CHAPTER 7

CONCLUSION

7.1 INTRODUCTION

This chapter starts with a brief review of the research works of all the previous chapters and summarizes the major contributions of this research. Following these sections, a discussion on the research work is presented. Finally this chapter ends with some future work in the related area.

7.2 REVIEW OF RESEARCH

This thesis provided an overview of congestion avoidance algorithm and highlighted the issues of congestion avoidance algorithm. A survey on congestion avoidance algorithm had been conducted and described. From the survey, it is found that packet loss differentiation is essential to improve the performance of congestion avoidance algorithm because the packet loss in heterogeneous wired-wireless network may be due to congestion or wireless transmission error. The packet loss differentiation algorithm takes the estimated RTT as one of the parameters to predict the cause for the packet loss. So a survey on RTT estimation and packet loss differentiation algorithm had been conducted and described in this thesis. As a result of these survey, we take three parameters 1) RTT 2) Number of backlogged packets and 3) Received signal strength, to differentiate the congestion packet loss from non-congestive packet loss.

The variation in RTT is caused by the sudden changes in the data traffic, congestion in the bottleneck link, temporary link failure in the wireless link and rerouting of packets due to congestion. Since the delay-based congestion avoidance
algorithm depends on variation in RTT for controlling the sending data rate, the estimation of RTT is very important. Therefore we proposed a RTT estimation using ARIMA(2,1,1) model (ARTT) to estimate the sudden changes in the RTT.

The number of backlogged packet in the network increases as the data traffic in the network increases. As the backlogged packets increases, the RTT of a packet also increases which leads to variation in the RTT. The variation or sudden changes are estimated by ARTT better than the other estimation techniques. Next the ARTT is used to estimate the delay caused by the backlogged packets in the network. Since the ARTT estimates the variation in RTT and delay caused by backlogged packets, we used ARIMA(2,1,1) model-based RTT estimation to predict the number of backlogged packets. It is then used to predict the cause for the packet loss. Therefore we proposed a packet loss differentiation algorithm which uses the estimated number of backlogged packets. Based on the cause of packet loss, the congestion window size is increased/decreased/unchanged to mitigate the data rate at sender.

The implicit information such as packet delay, available bandwidth etc is sometimes not adequate to differentiate the non-congestive packet losses from congestive packet losses because of unpredictable behavior of wireless node in the heterogeneous wired-wireless network. Therefore we used explicit information i.e. Received Signal Strength (RSS) of wireless receiver, to differentiate the non-congestive packet loss from congestive packet loss. In order to use the explicit information in the congestion avoidance algorithm, we used cross layer signaling methods.

7.3 CONTRIBUTIONS

7.3.1 ARIMA Model-based RTT Estimation (ARTT)

Since we proposed a delay-based congestion avoidance algorithm, the variation in RTT plays a vital role in controlling the sending data rate at sender. The sudden change in RTT is caused by
1. When the buffer gets filled with packets, the queuing time increases thereby the sudden change in RTT is happened.

2. When the data traffic increases abruptly at the bottleneck link, then the sudden changes in RTT is happened.

3. When there is a temporary failure in wireless link, then there is a sudden change in RTT.

In order to estimate the sudden changes in RTT, we analyzed the RTTs measured through the passive measurement and proposed an ARIMA (2,1,1) model-based RTT estimation (ARTT) in this research work. The ARTT is evaluated through the RTTs measured through the active measurement. The ARTT takes the RTTs of recently acknowledged three packets to estimate the RTT. The recent RTTs are essential for estimation of sudden changes in the RTT which are predicted by the proposed ARTT better than other estimation algorithms. Therefore we make use of this feature to differentiate the cause for packet loss. The experimental results show that the performance of proposed RTT estimation provides better performance than other estimation techniques.

7.3.2 Packet Loss Differentiation Algorithm using ARTT (LDA)

As the data traffic increases in the network, the number of backlogged packets also increases which leads to increase in RTT and causes congestion in the network. Since the ARTT estimates the sudden changes in RTT using the RTTs of recently acknowledged three packets, first we used it in estimating the backlogged packet in the network. Next the estimated number of backlogged packets is compared with two threshold values (α and β) for differentiating the congestive packet loss from non-congestive packet loss. Therefore the proposed packet loss differentiation algorithm (LDA) makes use of ARTT and proposed backlogged packet estimation to determine the three states of the network: 1) Congestion-free state, 2) Congestion loss state, 3) Non-congestion loss state.
In congestion-free state of network, the available bottleneck bandwidth is probably under-utilized and there is no possibility for packet loss in this state. In congestion-loss state, the available bottleneck bandwidth is probably over-utilized and there is a possibility of packet loss due to congestion. In the non-congestion loss state, the available bottleneck bandwidth is in between these two states. Any packet loss in this state is due to the reason other than congestion. We evaluated the proposed algorithm in heterogeneous wired-wireless network and compared with normalized throughput gradient algorithm (NTG), vegas prediction algorithm (Vegas), veno prediction algorithm (Veno) and robust end-to-end loss differentiation algorithm (RELD). The overall performance proposed algorithm achieves better than other algorithms.

7.3.3 Congestion Avoidance Algorithm using ARTT (CA-ARTT)

Based on the above said three state of the network, the congestion window size is increase/decrease/unchanged. Therefore the proposed congestion avoidance algorithm controls the sending data rate at the sender by checking these three states of network. If it is in congestion-free state, the congestion window size is increased in order to effectively utilize the available bottleneck bandwidth. If it is in congestion loss state, the congestion window size is decreased in order to avoid the further congestion in the network. If it is in non-congestive loss state, the congestion window size remains unchanged because the available bottleneck bandwidth is efficiently used and any packet loss in this state is due to the reason other than congestion. The proposed congestion avoidance algorithm is evaluated in the heterogeneous wired-wireless network with different network environments. The different network environments are simulated by varying the RTT from 100msec to 400msec, packet size ranging from 128bytes to 2048bytes, BER varying from $10^{-2}$ to $10^{-5}$ and bandwidth asymmetric network. The proposed algorithm achieves 10% to 15% performance improvement than the TCP-Vegas in the heterogeneous wired-wireless networks and in case of asymmetric network, it provides up to 4 times improvement in throughput.
7.3.4 Enhancement of CA-ARTT using Cross Layer Approach (CA-CLARTT)

In order to enhance the performance of CA-ARTT, cross layer information is used in the heterogeneous wired-wireless network. The received signal strength (RSS) of receiver is taken as cross layer information which is sent to the sender. The sender takes this cross layer information to know status of wireless link at the receiver side. Since the cross layer information is piggybacked with ACK packets, the communication overhead is negligibly small. Based on the cross layer information obtained from receiver, the sender predicts the non-congestive loss state. Therefore the enhanced algorithm provides 10\% to 25\% improvement in throughput in heterogeneous wired-wireless network.

7.4. RESULTS AND DISCUSSIONS

In order to analyze the RTTs and design the RTT estimator, the RTT data sets are collected from CAIDA (Center for Applied Internet Data Analysis). They measured the RTT using passive measurement approach and made available for researchers to analyze the behavior of RTT in the Internet. We collected 30 RTT data sets, each containing 120 RTTs measured at equally spaced time interval. We analyzed the RTT data sets using autocorrelation and partial correlation function and designed a RTT estimation using ARIMA(2,1,1) model (ARTT). We compared the proposed ARTT with ARIMA(1,1,1) and ARIMA(2,1,2) model and found that ARTT estimates the one-step-ahead forecast of RTT better than other two models. The performance of ARTT is measured through the Mean Square Error (MSE), Root Mean Square Error (RMSE) and Normalized Mean Square Error (NMSE). It is observed from the results that RTT estimation using ARIMA(2,1,1) model has low MSE, RMSE and NMSE as compared to other RTT estimations. Second, the RTTs measured from our University network using active measurement approach are taken to evaluate the performance of ARTT. It is found from the results that ARTT has less MSE, RMSE and NMSE as compared to other RTT estimations. Third we compared the ARTT with average RTT and Smooth RTT (SRTT) and found that the proposed ARTT estimates the sudden changes in the RTT better than other estimations. This is
because the proposed ARTT takes the RTTs of recently acknowledged three packets for estimation. But SRTT takes the weighted average of all previous RTTs and current RTT so that it is biased toward the history of RTTs which is not suitable for detecting the sudden changes in RTT. The RTTs of recently acknowledged packets are very much suitable for differentiating the cause for the packet loss.

We make use of ARIMA(2,1,1) model-based RTT estimation for differentiating the congestive packet loss from non-congestive packet loss by estimating the number of backlogged packets in the network. The proposed loss differentiation algorithm (LDA) provides improved performance over other LDAs in two network environments i.e. Single TCP flow and Cross traffic TCP flow. The performance of proposed algorithm is measured through the four metrics: 1) percentage of accuracy of congestion loss prediction ($A_c$), 2) percentage of accuracy of wireless loss prediction ($A_w$), 3) percentage of accuracy of LDA ($A_t$) and 4) percentage of misclassification ($M_t$). In presence of single TCP flow and packet loss rate varies from 0% to 10%, the $A_c$ of proposed LDA is consistently from 87% to 100% , the $A_w$ of proposed LDA is from 40% to 90%. $A_t$ of proposed LDA is 67% to 87.6% and $M_t$ is in between 10.3% and 32.5%. Similarly in the presence of cross traffic TCP flow, the overall performance of the proposed LDA is maintained from 55% to 89% in differentiating the congestion packet loss and wireless packet loss.

Since the proposed LDA takes the ARTT and delay due to backlogged packets for differentiating the congestive packet loss from non-congestive packet loss, the performance of LDA is improved. This is because the sudden changes in RTT are estimated by the proposed ARTT better. The main issue of a loss differentiation algorithm is that sometime it predicts the wireless packet loss as congestion packet loss and vice versa. This type of prediction is called misclassification. For a good LDA, the percentage of misclassification must be as low as possible. The percentage of misclassification is low as compared with other LDAs. Next the main advantages of the proposed LDA are 1) It is a simple algorithm and requires small changes at sender side TCP implementation only, 2) It does not require any coordination with the receiver side TCP other than acknowledgement, 3)
Only the RTTs of recently acknowledged three packets are required for estimation of RTT.

The proposed congestion avoidance algorithm (CA-ARTT) is evaluated using two networks (Heterogeneous wired-wireless network and Bandwidth asymmetric network) with different network environments such as varying RTT from 100msec to 400msec, varying BER from $10^{-2}$ to $10^{-5}$. The performance of CA-ARTT is measured in terms of throughput and is tabulated in Table 7.1. The performance of CA-ARTT is improved with respect to throughput using different network environmental settings because the proposed ARTT and LDA in the HWWN. In case of bandwidth asymmetric network, the performance of CA-ARTT is improved for the normalized asymmetric factor (K) ranging from 2 to 16. In case of $K = 32$, the bandwidth of reverse link varies from 2.5Kbps to 20Kbps as shown in Table 5.11. Since it is low bandwidth in the reverse link, the average throughput of CA-ARTT is low as compared to TCP-Vegas for $K=32$.

Table 7.1 Comparison of CA-ARTT and TCP-Vegas in terms of Throughput

<table>
<thead>
<tr>
<th>Type of Network</th>
<th>Network Environmental settings</th>
<th>% of throughput improvement compared with TCP-Vegas</th>
</tr>
</thead>
<tbody>
<tr>
<td>Heterogeneous Wired-Wireless Network (HWWN)</td>
<td>In the presence of single TCP flow</td>
<td>4.3% to 10.1%</td>
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<tr>
<td></td>
<td>In the presence of cross traffic TCP flow</td>
<td>12.8% to 24.4%</td>
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<tr>
<td></td>
<td>In the presence of cross traffic TCP flow and BER set to $10^{-2}$ bit/sec</td>
<td>11.1% to 22.8%</td>
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<tr>
<td></td>
<td>In the presence of cross traffic TCP flow and BER ranging from $10^{-5}$ to $10^{-2}$ bits/sec</td>
<td>5.9% to 13.2%</td>
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<tr>
<td></td>
<td>In the presence of cross traffic TCP flow, packet size ranging from 128bytes to 2048bytes and packet loss rate set to 0%</td>
<td>7.9% to 14.4%</td>
</tr>
<tr>
<td>Heterogeneous Wired-Wireless Network (HWWN)</td>
<td>In the presence of cross traffic TCP flow, packet size ranging from 128bytes to 2048bytes and packet loss rate set to 1%</td>
<td>6% to 12.6%</td>
</tr>
<tr>
<td>In the presence of cross traffic TCP flow, RTT ranging from 100msec to 400msec and packet loss rate set to 0%</td>
<td>0% to 31.8%</td>
<td></td>
</tr>
<tr>
<td>In the presence of cross traffic TCP flow, RTT ranging from 100msec to 400msec and packet loss rate set to 1%</td>
<td>0% to 18.5%</td>
<td></td>
</tr>
<tr>
<td>Bandwidth Asymmetric Network</td>
<td>In the presence of single TCP flow and packet loss rate set to 0%</td>
<td>Performance is same as TCP Vegas</td>
</tr>
<tr>
<td>In the presence of backward traffic TCP flow and packet loss rate set to 0%</td>
<td>1.5 to 4 times improvement for normalized asymmetric factor varying from 2 to 16. The throughput of CA-ARTT is degraded by 0.39 times</td>
<td></td>
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<tr>
<td>In the presence of backward traffic TCP flow, BER ranging from $10^{-5}$ to $10^{-2}$ bits/sec</td>
<td>0.5 to 1.5 times improvement for normalized asymmetric factor varying from 2 to 16. The throughput of CA-ARTT is degraded by 0.6 to 1.1 times</td>
<td></td>
</tr>
<tr>
<td>In the presence of backward traffic TCP flow, packet size ranging from 128bytes to 2048bytes and packet loss rate set to 1%</td>
<td>0.5 to 2.65 times improvement for normalized asymmetric factor (K) varying from 2 to 16. K=32, The throughput of CA-ARTT is degraded by 0.01 to 0.6 times for K=32</td>
<td></td>
</tr>
<tr>
<td>In the presence of backward traffic TCP flow, RTT ranging from 100msec to 400msec and packet loss rate set to 1%</td>
<td>4.3% to 65.3% improvement for normalized asymmetric factor (K) varying from 2 to 16. The throughput of CA-ARTT is degraded by 10.7% to 52.11% for K=32</td>
<td></td>
</tr>
</tbody>
</table>
In order to improve the performance CA-ARTT in differentiating the congestive packet loss from non-congestive packet loss, we used explicit information through a cross layer signaling method. The Received Signal Strength (RSS) of receiver is taken into account for differentiating the packet losses in HWWN. The performance of enhanced CA-ARTT called CA-CLARTT is measured in terms of throughput and found better than CA-ARTT and TCP-Vegas. The cross layer information i.e. RSS is piggybacked with ACK packet so that the communication overhead is negligible small. Since the wireless link quality in terms of received signal strength is informed to the sender, the non-congestive packet losses are determined at the sender better than TCP-Vegas and the congestion window size is adjusted accordingly. Therefore the performance of CA-CLARTT is improved by 15% to 25%.

7.5. FUTURE WORK

In this thesis, we proposed a congestion avoidance algorithm using ARIMA(2,1,1) model-based RTT estimation and received signal strength (RSS). The smooth round trip time (SRTT) is presently used to calculate the retransmission timeout (RTO) which is estimated as 4 times of SRTT. In future it can be calculated using ARIMA(2,1,1) model-based RTT estimation (ARTT). Next we used two thresholds $\alpha$ and $\beta$ for determine the state of the network (congestion-free state, congestion loss state and non-congestion loss state). Since the behavior of network is dynamically changed time to time, the value of $\alpha$ and $\beta$ can also be changed dynamically. Further, it is hard to describe the network traffic by any single model like ARIMA(2,1,1) because the RTTs have non-linear and time varying behavior in the Internet. Therefore it is necessary to design a set of models or dynamically identifying the order of AR and MA to cover all traffic delay patterns.

7.6 CONCLUSION

This chapter provided an overall review of this research work and a brief summary of proposed congestion avoidance algorithm which uses ARIMA model-based RTT estimation and received signal strength. Following to that, some of the future works that can be done related to this research are discussed.