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RARELD for Loss Differentiation and Reduction in Wireless Network

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Abstract- Data communication in networking is done through exchanging the data packets between the end points. The packet loss may occur in wired/wireless network during the data transmission. Losses in wired network are caused because of congestion in routers, but losses in wireless network may occur because of interference or network congestion. The main problem in wireless network compared to wired network is classification of losses. Transport protocols are response to handle the losses, if the transmission uses existing protocols like TCP, the losses are always classified as congestion loss by sender, causing reduced throughput. Main aim of this paper is to reduce the losses and improve the throughput of wireless network. This can be achieved by using both loss differentiation and reduction method together to achieve high throughput. RARELD (Receiver Assisted Random Early Loss Differentiation) is a loss differentiation method used to classify type of loss and also used to reduce losses in wireless network. In this method both sender as well as receiver are used to optimize loss occurred in network.

Index Terms- Wireless network, Transport protocol, LDA, Congestion control, ECN, Bandwidth estimation.

I. Introduction

Wireless technologies are playing a prominent role in the global internet infrastructure. They are ideal as internet access technologies, providing a convenient and cheap solution to the last mile problem but the performance of wireless network is low when compared to wired network. Losses in wired networks are mainly due to congestion in routers [13], because congestion is usually handled by dropping the received packets when the router waiting queues are full or nearly full. Hence, losses in wired networks can be seen as an indication of congestion. This is different in wireless networks where losses often occur for various reasons, for example due to interference or poor link quality (high distance between the base station and the mobile device). The performance degradation reported on wireless networks appears because TCP (Transport Control Protocol) [13], commonly used by Internet applications and initially designed for wired networks, classifies any data loss as a congestion loss; therefore it reacts by reducing the transmission rate. However, in wireless networks, losses are not necessarily caused by congestion.

Lot of applications used over Internet, with real-time audio and video streaming, can cope with a certain level of losses if they use TCP, then high reliability, congestion and flow control may achieve at the price of great latency. UDP (User Datagram Protocol), lacks congestion avoidance support and flow control mechanisms. SCTP (Stream Control Transport Protocol) provides some of the same service features of both message-oriented UDP and ensures reliable, in-sequence transport of messages with congestion control like TCP. Another promising protocol for these applications is DCCP (Datagram Congestion Control Protocol), recently standardized as RFC4340 [10], provide reliable connection setup, teardown, Explicit Congestion Notification (ECN), congestion control, and feature negotiation. It allows flow-based semantics like Transmission Control Protocol (TCP), but does not provide reliable in-order delivery.

DCCP also has the option of freedom choice for congestion control protocol. As described in [10], DCCP implements bidirectional and unicast connections of congestion-controlled unreliable datagram. DCCP is useful for applications with timing constraints on the delivery of data. DCCP also suffers from the same problem as TCP in wireless networks, meaning that any data loss is considered to be caused by congestion.

All the above protocols mentioned here, always consider the loss happens because of congestion, but losses may occur not only because of congestion errors but also wireless errors. So RELD with ECN and RTT used to differentiate the losses and this method is enhanced as RARELD with bandwidth estimation concept to reduce the losses and to increase the throughput of wireless network.

This paper is organized as follows, section 2 review the related work. Section 3 describes our new RARELD mechanism in detail. Section 4 presents the experiments of RARELD. Section 5 shows simulation and result analysis and we conclude this paper in section 6.

II. RELATED WORK

Several loss differentiation methods are existed to differentiate non-congestion loss from congestion loss. In this work we discuss the related approaches referred for RELD method to differentiate and to reduce the losses in network.

The first approach is ECN (Explicit Congestion Notification) referred from [3] with active queue management (e.g. RED) is used to control congestion in wired networks. RFC2760 [1] suggests that ECN can be used to distinguish congestion loss from wireless loss by diagnosing a loss. TCP-Eaglet uses ECN to discriminate the losses. When a loss is detected by sender, Eaglet reaction is based on whether it is in slow start or not. If the sender is not in slow start (i.e., it is in congestion avoidance phase), it considers the loss as a wireless loss and do not halve its congestion window size; Otherwise (i.e., the sender is in slow start) the loss is diagnosed as congestion loss and halves its congestion window size.

The second approach is based on ECN-D (Explicit Congestion Notification Differentiation) [16] used in SCTP to differentiate non-congestion losses from congestion losses. The optimal value of congestion window of SCTP source is taken from the ECN messages to maximize the throughput and maintain relatively small end-to-end delay. The simplified method provides an optimal congestion window and improves the total good put performance of SCTP. ECN-D SCTP differentiates non-congestion losses from congestion losses based on the congestion coherence concept. According to the time correlation between non congestion losses and congestion indications, two scenarios in which non congestion losses occur. a) Non-congestion losses occur when there are no ECN messages received for the current window of data at an SCTP source endpoint. b) Non-congestion losses occur when there are ECN messages received for the current window of data at an SCTP source endpoint.

The next approach is based on Loss labeling scheme [4] used to distinguish wireless losses from congestion losses. In this scheme, NewReno-LL uses P[CjC] (probability of congestion loss) and P[WjW] (probability of wireless loss) to differentiate the losses. Adaptive Flip Flop filter [9], is used for parallel estimation of RTT for every new ACK received in New Reno. Samaraweera [14] presents a method called Non Congestion Packet Loss Detection (NCPLD), to categorize the nature of the error. NCPLD uses the concept of the knee point of the throughput-load graph to measure the round trip delay. If the current (measured) round trip delay is less than the delay threshold at the knee point then the packet loss is assumed to be a wireless loss else it is assumed that loss caused by congestion (buffer overflow) errors, So New Reno-FF achieves a good-put (The rate of delivery of useful data measured by receiver) higher than New Reno, Westwood and New Reno-LL, for all values of P[CjC] and P[WjW].

The main approach of bandwidth estimation for loss reduction done on TCP Westwood relies, which is end-to-end bandwidth estimation [7] to discriminate the cause of packet loss (congestion or wireless channel effect) which is a major problem in TCP Reno. The key idea is to continuously measure at the TCP source the rate of the connection by monitoring the rate of returning ACKs. This estimation is used to compute congestion window and slow start threshold after three duplicate acknowledgments or after a timeout. TCP Reno, which blindly halves the congestion window after three duplicate ACKs, but TCP Westwood attempts to select a slow start threshold and a congestion window which are consistent with the effective bandwidth used at the time of congestion is experienced. TCPW provide improvements in throughput performance, as well as in fairness.

Another approach for loss reduction is RCP (Reception Control Protocol) [8] with general behaviors of TCP. RCP is explicitly used to controls the ACKs send back by receiver and to provide reliable delivery of data. In this function the receiver in RCP determines how much data the sender can send (via congestion control and flow control), and which data the sender should send (via reliability), the RCP receiver also takes total control over the bandwidth of connection and sends the congestion window size as a feedback to the TCP sender, from this feedback the sender takes corrective control action over the congestion in network.

The above shows that, Biaz and mBiaz performs well in low link rate, it has the problem when several streams share the same wireless link. Spike performs better in low network traffic but it has the problem to handle high network traffic. Samaraweera [14] represent that the RTT value decrease because of wireless errors but the assumption of this method is incorrect, because RTT value normally increased because of wireless errors, because the retransmission of MAC layer is not considered in this method so the assumption of RTT values are changed and used in my enhancement. TCP-Eaglet and ECN-D are not efficient differentiation schemes because they do not take into account congestion losses without ECN mark. Since our RELD method is used efficiently to differentiate the losses but this method not provides any solution to avoid losses in the network. So our proposed system RARELD used to avoid losses and to achieve high throughput for wireless network in my future work.

III. MECHANISM DESCRIBTION

In my proposed mechanism both sender as well as receiver is involved to control the congestion in the network.

A. Receiver Function

The receiver measures the bandwidth [12] according to the packet inter arrival interval and bandwidth is estimated as, $B_w = L/t_{int}$. Here B_w be the measured bandwidth, L be the data packet size and t_{int} be the packet inter arrival interval. The receiver measures the RTT based on the data packet arrival time.

The variable *receiver.rtt* is used to record the RTT. The algorithm can be described as follows.

i) When an ACK is sent, if *receiver.rtt* is zero, let *receiver.rtt* equal 1 and record the corresponding sequence (*rtseq*) of data packet (equals to the sum of the ACK sequence and the current congestion window). Otherwise, just send the ACK.

ii) If a data packet with a larger sequence number than "*rtseq*" is arrived in order and the new measured packet Inter arrival interval is larger than two times of pre-estimated packet inter arrival interval, set the new measured RTT to the value of *receiver.rtt*, and let *receiver.rtt* be zero.

iii) On each clock cycle, if *receiver.rtt* is not zero, *receiver.rtt* is added by 1.

The receiver window (*rwnd*) is defined from RTT values and bandwidth estimation $rwnd = B_w * RTT$. The advertised window is set from the minimum values of receiver available buffer and receiver window ad_wnd=min(r_abuf,rwnd).The receiver sent this advertised window as a feedback to sender then the sender takes corrective action by congestion avoidance algorithm to avoid packet losses in network.

B. Sender Reaction

In RARELD timely feedback from the receiver helps the sender to controls the congestion occurred in network. The sender first differentiates the loss [15] from the ECN and RTT values. The loss is classified as congestion if and only if,

1) ecn>0 or 2) n>0 and RTT<avg

Where *ecn* is the number of packets marked EC (Experienced Congestion), *n* the number of lost packets indicated by the received Ack, RTT the current RTT, *avg*+0.6dev are RTT threshold value. This function is used to identify whether, what type of loss (congestion/wireless loss) occurred in network. If network is in congestion then the sender decreases the congestion window by AIMD mechanism. If congestion is mitigated after one RTT, the sender will adjust the congestion window in the next window by using the receiver advertised window. When the sender receives an ACK, it will compare the current congestion window with the receiver advertised congestion window. If the receiver advertised window is larger than the sender's congestion window, and the difference is larger than a predefined threshold β, the sender will set the congestion window size to the receiver advertised window. Otherwise, the sender will ignore the receiver advertised congestion window by just performing the additive increase mechanism. As well as the advertised window send as a feedback from receiver used to define the congestion window size, this helps to maintain the AIMD mechanism in congestion avoidance stage, and the slow start stage can be eliminated also this function avoid bottleneck in network as well as to utilize available bandwidth of connection established.

IV. EXPERIMENT

Experiments are made using simulations with the network simulator ns-2 version 2.34 to differentiate the loss and to show the performance evaluation of RARELD in wireless network. Fig.1 shows the network topology used for all experiments. This topology allows the simulation of two groups of DCCP connections with different round trip times respectively initiated from src0 and src1.

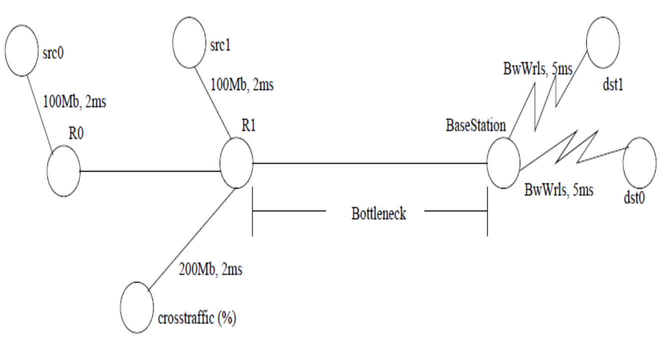

Fig.1 Network topology

These two groups of DCCP connections have to compete with a random cross traffic with bottleneck link capacity. Two wireless dedicated links connect the base station to dst0 and dst1. All links are labelled with bandwidth and propagation delay. The packet size is 500 bytes and the simulation time is 100 seconds. Experiments are conducted with different round trip time and bandwidths.

V. SIMULATION RESULTS

This section through extensive simulation, analyses RARELD classification rate and shows the performance gain of RARELD compared with RELD are shown in the fig. 3. For this analysis, two scenarios are used: the first scenario deals with the classification of packet loss and the second scenario deals with throughput gain. Results of this analysis are shown in fig. 2. In fig. 2 the RTT threshold (avg) is taken as (-1, 0, 1) and the numbers of losses are classified with RTT deviation values. In this analysis the congestion loss has an RTT value smaller

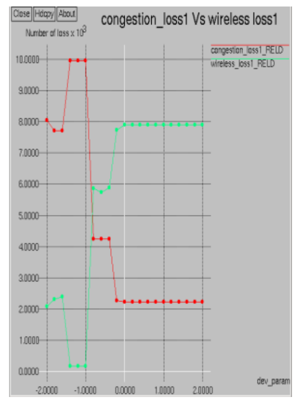

Fig. 2

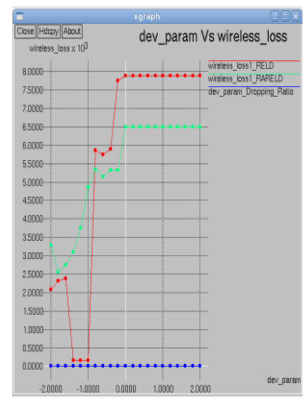

Fig. 3

than average RTT, while wireless losses are generally above the average RTT. Fig. 3 shows the classification rate of RARELD. In this RARELD threshold allows classifying congestion losses correctly between 80% and 98%.

RARELD is implemented in Network simulator-2.34 and used it to compare the performance of our mechanism with RELD and the simulation time taken for all scenarios described below are 100s. Fig. 5 shows the impact of packet error probability on the network throughput performance. The packet error probability is varied from 0 to 20. From fig. 5, it is observed that the throughput of DCCP RELD degrades drastically as the packet error probability increases. But our improved RARELD method doubles the throughput rate. So our algorithm appears to be robust to packet error by maintaining a high throughput.

This can be achieved by two ways, when the packet error probability is zero; congestion is the only factor that contributes to throughput degradation. Since RARELD can detect the packet loss for congestion earlier for the receiver's function and can use the receiver's advertised rate to adjust the congestion window, it can better utilize the bandwidth and have a better performance. As the packet error probability increases, packet loss due to packet error will cause RELD to perform unnecessary reduction of its congestion window and therefore its throughput decreases. Since RARELD can discriminate wireless packet drop from wired drops, it can reduce/eliminate the unnecessary throughput reduction due to wireless packet error so as to improve the throughput performance.

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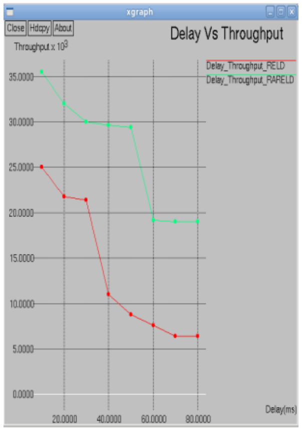

Fig. 4

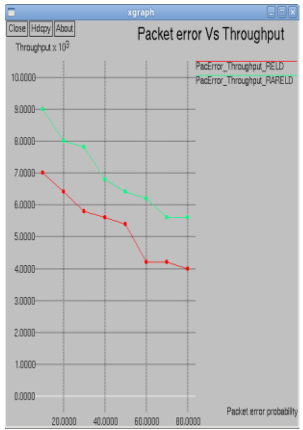

Fig. 5

In the second scenario, we study the effect of different wired delays on the throughput. The wired delay is varied from 10 to 100ms. The wireless bandwidth is 100 Mb/s, and the dropping ratio is 1.04008. Fig. 4 shows that the delay has a great influence on the throughput of RELD, while our mechanism is the most robust to increasing delay. The reason is that our mechanism has more accurate congestion control mechanism by extending the receiver function and can set an appropriate sending rate to avoid congestion.

Table: 1 shows the performance evaluation of RELD and RARELD; it demonstrates that our mechanism can improve the throughput performance further by about 20% by just reducing the impact of timeout. This is achieved by the observation that when congestion is about to occur.

RELD sim.time	**No. of. packet loss**	**Dropping ratio**	**Delay(ms)**	**Throughput**
25	55	4.30023	0.012326	41360.0
50	276	8.17536	0.012933	31615.0
75	477	8.75229	0.013092	23384.2
100	687	9.10416	0.013136	19702.1
RARELD sim.time	**No. of. packet loss**	**Dropping ratio**	**Delay(ms)**	**Throughput**
25	145	13.6278	0.0136512	19608.5
50	145	4.64744	0.0126204	23801.3
75	75	1.45518	0.0133841	25008.6
100	75	1.04008	0.0130322	25372.5

Table: 1 RELD and RARELD performance ratios

The available bandwidth for each connection is usually reduced gradually. Therefore, the receiver can judge the onset of congestion from the trend of the bandwidth variation and then reduce the receiver advertised congestion window. Since the sender will set its sending window size to the lesser of the congestion window and the receiver advertised window, network congestion will be alleviated when the window size is coming down, thus improving the network throughput performance.

VI. CONCLUSION

In this paper, we propose algorithm to make the DCCP capable over lossy links. Based on ECN and RTT values to differentiate the type of loss and the receiver function is added to this algorithm, to optimize the losses occurred in network. The advertised congestion window is send as a feedback from receiver to sender, which helps the sender to take timely action of congestion, occurred in network. The simplified method guarantees maximum utilization of the bottleneck link in heavy network traffic. Through theory analysis and network simulation, we find that the throughput of RARELD is higher than RELD. Since our algorithm is developed in context of DCCP, which is applicable to video streaming and gaming applications.

REFERENCES

[1] Allman M. and Dawkins S. "On going TCP research related to satellites" RFC 2760, Internet Request for Comments, 2000.

[2] Biaz S. and Vaidya N. "Discriminating congestion losses from wireless losses using inter-arrival times at the receiver" IEEE Symposium on Application-Specific Systems and Software Engr. and Techn., pp. 10–17, Richardson, TX. 1999.

[3] Cen S, Cosman P. C, and Voelker G. M."End-to-end differentiation of congestion and wireless losses". IEEE/ACM Transactions on Networking, 11 (5), 703–717, 2003.

[4] Barman D. and Matta I. "Effectiveness of loss labeling in improving TCP performance in wired/wireless networks" in ICNP (pp. 2–11). Los Alamitos: IEEE Computer society, 2002.

[5] Biaz S. and Wang X."Can ECN be used to differentiate congestion losses from wireless losses?" Tech. rep. CSSE04-04, Auburn University, 2004.

[6] Cen S. and Voelker G. M. "End-to-end differentiation of congestion and wireless losses" IEEE/ACM Transactions on networking, 2003 .

[7] Casetti C. et al "TCP Westwood, Bandwidth estimation for enhanced transport over wireless links" in Proc. 7th ACM Annu. Int. Conf. Mobile Comput. Netw., Rome, Italy, pp. 287–29, 2001.

[8] H. Hsieh et al., "A Receiver-centric transport protocol for mobile hosts with heterogeneous wireless interfaces" in Proc. 9th ACM Annu. Int.Conf. Mobile Comput. Netw., San Diego, CA, 2003.

[9] Kim M. and Noble B. "Mobile Network Estimation". In Mobicom 2001: The Seventh Annual International Conference on Mobile Computing and Networking, Rome, Italy, 2001.

[10] Kohler, E., Handley,M.., & Floyd, S. "Datagram congestion control protocol (DCCP)". RFC 4340, 2006.

[11] Kai Shi, Yantai Shu, Oliver Yan, and Jiarong Luo "Receiver-Assisted Congestion Control to Achieve High Throughput in Lossy Wireless Networks" IEEE Transactions on nuclear science, vol. 57, no. 2, April 2010.

[12] Tobe Y., Tamura Y., Molano A., Ghosh S., and Tokuda H. "Achieving moderate fairness for UDP flows by path-status classification". In LCN: IEEE Local computer networks (p. 252). 12) Postel, J. (1980).

[13] Postel, J. "Transmission control protocol. RFC 793", 1981.

[14] Samaraweera N. "Non-congestion Packet Loss Detection for Improving TCP Error Recovery using Wireless Links" In IEEE Proceedings Communications, 1999.

[15] Wassim Ramadan, Eugen Dedu, Dominique Dhoutaut and Julien Bourgeious "RELD,RTT ECN Loss Differentiation to optimize the

performance of transport protocols on wireless networks" Springer Science + Business Media, 2011.

[16] Ye G. and Lee M.J. "Improving stream control transmission protocol performance over lossy links" IEEE Journal on selected areas in communications, 2004.

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