This appointment has sharpen and deepen my interest in research. My sincere gratitude and appreciation also go Professor Clement Dzidonu and Professor Francis Allotey both of Accra Institute of Technology for their invaluable contribution and pedagogical guidance towards my academic progression and professional career. I also wish to recognise Prof. Daparti Subba Rao who is my supervisor for my Phd research for his invaluable contribution to my Phd work in general and this paper in particular. I also wish to acknowledge Professor John Tindle of the School of

Computing and Technology, University of Sunderland, UK for always be in touch and encourage me to do more.I also wish to acknowledge staff of Koforidua Polytechnic, especially Computer Science Department for an excellent working relation. Finally, I wish to express my heartfelt appreciation to my family for their moral support and prayer.

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