Rate-based Congestion Control Mechanism for Multicast Communication

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Abstract: This paper presents a multicast congestion control mechanism using combination of both sender and receiver-based approach for multicast communication. We focus on mechanism that the protocol can adapt with network condition by adjusting the transmission rate (RAP) based on ACK/NACK and detect a loss based on NACK transmitted by receivers. We add ACK/NACK suppression mechanism at receivers to avoid feedback implosion at sender. Our simulation result shows that packet flow with small path delay is able to increase transmission rate aggressively due to small feedback delay and the number of packet loss is decrease as an increasing the path delay. By adding ACK/NACK suppression mechanism at receiver, packet loss can be reduced as well as feedback implosion can be avoided.

Key-words: Multicast, Rate-based Congestion Control, Feedback Delay, Packet Loss, Feedback Implosion, Suppression Mechanism

1. Introduction

The bulk of the traffic on today's network is mostly unicast. A separate copy of the data is sent from the source to each client that requests it (one-to-one). Networks also support broadcasting. When data is a broadcast, a single copy of data is sent to all clients on the network (one-to-all). If the same data needs to be sent to only portion of the clients on the network, both of these methods waste network bandwidth. Unicast wastes bandwidth by sending multiple copies of the data. Broadcast wastes bandwidth by sending the data to whole the network whether or not the data is wanted. Broadcasting can also needlessly slow the performance of client machines. Each client must process the broadcast data whether or not the broadcast is of the interest. Multicasting takes the strengths of both of these approaches and avoids their weaknesses. Multicasting sends a single copy of data to those clients who request it (one-to-n, n is varied from zero to all). Multiple copies of data are not sent across the network, nor data sent to clients who do not want it.

Many emerging application in the internet need multicasting services such as Internet TV, teleconferencing and distance learning. Today, Most internet multicast application do not support end-to-end congestion control mechanism, hence, this leads to endanger stability of the network in terms of fair sharing a connection among other traffic sources which is mainly TCP traffic. Rate-based congestion control is a best alternative in order to achieve fairness among connections in the Internet in particularly in multicast communication.

Our research work aims to investigate the application of the rate adaptation protocol (RAP) [1] as part of reliable multicast congestion control scheme and create self-develop RAP extension to Multicast protocol for reliable multicast communication.

2. Multicast Congestion Control Issues.

The main goal of congestion control is to let the application use the network efficiently by being responsive and adaptive to the network condition where congestion is occurred. There are two crucial issues must be considered when designing a multicast congestion control protocol [2]:

TCP-friendliness

Today, a dominant portion of Internet traffic is a TCP-based traffic (e.g. email, FTP, Web-traffic), consequently, design a multicast congestion control mechanism that able to coexist and share the bandwidth fairly with TCP is very important.

Scalability

The scalability issue is crucial to all multicast based protocols. A multicast congestion control protocol not only needs to scale to a large number of receivers but also needs to scale in a more heterogeneous environment with different link capacities and delays. Two resulting problems need to be addressed: feedback implosion and rate drop to zero.

This research work tries to address both issues above and implemented into our design protocol in order to achieve reliable multicast communication.

3. Design Protocol

Our protocol design is based on notion of rate adaptation protocol (RAP) which has been proved that this protocol achieves TCP friendliness [1]. RAP is a sources-based rate control and ACK-based loss detection (timeout & gaps in sequence space). RAP source maintain the transmission history: packet sequence number, packet departure time and packet status, in order to estimate the round trip time as well as detect loss. However, RAP is an end-to-end rate based congestion control mechanism that is suited for unicast. Therefore, we add some properties that this protocol can be extended to multicast communication.

Our design protocol uses NACK [5] for loss detection as well as transmission rate adjustment and ACK for transmission rate adjustment only. Sender adjust transmission rate based on Additive Increase Multiplicative Decreases (AIMD) algorithm like TCP does. Round trip time (RTT) between sender and receiver is used as a control parameter and RAP adjust the transmission rate or adjust the Inter Packet Gap (IPG) every RTT. If no congestion is detected, sender periodically increases the transmission rate using the following algorithm:

$$S_i = S_i + \alpha\,(step\ height) \qquad (1)$$

$$S_i = PacketSize/IPG_i \qquad (2)$$

$$IPG_{i+1} = IPG_i * C\,/\,(IPG_i + C) \qquad (3)$$

$$\alpha = S_{i+1} - S_i = PacketSize/C \qquad (4)$$

If congestion is detected (upon packet loss is detected), immediately decrease the transmission rate using the following algorithm:

$$S_{i+1} = \beta * S_i \quad (5)$$
$$IPG_{i+1} = IPG_i / \beta \quad (6)$$
$$B = 0.5$$

Receiver sends the ACK packet only when requested by sender and both ACK and NACK are used to sample round trip time and transmitting in multicast way. The important point in our design protocol is receiver randomly delays transmitting ACK/NACK. When a receiver receives ACK/NACK generated by another receiver for the same packet, it cancels the transmission of its own pending ACK/NACK in order to avoid feedback implosion which is one of crucial issues in multicast communication.

4. Simulation Scheme and Results

We used a plain topology of 5 nodes where each node is linked each other with every links is symmetric.

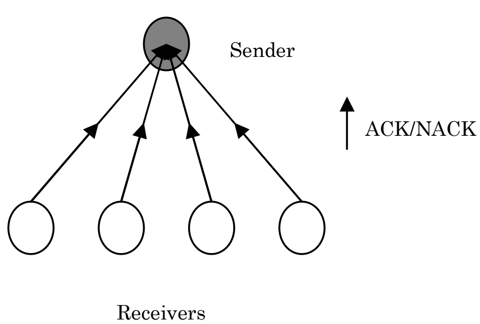
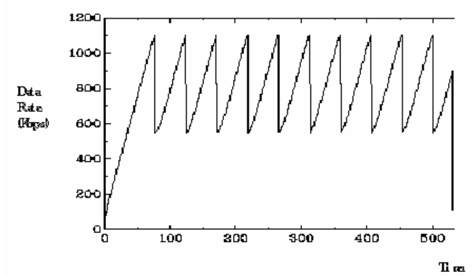
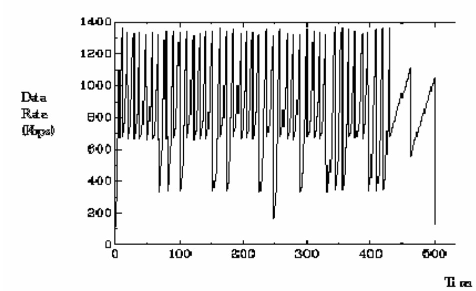


Fig.1. A plain Topology

The bandwidth and feedback delay in every link is set to equal value for examining basic performance of the design protocol.
Figure 2 shows that how the protocol adapts its rate in time. Our numerical result shows that packet flow with small path delay is able to increase transmission rate aggressively due to small feedback delay and the number of packet loss is decrease as an increasing the path delay.



(a). Delay: 250 ms with bandwidth 1 Mbps



(b). Delay: 50 ms delay with bandwidth 1 Mbps

Fig.2. Transmission Rate vs. Time

Our design protocol can adapt with network condition by adjusting the transmission rate and detect a loss based on NACK transmitted by receiver. We add ACK/NACK suppression mechanism at receiver to avoid ACK/NACK implosion at sender.

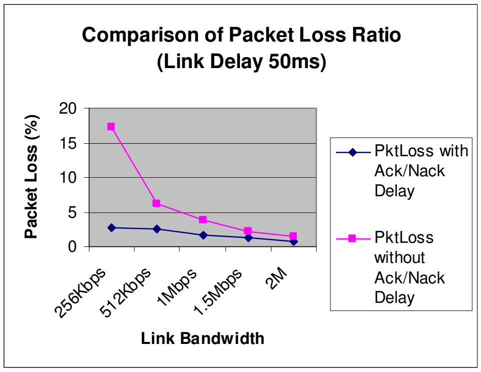


Figure 3. Packet Loss Ratio

As shown in figure 3 that by adding ACK/NACK suppression mechanism at receiver, packet loss can be reduced as well as feedback implosion can be avoided.

5. Conclusion and Future Direction

Currently, Most internet multicast application do not support end-to-end congestion control (usually UDP), hence, this may leads to endanger stability of the network. This paper presents combination of a sender and receiver-based approach for multicast congestion control mechanism. We focus on mechanism that the protocol can adapt with network condition by adjusting the sending rate and detect a loss based on NACK transmitted by receiver. We add ACK/NACK suppression mechanism at receiver to avoid ACK/NACK implosion at sender. The simulation results shows that packet flow with small path delay is able to increase transmission rate aggressively due to small feedback delay and the number of packet loss is decrease as an increasing the path delay. Furthermore, by adding ACK/NACK suppression mechanism at receiver, packet loss can be reduced as well as feedback implosion can be avoided.

To this point, our evaluation is only for constant feedback delay and bandwidth in every links, therefore, we are going to evaluate our design protocol in heterogeneous environment (varying feedback delay and bandwidth) with a huge number of receivers.

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