FLoadAutoRED: An Active Queue Management Scheme to Prevent Congestion in a Dynamically Varying Traffic in IP Networks

CHAPTER 8

CONCLUSION AND FUTURE SCOPE

8.1 SUMMARY AND CONCLUSION

In recent years, real-time audio and video applications over IP networks, e.g. IP telephony, teleconferencing, have grown enormously. These applications enable efficient communications through IP networks. While different from the traditional applications, these real-time audio/video applications are sensitive to end-to-end delay and delay variation, but can tolerate occasional loss. IP networks today only provide the best-effort service, and such a service does not make any guarantees for delay, jitter or loss of individual packets. Hence it is a challenge to develop successful real-time audio/video networking applications over best effort IP networks. In addition, the quality of real-time traffic deteriorates to intolerable levels when it is delivered over a moderately congested link. This characteristic does conflict with the goal of fully utilizing the bandwidth of the network. In recent years there have been numerous AQMs to introduce QoS into IP networks, although most of them have not failed. The thesis presents a method of improving the QoS for dynamically varying traffic over IP network.

AutoRED, an active queue management mechanism helps to control the average queue size within a network for traffic with cooperative flows. However, AutoRED is widely investigated in the context of UDP traffic and it has been realized that it faces the problem of higher queuing delay. Hence this deteriorates the quality of the applications in the network.

The thesis presents a thorough study of AutoRED AQM to improve its QoS in terms of throughput, delay and loss. First, extensive simulation results show that the AutoRED mechanism is able to regulate queuing behaviour for cooperative traffic with QoS requirements. This is especially made possible only with larger queue sizes. Because the average queue size is determined by the queue state distribution, burstier traffic causes a longer queue state tail thus resulting in a longer average queue size. The packet arrival rate as a metric helps to decrease the probability of the queue being large and hence lowers both the mean queue size and then the mean queuing delay. Secondly, a detailed evaluation is given of how AutoRED requires controlling the delay, jitter and loss. This study indicates that by reducing the probability of larger queue sizes in AutoRED, FLoadAutoRED keeps the mean queuing delay within an acceptable range, and maintains the jitter at a stable level. This delay control can also help to reduce the probability of late packet delivery. In addition, consecutive loss, which is harmful to the real time traffic and is more difficult to recover, has been decreased with FLoadAutoRED due to its early drop behaviour.

Simulations study has demonstrated that FLoadAutoRED is able to improve the QoS of dynamically varying traffic in IP networks. The experimental results show that the average queue size of traffic increases with queue maximum threshold. As for as loss rate is considered the experiments show that the loss rate decreases as the maximum threshold increases, but this change in loss rate is not as obvious as the increase in average queue size. Thus the loss rate of the bursty traffic under a dynamic maximum threshold is still to be improved. As the required QoS is not met, the performance is gained by considering both the load and queue size as congestion indicators.

The use of FLoadAutoRED in managing dynamic traffic can control queue size, delay, and minimize the loss whatever the load may be. When the link is congested and overloaded, the control of queue size, delay and effective loss are especially obvious when compared with AutoRED. The proposed approach provides a good solution to performance degradation of varying traffic under congested network conditions.

The thorough evaluation of FLoadAutoRED's performance improvements for varying traffic conditions can be considered as the first successful step in the whole development of improving QoS by active queue management to inelastic UDP traffic. In the thesis several topics related to the congestion prevention and the fair bandwidth allocation are studied. As the

Internet evolves, the congestion prevention mechanism is facing many challenges. The study indicates that some issues are faced when TCP flows exist along with the misbehaving flows in the IP network. First, the loss effect in IP networks and its impact on the TCP flows in these networks. It turns out that most of the high-speed UDP flow variants result in severe unfairness. It makes TCP flows passing through multiple congested links only get extremely moderate share of the bandwidth, in the presence of high packet losses among both the flows.

An effective way to address this problem is to employ AQM schemes. The proposed AQM has good QoS requirements for dynamically varying traffic, but for traffic characteristics, such as with varying constant bit rate allows the proposed algorithm’s control behaves in subtly different ways. When the link is not overloaded with low UDP rate, the performance of the traffic is stable. However, once a link is overloaded with high UDP rate, the delay of traffic becomes very long, resulting in a high probability of late delivery and hence effective loss. The QoS of higher rate traffic with such long delays deteriorates rapidly. Therefore, in case of higher rate UDP traffic, the loss increases heavier worsening the quality. Moreover, these high-speed UDP flows are generally delay sensitive and need preferential treatment in order to satisfy a desired level of Quality of Service constraints.

8.2 FUTURE SCOPE

The current Internet supports applications with varying QoS requirements, as the emerging interactive applications and file transfer application requirements are delay sensitive and throughput sensitive respectively. Therefore an AQM that can distinguish difference between these applications is the requirement and the service may be provided to all applications with varying loss and delay. The future scope thus can be totally an application dependent solution to packet loss and delay.