Technical Computation and Communication Delay in Distributed System

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ABSTRACT
The effect of the packet loss is decreases the QoS of real-time applications. It also hampers the effectiveness, accuracy of the data transmission technique, scheduling and processing time of the technique. So, it is essential to consider the factor that forms the packet loss.
The main causes of the packet loss are the delay and congestion that forms when data is flowing on the Internet. There are several techniques are designed and developed to come out from the delay and congestion. In this paper, the focus is on the techniques that are designed specially to avoid the congestion and delay situations that exhibit adaptive approach for providing efficient and effective performance in the distributed system.

Keywords
Congestion, Delay, Distributed System, Internet Traffic, Multicast System, Multigroup Network.

1. INTRODUCTION
In the Internet, usually large volume of data is transferred from source towards the destination. Many times in an overlay multicast system, streaming contents are delivered towards user. For that it uses user's upload capacities. Therefore, the quality of the delivery of the streaming media is cheap to a large population of the users. While transferring the data over the internet, the data is divided into number of packets and then transferred through the different scheduling techniques in the variety of the environment with the path diversity for subsequent packets. It uses several protocols that were designed for multigroup networks.

1.1 Resources of the Delay in Distributed System
In the literature, it is seen that a majority of the packet loss occurs due to the result of the congestion in the links, form a bottleneck due to the high rate real time flows, lack of control services of the traffic transmission, heavy load on the traffic and low bandwidth. Many applications like multimedia are very sensitive to packet loss, where retransmission of packet is not a feasible option. Therefore, it is necessary to find out congestion in order to take appropriate rate control actions. [1].
In a multigroup network, many end host join in several multicast group. If one host belongs to multiple groups, then the end host has to process multiple entering flows simultaneously due to the group flows are usually high-rate real-time flows. Therefore, it is possible to form bottleneck in multiple group. It invites the multicast delay and compromised scalability performance [2]

1.2 Congestion in Distributed System
The TCP congestion control scheme is perform well to improve the congestion and sharing the bandwidth from the end-to-end point of view. It is also well known for TCP's additive increases multiplicative decreases due to queue length oscillation. In the mild oscillation, end-to-end application may face the large delay jitter due to the queuing delay variations. In fast oscillations the link frequently suffers from under utilization and packet drops in burst [3].

2. TECHNICAL CRAM OF THE TECHNIQUES FOR THE DELAY
In this section, the cram of the technique is done from the different point of views such as congested link, rate control bandwidth and so on.

2.1 Reducing Queue Oscillation at a Congested Link
At the congested link, due to the queue length oscillation, many properties are burst for instance large delay jitter, underutilization of the link and packet drops. The queue management scheme is the main reason of the oscillation. It determines the drop probability of the current traffic that is based on the current traffic without consideration of the impact of that drop probability on the future traffic.
TCP's congestion control scheme is important to improve congestion and sharing bandwidth from an end-to-end point of view. On the other hand, it is also well known that TCP's additive increase / multiplicative decrease (AIMD) behavior may cause queue length oscillation. When mild oscillations are occurred in end-to-end application, it is observed that there is large delay jitter due to queuing delay variations. When severe oscillations are happen, the link repeatedly suffers from underutilization and packet drops in burst [3]. Therefore, a new queue management scheme for a stable queue length. In this scheme, it first quantitatively estimate the amount of TCP traffic and then the average RTT from the amount of traffic variation and time interval from congestion notification to its reaction. Using this estimation it calculates the drop probability that expected to keep the arrival rate stable.

2.2 Delay Asymptotic and Scalability for P2P live Streaming.
In [4] define a benchmarking metrics for the performance evaluation of overlay multicast systems. It considers the two problems related to the playback delay. In the first, how fast the probability of missing a packet decrease as a function of the playback delay. Another one is how fast should the playback delay be increased. It maintains the probability of missing a packet unchanged as the overlay's size increases. Through the derived bounds, it defines the factors that influence the system scalability.

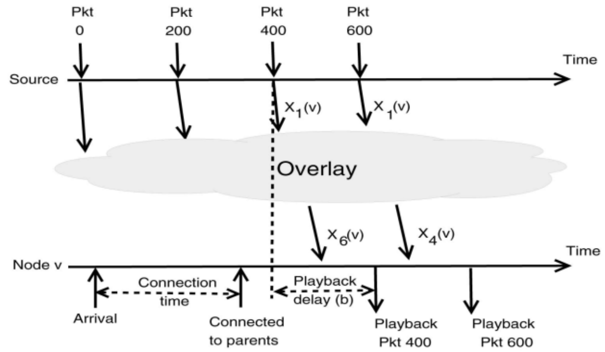

Fig.1. The playback delay and the time needed to connect to the overlay.

The tail behavior of the per-hop delay distribution determines the evolution of the packet missing probability as a function of the playback delay as shown in Fig.1. The back-off scheme is used for retransmission. If the per-hop delays are LT, then the packet missing probability shows an asymptotically at least exponential decrease as a function of the playback delay, while it exhibits a heavy-tailed distribution otherwise.

To assess the structure of the overlay, one has to look at the scaling of the playback delay with respect to the overlay size. In an overlay, in which the distance of the peers from the source is a logarithmic function of the overlay's size, the playback delay does not have to be increased faster than the logarithm of the overlay's size to keep the packet missing probability constant.

2.3 Worst-case Delay Control in Multigroup Overlay Networks

The end host multicast is now famous as a substitute to inter-domain Internet Protocol (IP) multicast. End hosts may join in a number of multicast groups in a multigroup network. When one end host belongs to more than one group, at the same time the end host has to process multiple entering flows. Usually the group flows are high rate real time flows; the end hosts that join in multiple groups are prone to become bottleneck. It sustains unacceptable multicast delays and compromised scalability performance.

One of the most popular ways to recover from the bottlenecks is to design capacity aware EMcast protocols. That assigns the direct child members for each end host has enough capacity to output capacity. In this way, the end host has enough capacity to output the received packets to all its direct child members. It will not fall in a communication bottleneck. On the other hand, such bottleneck is achieved at the cost of increasing the lengths of the multicast path from the source to the group receivers.

In [2] present a novel simple adaptive control algorithm implemented in the overlay network. This algorithm adaptively uses the novel regulators to free bottlenecks without increasing the lengths of multicast paths. When the network traffic becomes heavy, the forwarding of flows at each end host is controlled in turn based on the current network state with the regulator.

2.4 Influence of start-up costs in scheduling divisible loads

In research area of computation and communication to minimize the processing time of the jobs incoming at any computing site is the main objective in the distributed computing system. So, the need to develop the efficient scheduling algorithm that is minimizes the total processing time of the jobs.

In [5] presents a modified model of the theory of divisible load scheduling. With previous linear model and overheads, first derive the closed form solution for the optimal processing time. It exhibits the effect of the overheads on the processing time and then identifies certain properties. Then, identify the important property so that it gives the tree structure of the heterogeneous tree network. Then by changing the order of the load distributed among the processors and observed that the processing time is affected. By using the concept of the sequencing, it identifies the optimal sequence with the special property. When such property exists, it finds the combined effect of the overhead components. If it does not exist a greedy approach is used to identify the sequence that generates a suboptimal solution that matches most. Then it gives an integer approximation algorithm, which generate integer load fractions. That is within a radius which is not more than the sum of computation and communication time w.r.t. the optimal solution.

2.5 Packet Loss Prediction

Authors present a design of framework for accurate packet loss prediction and congestion detection, which is useful for congestion avoidance and proactive error recovery. It is applicable to a wide range of application. Such as traffic engineering and provisioning, wireless loss discrimination and Forward Error Correction (FEC) [1][6].

A framework of a Packet Loss Predictor is a novel mechanism. It predicts the packet loss in real-time audio streams. It observes the delay variation and trends. It is based on the:

- Determining certain metrics for measuring the delay variation characteristics from the on going traffic.
- Combine them with weights based on their importance and
- Deriving a predictor value as the measure of packet loss probability.

3. CRITICAL ANALYSIS OF THE TECHNIQUES

In this section different factors of the techniques are critically analyzed and presented.

3.1 Advantages of the Queue Oscillation at Congested Link

This scheme is effective. It reduces the queue oscillation and realizing stable queue length. The main advantages of the scheme is that the queue length can be reduces without degrading the utilization and fairness among TCP flows can be improved with a stable drop probability. It also improves fairness among TCP flows due to the stable drop probability. It also reduces delay jitter and minimizes the queuing delay without loss of link utilization [3].

3.2 Properties of the P2P Live Streaming

In the model of a push-based overlay, the asymptotic scaling properties hold, using various retransmission schemes and FEC. Simple overlay management solutions is provided good scaling properties. It provides metrics to assess the scalability of peer-to-peer streaming systems and give a basic understanding of the dependencies between streaming performance, overlay, and data transmission control [4].

3.3 Advantages of Adaptive Control Algorithm

The main advantage of this algorithm is the worst-case delay improvement of the regulator over the previous one and rate threshold. The possible bottleneck in multigroup network can be avoided without increasing the lengths of the multicast paths [2].

3.4 Performance of the Load Distribution Technique

It minimizes the processing time when the load distribution sequence follows the order where the processors speed decrease. It gives an acceptable performance even though it is applied to a heterogeneous set of processors. The upper bound of the suboptimal solution generated by the algorithm is lies within a radius given by the sum of the computation and communication delays [5].

3.5 Accuracy and Efficiency of the Proactive Approach

The predictor feedback makes the framework superior to mechanisms with static and reactive feedback. This approach gives efficiency and accuracy [1].

4. CONLUSION AND FUTURE SCOPE

In this paper, the performances of the delay and congestion avoidance techniques are discussed analyzed. It is seen that, these methods are easily improve the worst case delay, rate threshold, minimizes the processing time, delay jitter and queuing delay without loss of link utilization.

One can easily modify the metrics and weight factors using statistical methods and also extend the concepts of the loss predictor to DiffServ, Overlay and Multicast frameworks.

5. REFERENCES

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