Throughput Analysis of TCP Congestion Control Algorithms in a Cloud Based Collaborative Virtual Environment

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Abstract— **Collaborative Virtual Environment (CVE) has become popular in the last few years, this is because CVE is designed to allow geographically distributed users to work together over the network. In CVE the state of the virtual objects is witnessing unprecedantant change. When a user performs an action in CVE, the information of the action needs to be transmitted to other users to maintain consistency in the cooperative work. TCP is the most widely used protocol in the design of CVE, and its throughput deteriorates in the network with large delay. Gital et al, 2014 proposes a cloud based architectural model for improving scalability and consistency in CVE. Therefore, this paper aim at evaluating and comparing the performance of different TCP variant (Tahoe, Reno, New Reno, Vegas, SACK, Fack and Linux) with the cloud based CVE architecture to determine the suitability of each TCP variant for CVE. A comparative analysis between the different TCP variants is presented in terms of throughput verses elapse time, with increasing number of users in the system. TCP with the cloud based model was found to be effective, promising and robust for achieving consistency requirement in CVE systems.**

Keywords- Congestion Control Algorithm; Collaborative Virtual Enviroment; Cloud based Architectural model.

I. INTRODUCTION

Collaborative Virtual Environment (CVE) allows participant from distant geographic location to share a common virtual environment, including virtual entities and resources maintained by a group of computers, in such a way that it can support effective communication among the users to achieve synergistic coordination of tasks [1-3]. Applications of CVEs include Education, massively multiplayer online games (e.g., World of Warcraft), virtual worlds (e.g., Second Life), military training, industrial remote training, and collaborative engineering [4-7]. As the number of concurrent participants is becoming larger, data transmission alone the network may no longer provide the

level of consistency required, typically in terms of response time [1, 7, 8].

In the TCP/IP network model, TCP is the widely used transport protocol that provides reliable packet delivery over an unreliable network. This protocol is designed to be used with the Internet Protocol (IP). Virtual collaborative applications have been designed with protocols that provide timing of data transmission due to the consistency requirement in the system. Users in CVE cooperate with each other and interact with the virtual environment; the state of the virtual environment is changing fast, how to transmit the interpretation of the user level interaction in the network is a challenging task. This is because CVE is more about the performance of the virtual world, consistency is more critical since a delay in data transmission leads to unsatisfactory results. UDP has been criticized for use as the transport layer protocol for its lack of congestion control mechanism [9-11]. With the increasing speed of network and readily available internet, bandwidth is no longer a limiting factor in the internet. The CVE application today are built over TCP. The TCP provides sequence deliverance of data and unfailing data transmission among communicating nodes. One of the strengths of TCP is its high responsiveness toward network congestion. TCP is also a defensive protocol as it detects incident congestion as its result to try and lessen the impacts of the congestion. Thereby prevent collapse of communication [12]. The TCP focuses on reliability, stability, and correctness of data transfer which fits well with requirements of loss sensitive applications such as web browsing and file transfer and left with the problem of delay in time-dependent applications. The reliability and stability comes at the cost of variable delays in data transmission that can create problems for TCP [13].

As different implementations of TCP protocols have been introduced, analysis and evaluation studies have been conducted to measure the performance of different TCP

variants. For example, [14] compare the performances of different TCP variants with the routing protocols DSDV and AODV, experimented in 20 different ways and find out that TCP Tahoe has the least number of packet drops against the simulation time. Some of the other variants even though they started with a lesser number of packet drops, the TCP Tahoe variant has always the least amount of packet drops in all cases when using AODV and DSDV. [15] study the performance of TCP Vegas versus different TCP variants in homogeneous and heterogeneous wired networks are performed via simulation experiment using network simulator 2 (ns-2). The performance of TCP Vegas outperforms other TCP variants in the homogeneous wired network. However, it achieves unfair throughput in heterogeneous wired network. [16] presents a comprehensive experimental analysis of TCP variants under MPLS with emphasis on Tahoe, Reno and Vegas under different traffic load. It has been found that Reno and Tahoe fail to take advantage of MPLS features whereas Vegas has shown promising results with almost stable, constant end-to-end delay after a transient. [17] Compared TCP Tahoe, NewReno, Vegas, and Sack overself-similar traffic. They found that NewReno did better than other TCP variants with respect to efficiency and throughput. TCP Vegas showed better throughput than Reno.

However, we have not found studies that compare the performance analysis of TCP variants in cloud based CVE architecture. In this paper, we present a performance analysis of different TCP variants with cloud based CVE architectural model [18] to determine the suitability of each variant for CVE systems.

The rest of the Organization of the paper is as follows: The Section II briefly describes the cloud based architectural model. The evaluated TCP variants are described in section III. An overview of CVE data types is described in section IV. Section V presents the simulations, and Finally, section VI presents the analysis on simulation result and conclude the paper.

II. CLOUD BASED ARCHITECTURAL MODEL

The architecture of the cloud based CVE is proposed by the modification of previous CVE. The CVE moves into the cloud instead of the conventional environment presently in used. The cloud based CVE comprised of the cloud infrastructure. This layer enables the provision of networking components, servers, storage, routers, and switches. The Cloud Computing infrastructure heavily influences application performance and throughput in a distributed computing environment [19]. It is responsible for hosting and given supportive coordination to infrastructures including the platform of cloud, repositories, computers, servers, network communication devices, storage units among other physical structures like building. The resources of the information and communication technology are distributed by the cloud infrastructure. The cloud platform provides both services and inters connections among the systems on the platforms as well as to provide an easy way for the system's hardware to operate just like the internet. Furthermore, the cloud infrastructure allowed the hardware to securely access data in a sharable platform. Data transmission between the different components of the cloud architectural model uses UDP and TCP as the transport layer protocol.

III. OVERVIEW OF SOME SELECTED TCP VARIANTS

A. TCP Tahoe

TCP Tahoe [20-23] by Van Jacobson is a method based on the principle of packet conservation. Packets get into the network only when there is bandwidth available. This principle is implemented by using acknowledgement. By sending acknowledgement, it means that a packet has reached its destination, leaving available bandwidth it occupies for sending another packet. It also maintains a congestion window CWD to reflect the network capacity. TCP Tahoe suggests that whenever a TCP connection starts or re-starts after a packet loss it should go through a procedure called slow-start. This is because an initial burst might overpower the network and the connection might never get started. The congestion window size becomes double (Multiplicative Increase) for each transmission until there is congestion in the network. Slow start suggests that the sender sets the congestion window to 1 and then for each ACK received it increase the CWD by 1. This implies that in the first round trip time (RTT) only one packet is sent, and keep doubling after each RTT. When there is congestion, the sending rate and the congestion window are set to 1 and start over again. The important thing is that Tahoe detects packet losses by timeouts. The sender is notified that congestion has occurred based on the packet loss.

B. TCP Reno

TCP Reno [20, 22-24], also known as standard TCP is the most widely adopted Internet TCP protocol. The method uses the four phases of transmission: slow start, congestion avoidance, fast retransmit, and fast recovery. Link congestion is indicated by either receives of duplicate acknowledgment or expiration of retransmission time out (RTO). When the sender receives duplicate acknowledgements (ACKs), the sender activates TCP fast retransmit and recovery algorithms and reduces its congestion window size to half. It then linearly increases the congestion window as in TCP Tahoe. This is the same in the case of congestion avoidance. This increase in transmission rate is slower than in the case of a slow start and helps relieve congestion. TCP Reno fast recovery algorithm improves TCP performance in case of a single packet loss within a window of data. However, the performance of TCP Reno suffers in case of multiple packet losses within a window of data [21, 25, 26].

C. TCP New Reno

TCP NewReno [27] is a modification of TCP Reno. It improves the retransmission process during the fast recovery phase of TCP Reno. TCP NewReno can detect multiple packet losses. It does not exit the fast recovery phase until all unacknowledged segments at the time of fast recovery are acknowledged [20, 28, 29]. TCP Reno overcomes the problem of reducing the congestion window size many times when there is time there is multiple packet losses. TCP NewReno maintains the slow start, congestion avoidance, and fast retransmits of TCP Reno. It exits fast recovery after receiving acknowledgement of all unacknowledged segments and then sets the congestion window size to slow start threshold and continues the congestion avoidance phase [30]. TCP New Reno retransmits the next segment after the received partial acknowledgment. (Partial acknowledgments are the acknowledgments that do not acknowledge all outstanding packets at the onset of the fast recovery.)

The critical issue in TCP New Reno is that it is capable of handling multiple packet losses in a single window. It is limited to detecting and responding only one packet loss per RTT. This insufficiency becomes more distinct as the delay-bandwidth becomes greater. However, still there are situations when stalls can occur if packets are lost in successive windows, like all of the previous versions of TCP New Reno which infer that all lost packets are due to congestion and it may therefore unnecessarily cut the congestion window size when errors occur [28, 31].

D. TCP VEGAS

TCP Vegas [32] is a modification of TCP Reno [32]. It builds on the fact that proactive measures to encounter congestion is much more efficient than reactive measures. Vegas tried to get around the problem of coarse grain timeouts by suggesting an algorithm which checks for timeouts at a very efficient schedule [32]. Also, it overcomes the problem of requiring enough duplicate acknowledgements to detect a packet loss, and suggests a modified slow start algorithm which prevents it from congesting the network [20, 26]. The three major changes induced by Vegas are: New Re-Transmission Mechanism, Congestion avoidance and Modified Slow-start. Vegas try extending on the retransmission mechanism of the standard TCP. It monitors when each segment was sent and calculates an estimate of the RTT by monitoring the time it takes for the acknowledgment to get back [20, 32]. During Congestion avoidance, TCP Vegas does not use the loss of segment to signal that there is congestion, instead, it determines congestion by a decrease in sending rate as compared to the expected rate, as a result of large queues building up in the routers. It used Tri-S scheme [20, 32].
In the case of slow-start phase, TCP Vegas differs from the other implementations. This is because at the beginning of each connection, Vegas have no idea of the available bandwidth. It is possible that the bandwidth was overshoot during the exponential increase by a big amount and thus induces congestion. Vegas increases exponentially after every RTT, and calculates the actual sending throughput to the expected and when the difference goes above a certain threshold, it exits slow start and enters the congestion avoidance phase [20, 32].

E. TCP SACK

SACK [33] algorithm is an extension of the standard TCP. It allows a TCP receiver to acknowledge out-of-order segments selectively rather than cumulatively by acknowledging the last correctly in order received segment. The sender retransmits only missing segment after receiving acknowledgement of out of order packets from the receiver. It does not send the entire unacknowledged segment. A TCP SACK behavior is similar to that of TCP Tahoe and TCP Reno, which are robust in case of out of order packet arrivals [22, 23, 34]. However, it improves the performance of the TCP Reno when there are multiple packet losses. TCP SACK maintains a variable called *pipe* that represents the estimated number of outstanding packets during fast recovery phase [17]. The sender only sends new or retransmitted data when the estimated number of packets in a router is smaller than the congestion window. The *pipe* variable is incremented by 1 when the sender either sends a new segment or retransmits old segment. It decreases by 1 when the sender receives the duplicate ACK.

The disadvantage with SACK is that currently selective acknowledgments are not provided by the receiver. To implement SACK, there is a need for full implementation of the selective acknowledgment.

F. TCP FACK

FACK TCP (TCP with forward acknowledgement) [35] was developed to decouple the congestion control algorithms from the data recovery algorithms. It uses the additional information provided by the SACK option to maintain an explicit measure of the total amount of outstanding data in the network. However, standard TCP Reno and TC SACK both attempt to estimate this by assuming that each drawback received represents one segment that has left the network. The basic concept of Fack mechanism is by considering the greatest sequence number of forward selective acknowledgement as a mark that completely previous segments which unselectively acknowledged were lost. This method improved the recovery process of packets losses significantly. Fack algorithm takes an aggressive method, and considers all unacknowledged holes as lost packets and Sack blocks. FACK implementation improved TCP performance than the traditional approach; it is excessively aggressive if packets have been rearranged in the pipeline, due to these holes between the blocks of Sack, FACK does not designate packet loss in this state.

IV. OVERVIEW OF CVE DATA TYPE

Similar to another update message in standard applications, message update in CVE have strong delay requirements. According to [36], the delay in update transmission in CVE should not exceed 100msec. [37] argue that up to 200msec delay is acceptable. This requirement is based on the real time and highly reactive multiuser processes where users' actions are based on action of another user; therefore requiring a very low transmission delay of updates. Also, collaborative update messages are severely affected by jitter. It has been shown that a CVE session with a low 10 msec delay that has jitter, results in a collaborative environment which is almost as bad as one with 200 msec delay but no jitter [37]. Finally, collaborative update messages have strict reliability requirements. It is obviously pertinent that all users receive update messages or they won't be able to collaborate. In a typical CVE system, the last state of a shared object is the most crucial data. For example, if a user moves an object, generating 10 update messages each 50msec apart. If the update messages 1, 7 and 9 are lost; there is no need of retransmission because the last state of the object is received correctly.

In CVE systems, a server executes the function of: receiving the update messages from the clients; updating the whole virtual environment and transmitting updates of the virtual environment to other clients and servers to keep consistency in the virtual environment. A client also executes functions to: receiving the user's input as the update message; transmitting the update message to the server and receiving the update messages from the server to keep consistency in the virtual environment.The size of the message packets transmitted for these functions in CVE are mostly uniformly sized (80, bytes).

V. SIMULATION

Ns2 simulator is used for the simulation of our experimental setup on a machine with the following configurations: Intel (R) Core (TM) i5-2410m processor, 2.30GHz speed, 4.00GB RAM with Obuntu operating system. The network topology used in the simulation is flex bell topology shown in Fig. 1. The topology consists of TCP senders, TCP receivers and a pair of routers. The link between the sender's nodes and routers is termed sender's link and it is connected to different router because the users are formed from different subnet and each subnet is connected to a router A, while the link between the receivers and router B is called the receiver link. The sender and receiver links represent a local area network (LAN). The link between router A and router B represent the bottleneck link within the cloud in the form of a wide area network (WAN).
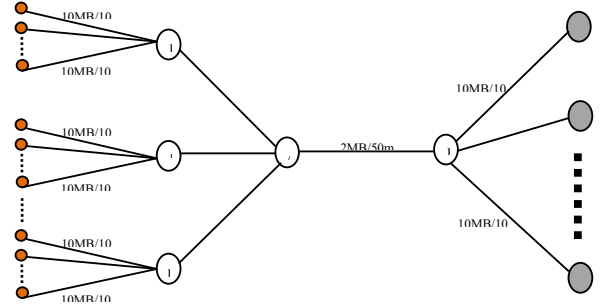

Fig. 1 Simulation topology

The links between the sender's nodes and the Cloud link are full wired duplex link. The bandwidth of the sender's links is set to 10Mbps with 10ms delay. The bandwidth of the receiver links that represent the cloud is also set to 10Mbps with 10ms delay. The speed of the cloud and that of local area network of the senders are assumed to be equal. The bottleneck link is set to 2Mbps with 50ms delay to represent a connection to cloud infrastructures. The number of sender's node which is equivalent to the number of concurrent collaborators, is set to 200 with 2 receivers' node in the first simulation. In the second simulation the sender's nodes increase to 600 with 6 receivers' node. This setting represents a virtual environment with two partitions and six partitions respectively, with 100 users as server threshold. The simulation parameters of the network topology are shown in Table I. In this simulation, the throughput of Tahoe, Reno, NewReno, Vegas , Fack, SACK and Linux TCP in a cloud setting is evaluated.

TABLE I. SIMULATION PARAMETERS

Link	Bandwidth	Delay	Queue Limit	Window Size	Packet Size	Traffic Types
Link to Cloud (Bottleneck)	2Mbps	50ms	100	8000kb	552B/200B	-
Senders Link	10Mbps	10ms	-	-		Telnet/ CRB
Link to Cloud Infrastructures (Receivers) within the Cloud	10Mbps	10ms	-	-		Telnet/ CRB

VI. RESULTS AND DISCUSSION

During this simulation, the throughput of TCP Variants (Tahoe, Reno, NewReno, Vegas, Fack, SACK, and Linux) ware measured in two different simulations as described in the previous section. The simulation period is set to 300sec in each case. The throughput analysis results of the simulation are shown in Fig. 2a and 2b.
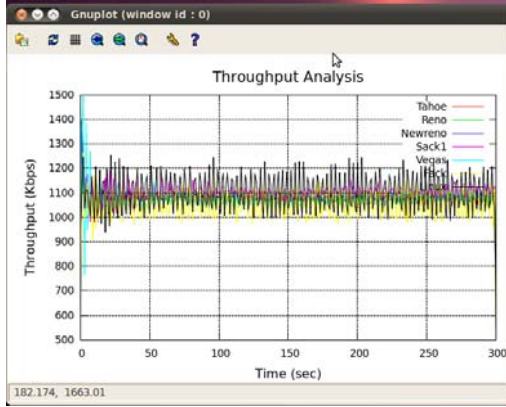
(a)
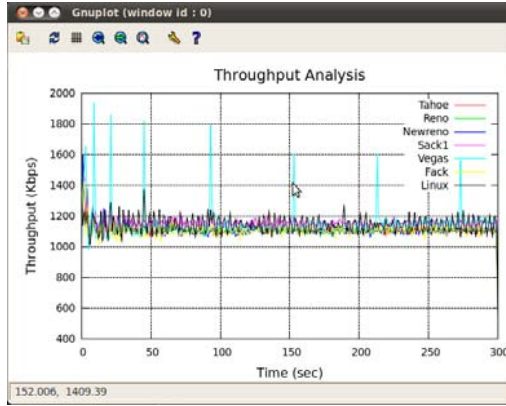
(b)

Fig. 2. (a) Throughput analysis with 200 users and 2 Servers. (b) Throughput analysis with 600 users and 6 servers.

The average throughput of the variants with 200 users and 2 servers, and the average throughput of the variants with 600 users and 6 servers are shown in Table II. In the first simulation with 200 users and 2 servers, TCP Vegas indicate a fair performance followed by NewReno, Tahoe, Linux, Reno, Sack1 and Fack, respectively. Therefore, it can be stated that the TCP Vegas performs better in the cloud based CVE model. In the second simulation with 600 users and 6 servers, the throughput increases due to the size of the traffic generated. TCP NewReno show a fair performance, followed by Vegas, Reno, Fack, Sack, Linux then Tahoe

respectively. It shows that, the variants performance is affected by the increasing number of users in the system.

TABLE II AVERAGE THROUGHPUT OF THE VARIANTS FOR THE TWO SIMULATIONS

TCP Variant	Average throughput with 200 users and 2 servers	Average throughput with 600 users and 6 servers
Vegas	1130.872	1468.377
Fack	997.6375	1235.005
Linux	1100.007	1191.41
Tahoe	1100.415	1134.994
Reno	1097.21	1240.135
NewReno	1225.435	1329.895
Sack1	1090.798	1176.02

VII. CONCLUSIONS

The paper is to compare the performance of TCP variants in cloud based CVE architecture. The Tahoe, Reno, New Reno, Vegas, Sack1, Fack and Linux TCP were evaluated and compared in the cloud based architectural model of Gital et al. (2014) in the study using ns2 simulator. Comparing the results obtained from the study indicated that the performance of the TCP Vegas with cloud based architecture is better than the other TCP variants. The performance metrics use in our study are throughput and time.

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