Enhancement of TCP Performance by Jitter Control Scheme for Communication Networks

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Abstract: In the recent years, by significant popularities in Internet services, Communication Network (CN) traffic is dominated by the data generated by the applications such as web access, multimedia downloads, email, file transfer etc. The Communication network may be wired and wireless network. Since the demand of mobile users grows rapidly, the integration of wired and wireless networks is widely deployed. One of the important affecting factors for providing performance in the communication network is packet losses (PL). In CN, packet losses are due to non-congestion as well as congestion. The Congestion loss (CL) is due to burst traffic in the network and non-congestion loss (NL) is due to high bit error rate or jitter or frequent disconnections. Transmission control protocol continues to be the most important transport layer communication protocol. There exist some unsolved challenges for applying TCP over communication networks. One of which is to understand the reason for packet loss due to no-congestion or congestion in communication networks. On the other hand, to utilize the bandwidth effectively is another tradeoff issues. In order to satisfy these problems and improve performance, we propose to Enhance Jitter-based TCP (EJTCP) scheme. The experiment results show that EJTCP carries out good performance over the communication networks.

***Index Terms* - Jitter; Jitter control; Congestion loss; Non-congestion loss; random loss; transmission control protocol; end-to-end control.**

I. INTRODUCTION

Communication Networks (CN) have spread significantly, in the last decade. Therefore packet switching processing ultimately combined the voice and data networks with each other into a converged and integrated multimedia network. The perspective of the converged integrated network is growing further to integrate wire, wireless and cellular technologies. The all Internet protocol wired and wireless hybrid network is becoming sincerity and the wireless network is getting more associated with our daily communications. The developing gradually cellular technologies, integration of wireless networks, and cellular networks reach out the Internet protocol beyond the limits of geography and territory [1]. Transmission control protocol / Internet protocol (TCP/IP) is the powerful communication protocol suite in today's multimedia application. TCP is one of core protocol of the Internet protocol suite. Within a connection, TCP can guarantee such as in order delivery or reliable data transmission using retransmission techniques. Nowadays, most of the Internet traffic is take away by TCP [2], including traffic generated by web access-mail, bulk data transfers, and remote terminals. Encourage by the demand for wireless Internet, TCP/IP needs to get off from its original wired network oriented design and develop to meet the challenges introduced by the wireless portion of the network. TCP was designed for wired networks. In such networks, the network error rates are very low and congestion is the main cause for packet loss. TCP reacts to packet losses by retransmitting the missing packet and call upon congestion control. It is observed that TCP, when used for wireless networks, results in degraded end-to-end throughput and sub-optimal performance [3]. This is due to the communication characteristics of the wireless networks. Wireless networks have different communication characteristics such as low bandwidth (LB), high bit error rates (BER) and handoffs. TCP misinterprets the packet losses due to the above reasons and hence inspire for congestion control [4], resulting in performance degradation.

In wired networks, the random bit error rate (BER) is negligible, and congestion is the main cause of packet losses. The TCP sender adjusting the sending rate of data packets is activated by the self-clocking acknowledgement (ACK) sent by the corresponding receiver after successfully receiving the data packet. When packet loss occurs at a congested link due to buffer overflow at the intermediate router, the sender receives either duplicate ACKs or the sender receives retransmission timeout (RTO) timer expires. This event activates the sender's congestion control mechanism by which the sender reduces the size of its transmission window, or congestion window, resulting in a lower transmission rate to cause the link congestion. Such TCP senders conduct works properly well in the wired networks where packet losses are almost caused by bandwidth congestion; and packet losses due to bit error are normally negligible or, if any, not exceeding one packet loss per congestion window . However, in heterogeneous networks, high BER, fading, and blackout become non-congestion factors for packet losses. Standard TCP congestion control and congestion avoidance mechanisms based on the presumption that all packet losses are due to congestion become incapable of handling the mixed packet losses. TCP without modification display throughput degradations when used in heterogeneous networks [5]. In the wired network, the losses are usually caused by congestion; however, the non-congestion losses may occur due to random or burst errors in the wireless networks. TCP will unnecessarily half down it's transmitted window size before retransmitting lost packets or initiate its congestion windows and backoff its retransmission timeout (RTO). It is not good to reduce the sending rate especially when the event is not caused by congestion. Since TCP has much reason to improve, some of them is an inability to distinguish the actual reason of congestion or not, packet reordering which is not a rare event in wireless networks and out of order delivery in wireless networks. Therefore, TCP will participation in performance degradation since it cannot distinguish the lost event caused by real congestion or random drop of the wireless characteristics [6].

In the wireless network, packet losses are due to congestion as well as non-congestion [7]. The congestion loss (CL) is due to burst traffic in the network and non-congestion losses (NL) is due to high BER or jitter or frequent disconnections. Packet loss (PL) due to transmission in the wired network can be ignored [8]. The variation in inter arrival time or the packet delay variation is called jitter [9]. Increasing Jitter is an indication of an increase in congestion and decreasing jitter is an indication of smoother transmission. This nature of jitter helps to distinguish the congestion loss and non-congestion loss. A positive jitter value which is greater than half of the average round-trip value results in timeout loss. Any packet arriving after its scheduled time is discarded by the receiver. The problem of delay jitter is thereby transformed into end-to-end delay and packet losses. Moreover, for a timeout loss, the sender TCP does an exponential backoff for some time. The effect of the backoff is that the window size does not grow for a small period, after which it starts growing at a normal rate [10].

In wireless networks, the loss of data segments does not necessarily correspond to network congestion, because wireless networks are more facing downwards to high error rates, variable delays and low transmission rate than wired networks [11]. These characteristics determine different types of packet losses. So packet losses (PL) can be classified as random and burst non-congestion losses [12].

In random non-congestion losses, wireless network losses often depend on environmental characteristics introducing network problems, such as fast fading, interference, and shadowing effects. These effects lead to the corruption of some bits in the data packet. If such errors cannot be recovered by base levels, the packet is considered lost, being thrown away without acknowledgement. Moreover, if the Round Trip Time (RTT) delay variation reaches greater values than the estimated Retransmission Timeout (RTO), a counterfeit timeout event occurs, and the packet is by mistake considered lost. These random losses are often uncorrelated and not predictable.

In burst non-congestion losses, wireless networks can also be affected by correlated packet losses, caused by lengthy channel problems. These problems can be produced by interferences, long fading events or disconnections due to user mobility. If the loss event dues to congestion loss sender most reduce its sending rate for too many data flows were transmitted on a network. On the other hand, a loss event due to wireless random loss does not denote a bad network condition. Misjudging of loss event will decrease transmission performance of the connection. The main issue of wireless TCP is how to distinguish packet loss events due to congestion or wireless network random loss [13]. In these cases, several continually data packets can be lost, triggering a strong reduction of TCP congestion windows and consequently of its throughput. In this paper, we consider jitter which cases wireless non-congestion loss.

II. JTCP Algorithm

The Jitter-based TCP (JTCP), developed by Wu and Chen [14], represents an innovative solution to differentiate between congestive (CL) and non-congestive losses (NL) in wireless networks. JTCP conclude congestion by choosing the Jitter ratio (JR) as a loss ratio predictor. In order to calculate the JR, the interarrival jitter has been introduced [15]. The interarrival jitter $D(i,j)$ for the pair of packets (i,j) defined as the difference between the inter-packet times at the receiver and the inter-packet times at the sender, as shown in fig. 1.

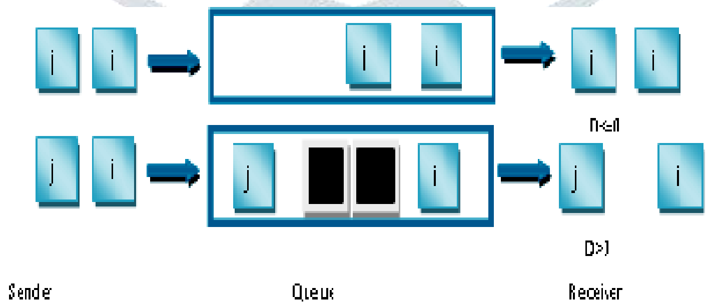

Fig. 1 Description of Interarrival Jitter

If we assume S_i, R_i and S_j, R_j the sending and receiving times of data segments i and j, respectively then the interarrival jitter is defined as

$$\begin{aligned} \boldsymbol{D(i,j)} &= \boldsymbol{(Rj - Ri) - (Sj - Si)} \\ &= \boldsymbol{(Rj - Si) - (Rj - Si)} \qquad (1) \end{aligned}$$

The jitter ratio JR is calculated as:

$$JR = \frac{D(i,j)}{Rj - Ri} = 1 - \frac{Sj - Si}{Rj - Ri} \qquad (2)$$

To differentiate between congestion loss and non-congestion loss the JTCP used following steps

1. $JR < Th$ i.e. if the average jitter ratio is less threshold values, an inferred data segment loss is considered as a con-congestion loss (NL)
2. $JR \geq Th$ i.e. if the average jitter ratio is greater than the threshold values, a data segment loss is considered as congestion loss (CL) and reduces its congestion window size to one half. Where Th is a threshold, which is defined as $Th = \frac{k}{cwnd}$, and $cwnd$ is a congestion window value (in Segments) and k is a control parameter.

III. PROBLEMS IN JTCP

The Transmission control protocol offers a connection oriented, reliable delivery service for a stream of a packet sent from one system to another without duplication or data loss over an Internet Protocol network. One of the most critical issues for TCP is congestion control. In the wired networks, the losses are usually caused by congestion; however, the non-congestion losses may occur due to random or burst errors in the wireless networks. Hence JTCP uses jitter ratio to compare with the threshold. When TDACKs is detected and jitter ratio is estimated to be larger than throughput, the sender regards that its sending rate is larger than the bottleneck link's capacity and packet loss was caused by congestion but channel loss. Although JTCP outperforms other schemes in throughput performance, it may make false determine lost event under bursty cross traffic. The JTCP uses jitter ratio to determine the relationship between transmission rates of JTCP and utilization of bottleneck's queue. However, this threshold may be a problem under bursty cross traffic because it uses extra timeout by burst transmission. The next problem is to the computation of an inaccurate jitter and influence of a jitter ratio.

IV. PROPOSED EJTCP SCHEME

In Communication Networks (CN) with the wireless network, packet loss is not only due to congestion events, but also caused by random or burst errors on the wireless network. TCP congestion control mechanism is not able to recognize between congestive or non-congestive losses. Consequently, it unnecessarily halves down its congestion window $(cwnd)$ and backoff its retransmission timeout (RTO). In this way, TCP flows on wireless network suffer heavy performance degradation, because of an uncorrected deduction of network congestion. Therefore a very favorable solution to improve the performance is the enhance Jitter based TCP (EJTCP) that is able to recognize if a packet loss is due to congestion events or random events.

The key idea of the proposed mechanism is to monitor the wireless networks error status continuously by counting the number of concluding CL and NL events. According to this monitoring activity, the congestion control scheme takes an appropriate decision about the transmission control protocol window size reduction and retransmissions. In particular, when three DACKs or an RTO are detected, the algorithm evaluates whether it is a CL or NL and it adjusts the congestion window opportunely. To monitor the wireless network we define two variables CL_i which is a congestion loss counter and NL_i which is a non-congestion loss counter. We know from JTCP the threshold value is

$$Th = \frac{1}{cwnd} \qquad (3)$$

If non-congestion events inferred (occurred) i.e., $JR < Th$, NL_i loss counter is increased otherwise congestion events inferred (occurred) i.e., $JR \geq Th$, then the CL_i loss counter is increased. Finally, in order to obtain a smoothed value of CL_i and NL_i a low pass filter is used at each monitoring window, by adopting the exponential weighted moving average (EWMA) [16], as given in the following equation (4) and (5)

$$CL_i = \beta\ CL_i + (1-\beta)\ CL_{i-1} \qquad (4)$$

$$NL_i = \beta\ NL_i + (1-\beta)\ NL_{i-1} \qquad (5)$$

Where β is a smoothing factor, which denotes how recent and old samples of CL_i and NL_i influence the EWMA estimates. A non-congestion loss can be detected at the transport level by the TCP sender decides which counter i.e. CL_i and NL_i should be increased. In order to better adjust the congestion window size and hence the aggressiveness of TCP reaction, the following parameter α has been introduced as follows:

$$\alpha = \frac{NL_i}{CL_i} \qquad (6)$$

Where α is the ratio between non-congestion and congestion loss rate. This parameter helps to understand whether corruption or congestion is the predominating condition in a certain time window. So when α>1 indicates that non-congestion losses are predominating in the wireless network; thus a loss event is expected to occur mainly due to a loss-prone network. On the other hand, when α<1, congestion losses predominating in the monitored window, and the TCP congestion control action is needed.

Therefore when three duplicate ACKs are received or α>1, the event is considered caused by a lossy link and immediate recovery is performed. Immediate recovery is also performed if α =1 and $JR \leq Th$.

Otherwise, if α<1, TDACKs are considered caused by a congestion, and fast recovery action is performed. Fast recovery is also performed if α =1 and $JR > Th$. As given in the algorithms 1 and algorithm 2. The parameters used in Algorithms are given in table 1.

Table 1 Description of parameters used

Definitions	Description
α	Control parameter
JR	Jitter ratio
$cwnd$	Congestion window
$ssthresh$	Slow start threshold
Th	Threshold value
NL	Non congestive loss
CL	Congestive loss

Algorithm 1: Pseudo code for 4DACKs

$$
\begin{array}{l}
if\ (4\ DACKs\ are\ received)\ then \\
\quad if\ (\alpha \geq 1) \\
\quad\quad if\ \left(JR \leq \frac{1}{cwnd}\right)\ then \\
\quad\quad ssthresh = cwnd \qquad //\ NL \\
\quad\quad cwnd = ssthresh \\
\quad\quad else \\
\quad\quad if\ \left(JR > \frac{1}{cwnd}\right) then \\
\quad\quad ssthresh = \frac{1}{2}\ cwnd \qquad //CL \\
\quad\quad cwnd = ssthresh \\
\quad\quad endif \\
\quad else \\
\quad\quad if\ (\alpha = 1) then \\
\quad\quad ssthresh = \frac{1}{2}\ cwnd \qquad //CL \\
\quad\quad cwnd = ssthresh \\
\quad\quad endif \\
\quad endif \\
endif
\end{array}
$$

After an RTO, if α >1 or α =1 and $JR \leq Th$, fast recovery is performed, since a loss is inferred due to the non-congestion event. On the other hand, if α <1 or α =1 and $JR > Th$, the occurred loss is inferred as a congestion event, thus triggering a slow start phase. The following figure 3 shown the retransmission timer expiration events.

Algorithm 2: Pseudo code for RTO

$$
\begin{array}{l}
if\ (Timeout\ expires)\ then \\
\quad if\ (\alpha \geq 1) \\
\quad\quad if\ \left(JR \leq \frac{1}{cwnd}\right)\ then \\
\quad\quad ssthresh = \frac{1}{2} cwnd \qquad //\ NL \\
\quad\quad cwnd = ssthresh \\
\quad\quad else \\
\quad\quad if\ \left(JR > \frac{1}{cwnd}\right) then \\
\quad\quad ssthresh = \frac{1}{2}\ cwnd \qquad //CL \\
\quad\quad cwnd = 1 \\
\quad\quad endif \\
\quad else \\
\quad\quad if\ (\alpha < 1) then \\
\quad\quad ssthresh = \frac{1}{2}\ cwnd \qquad //CL
\end{array}
$$

$cwnd = 1$
$endif$
$endif$
$endif$

V. SIMULATION AND RESULTS

In our simulation results, we used NS-2 network simulator for the different parameters such as throughput, packet lost and delay, and jitter.

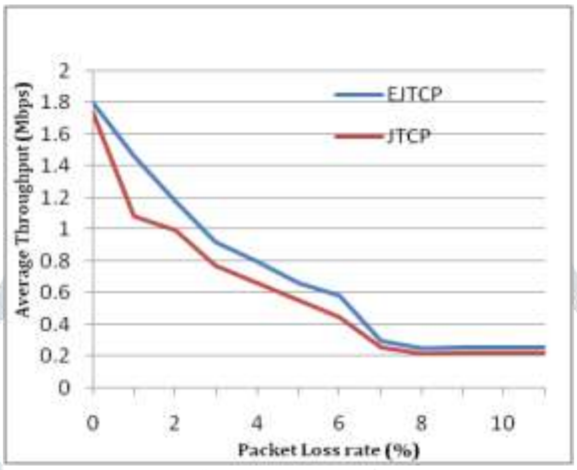

Fig. 2 Average Throughput v/s Packet loss rate

In Fig. 2 we compare the throughput of enhancing version of JTCP with jitter based TCP assuming independent errors ranging from the percentage of 0 to 10 packet loss probabilities. In Fig. 4 EJTCP has more than 15 percent average improvement for the percentage between 1 to 5 packet error rates. Even though the packet discard rate reaches percent of 10, the average throughput for ETCP is still higher than JTCP.

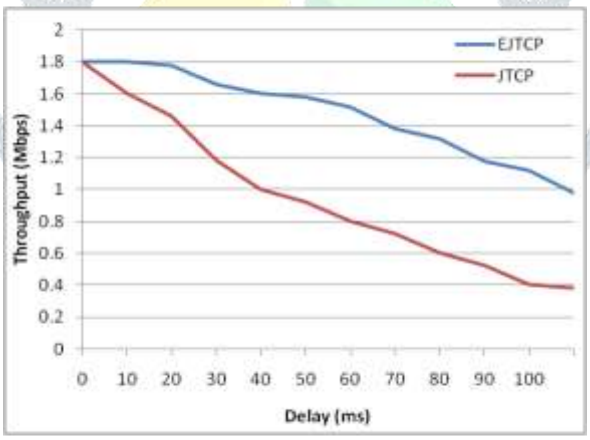

Fig. 3 Average Throughput v/s Delay

In Fig 3 we compare average throughput with delay. The average delay has been calculated as the sum of the delays needed for every single packet to be correctly received, divided by the number of received packets. As we know that the length of propagation delay may affect average throughput. So we apply the percentage of 1 loss rate to the wireless network. When the propagation delay is increased, the influence of loss misjudgment is more severe. Therefore EJTCP can maintain good throughput as the propagation delay is increased.

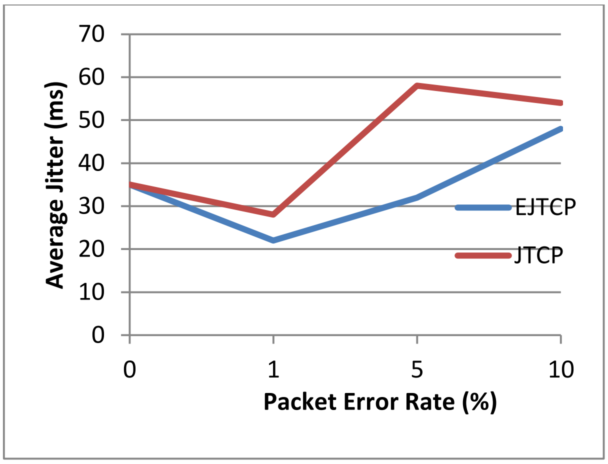

Fig. 4 Average Jitter v/s Packet Error Rate

In Fig 4 we compare average jitter with packet error rate. JTCP exhibits a worse behavior with a higher number of lost packets than EJTCP. In fact, packet error rate augments; JTCP is less capable than EJTCP to distinguish between congestion and corruption problems which caused the loss event. Therefore, Transmission Control Protocol connections undergo many consecutive timeouts under a high jitter value due to congestion. This fact impacts on the average jitter, since the higher is the number of slow start phases, the more are the oscillations on the degree of congestion in the network and consequently the jitter values.

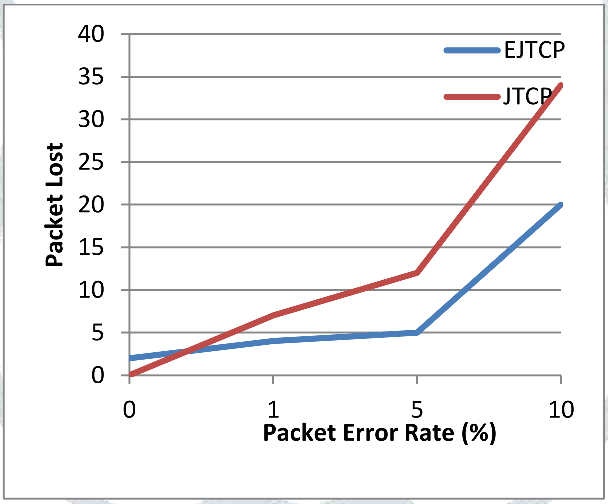

Fig. 5 Packet Lost v/s Packet loss rate

In Fig 5 we compare packet lost with packet error rate. Packet error rate is considered at the link layer. According to this, several corrupted frames can be recovered by link layer retransmission; hence, TCP layer sees a more reliable network, with a lower error rate.

VI. Conclusion

In this paper, we present a new EJTCP enhancement which considers the following characteristics i.e. throughput, delay, average jitter and packet lost. Our scheme should contain better loss differentiation scheme to alleviate misjudgments of loss events and effective congestion control mechanism to utilize better throughput and avoid causing network congestion.

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