2001

Dynamics of the "pgmcc" Multicast Congestion Control Protocol

Chin-ying Wang

Sonia Fahmy
Purdue University, fahmy@cs.purdue.edu

Report Number:

01-015

DYNAMICS OF THE "PGMCC" MULTICAST CONGESTION CONTROL PROTOCOL

Chin-ying Wang
Sonia Fahmy

Department of Computer Sciences
Purdue University
West Lafayette, IN 47907

CSD TR #01-015
September 2001

Dynamics of the "pgmcc" Multicast Congestion Control Protocol

Chin-ying Wang and Sonia Fahmy
Department of Computer Sciences
1398 Computer Science Building
Purdue University
West Lafayette, IN 47907-1398
E-mail: {fahmy,chinwang}@cs.purdue.edu

Abstract—Fairness to current Internet traffic, particularly TCP, is one of the important requirements for deploying multicast protocols. In this paper, we investigate the fairness of the multicast congestion control protocol "pgmcc," implemented on top of the PGM multicast protocol. Pgmcc is one of the most promising multicast congestion control proposals, but it has not yet been extensively stress-tested in the literature. Two sets of experiments are conducted in this paper. In first set of experiments, we examine the effect of feedback aggregation on pgmcc. In the second set of experiments, we investigate the performance of pgmcc when competing with bursty TCP and UDP flows in scenario with multiple time-varying bottlenecks and round trip times. Our results indicate that pgmcc is robust, but may need some modifications such as an algorithm for dynamically determining the timeout algorithm and handling switches among receiver representatives better.

Keywords—multicast, congestion control, pgmcc, fairness, feedback aggregation

I. INTRODUCTION

Multicasting allows information exchange among multiple senders and multiple receivers. Popular multicast applications include audio/video conferencing, distance learning, distributed games, server and replicated database synchronization. Pgmcc [1] is a single rate representative-based multicast congestion control scheme that is designed to achieve fairness when competing with TCP flows. Pgmcc sets its window according to a representative called the "acker." The acker is the receiver with the lowest throughput among all receivers within a group. A tight control loop is run between the acker and the sender [1].

Although pgmcc is one of the most promising multicast congestion control proposals, it has not yet been extensively tested under stressful conditions in the literature. In this paper, we examine the pgmcc protocol implemented according to the standard being discussed by the IETF [2]. In our first set of experiments, we demonstrate the feedback aggregation problem caused by the NAK suppression at network elements and show its effect on acker selection. In the second set of experiments, the performance of pgmcc is evaluated when competing with TCP and UDP flows in a more realistic scenario. The fairness of pgmcc to TCP flows is examined with different bottleneck link bandwidths. Simulation results show pgmcc may achieve higher throughput than competing TCP flows during acker switching, especially the first few acker switches. In other cases, pgmcc performance degrades due to the fixed timeout interval used in pgmcc. Experiments are performed using ns 2.1b5, and pgmcc is implemented on top of the PGM [3] multicast transport protocol.

—This research is sponsored in part by *Tektronix*, and the *Schlumberger* Foundation technical merit award.

The remainder of this paper is organized as follows. Section II briefly discusses reliable multicast protocols, specifically PGM and pgmcc congestion control. Section III examines the effect of the feedback aggregation problem. Section IV discusses simulation results on fairness among pgmcc and TCP. Future work is discussed in section V.

II. RELATED WORK

This section discusses reliable multicast protocols, including detailed descriptions of the PGM and pgmcc protocols.

A. Reliable Multicast Protocols

Figure 1 illustrates how reliable multicast protocols work. S represents the sender, and each R represents a receiver in a multicast group. S sends a single copy of the datagram into the network. As the datagram is forwarded (the boxes in Figure 1 represent network routers), it is replicated when needed and forwarded via multiple outgoing links of the router. The original datagram should reach all the receivers in a group. In order to enforce reliability of the protocol, each receiver has to provide some form of feedback to notify the sender whether the data has been received. A receiver may send an ACK if it receives data successfully, or it may send a NAK if a data packet is assumed to be lost.

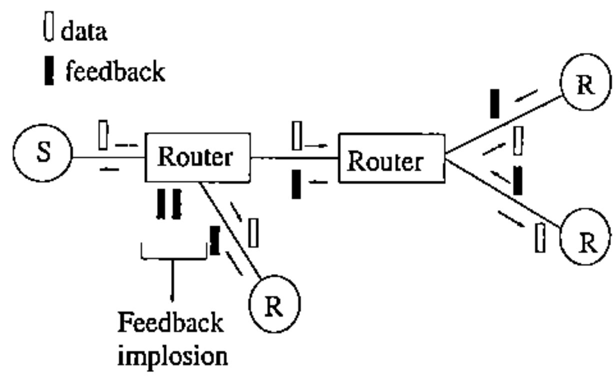


Fig. 1. Reliable Multicast Protocols

Two problems with reliable multicast protocols are shown in Figure 1 and Figure 2. Since each of the receivers will send feedback to the sender, the sender may be overwhelmed by the implosion of ACKs/NAKs when the number of receivers becomes very large. Feedback uses bandwidth unnecessarily, and the sender is burdened with processing all the feedback packets.

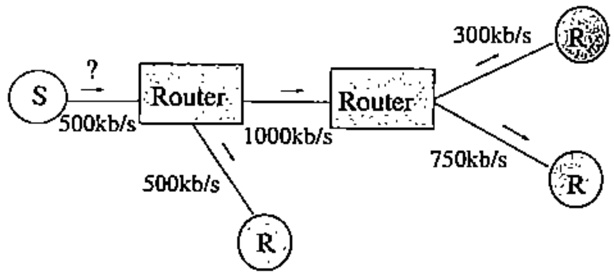


Fig. 2. Determining the multicast sending rate

Determining the appropriate sending rate at the multicast sender is the second problem and is shown in Figure 2. From figure 2, we note that each of the receivers in a multicast group may have a different capacity. The problem of determining the sending rate to achieve the "optimal" bandwidth usage depends on the application reliability semantics.

B. Pragmatic General Multicast (PGM) Protocol

PGM is a single-sender multicast protocol. PGM provides a reliable service by using NAK-based retransmission requests. The use of feedback suppression at PGM network elements improves the scalability of the protocol. The feedback suppression technique allows PGM network elements to only forward the first NAK to arrive at the router serving as the root of a subtree for each missing or corrupted datagram [4]. An example data/feedback packet flow in PGM is shown in Figure 3.

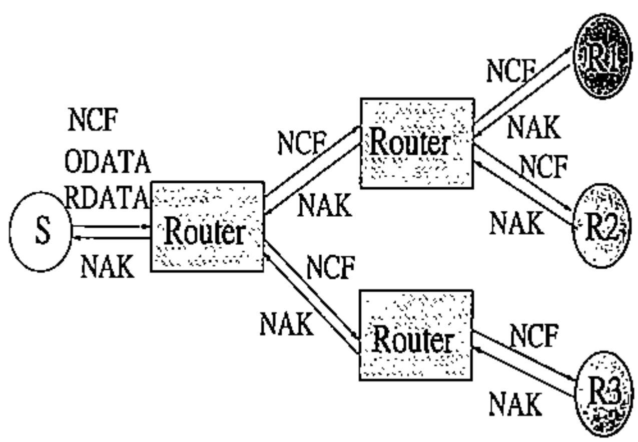


Fig. 3. PGM Multicast Protocol

Each original data packet (odata) sent from the PGM sender is replicated at each network element (router) and forwarded to each of the receivers in a group. If a receiver does not receive odata, sends a NAK upstream. When the upstream router receives a NAK, it sends an NCF packet to the receiver indicating the reception of the NAK, and forwards only one NAK among all NAKs sent from different receivers in the same subtree for each of the lost/corrupted packets. Eventually, the sender will receive the NAK, and repair of the data (rdata) is transmitted only to the receivers who requested the retransmission.

C. Pgmcc

One method of adjusting the sending rate at the PGM sender is to pace the sender according to the "slowest" PGM receiver. This slowest receiver can change at any time since receivers are continuously joining and leaving the multicast group, and bottlenecks vary over time. Pgmcc continuously paces the sending rate according to the receiver with the worst throughput, which serves as the group representative. This receiver is called the "acker."

A tight control loop is run between the acker and the sender. Only the acker sends ACKs to the sender to adjust the sender transmission window and token bucket. Other receivers may send NAKs when they lose packets. Both the loss rate and the round trip time (RTT) are needed to calculate the throughput of each of the receiver. This information is carried in both NAK and ACK packets, and the sender uses the throughput equation as specified in [1] to compute the throughput of each receiver. The acker is switched from one receiver to another if a receiver with a lower throughput is found. An example of acker switching is illustrated in Figure 4.

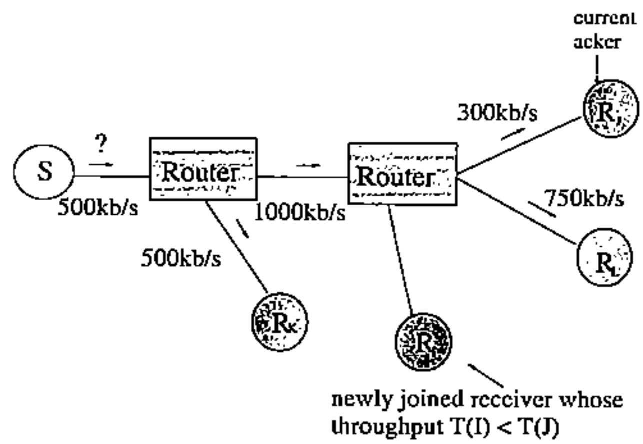


Fig. 4. Acker selection

A window based congestion control scheme similar to that used by TCP is run between the acker and the sender. In the scheme specified in [1], the sender maintains two state variables: a window W, and a token count T. W represents the number of packets in flight, while T is used to regulate the generation of data packets. One token is needed and consumed in order to transmit one data packet. Initially, both W and T are initialized to one. The values of W and T are updated with every ACK, NAK, timeout, and packet transmission [1].

III. Feedback Aggregation

In this section, we illustrate the effect of feedback aggregation on pgmcc. Due to the suppression of PGM NAKs containing RTT and loss rate information needed by the pgmcc sender to select the acker, incorrect acker switches may occur in some cases. An example is shown in Figure 5, where one PGM session runs pgmcc at the sender and each of the six receivers. There are three PGM network elements, and all the links in this topology have the same bandwidth and delay. Among the six PGM receivers, we are interested in receiver 1 (PR1) and receiver 2 (PR2). PR1 is closer to the sender and has a lower loss rate;

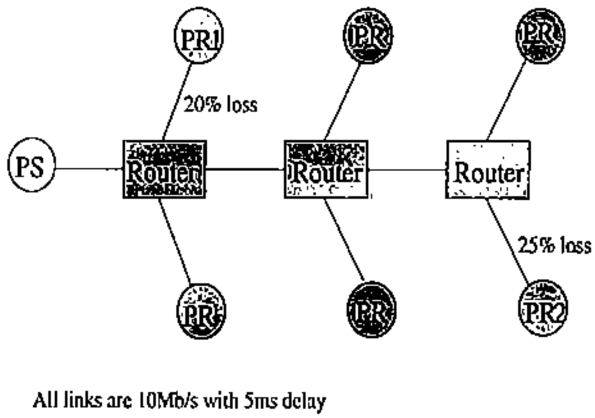


Fig. 5. Scenario to illustrate the effect of feedback aggregation on ACKer selection

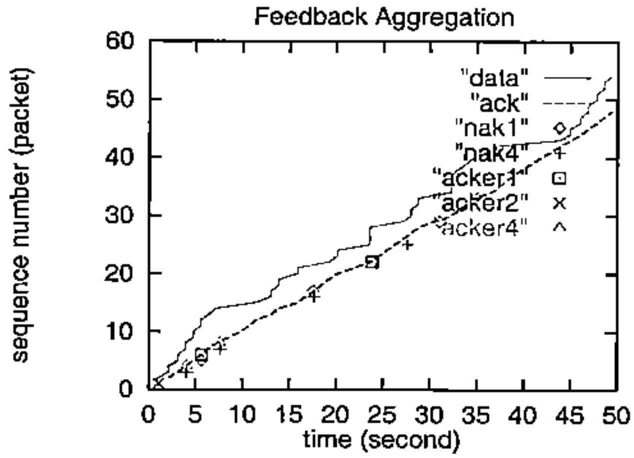


Fig. 7. Simulation results for feedback aggregation

PR2 is further away from the sender and exhibits a higher loss rate. PR2 has an additional 5% loss over PR1.

Suppose the PGM sender begins to send data to all its receivers, and both PR1 and PR2 lose packet number 5. Due to the shorter delay to PR1, the router closest to the sender will receive the NAK from PR1 before receiving one from PR2. Hence, the router forwards the NAK sent from PR1 to the sender, and the NAK sent from PR2 is suppressed. Since pgmcc needs the information of loss rate and RTT carried in NAK packets to perform acker switching, the sender may select PR1 as the acker instead of PR2 at certain instances, even though PR2 clearly has lower throughput.

To verify that this scenario occurs, we simulated a topology similar to Figure 5, which is shown in Figure 6. The only difference is that we simplify the scenario by including only 4 PGM receivers and two network elements so that the simulation is easy to trace and analyze. The simulation is run for 50 seconds.

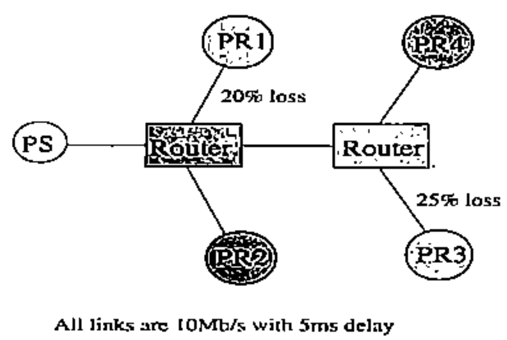


Fig. 6. Simulation topology to examine feedback aggregation

We plot the sequence numbers and acker switches in Figure 7. In the figure, "data" is sent from the sender to the receivers, "ack" is the acknowledgment sent from the acker upon receiving a packet, "nak1" is the NAK sent from PR1, and "nak4" is the NAK sent from PR4. "Acker1" shows that the current acker is PR1 at the specified time and "acker2" and "acker4" denote PR2 and PR4 are the ackers respectively.

From the simulation results, we can see that PR2 is selected as the acker at the beginning of the simulation. About 4 seconds later, the sender receives a NAK from PR4, and it switches the current acker from PR2 to PR4. The acker is switched again to PR1 because the sender receives a NAK from PR1, and the one possibly sent from PR4 is suppressed. Thus, the acker is switched back to PR4 again at the 8th second. The sender switches the acker between PR1 and PR4 a number of times. Unnecessary acker switches occur between PR1 and PR4 although the acker should (on a larger time scale) always be PR4 which has a higher loss rate and higher RTT. Thus the time scale of pgmcc may be too fine, and coarser time scales may increase stability.

IV. Pgmcc Fairness Dynamics

In this section, we simulate pgmcc in more complex configurations. The objective of our experiments is to determine whether pgmcc remains TCP friendly in realistic scenarios more similar to the Internet.

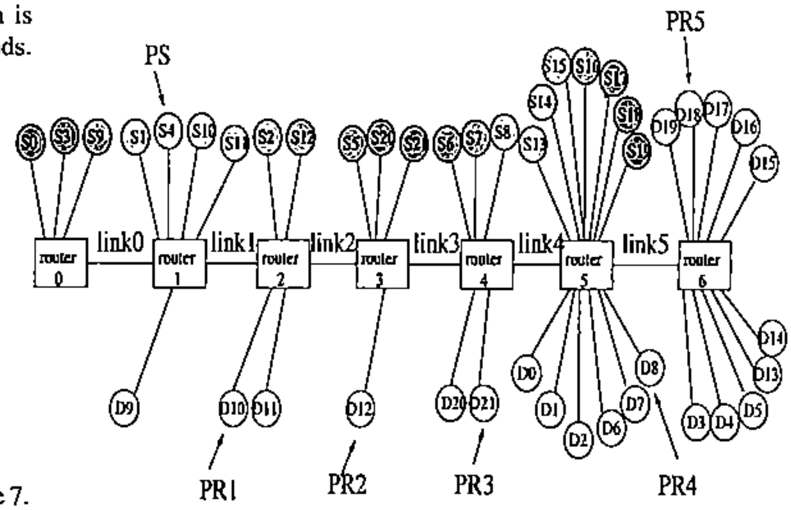


Fig. 8. Simulation topology

Figure 8 shows the simulation topology which includes 22 source nodes (s*) and 22 destination nodes (d*). The link between each node and router has a bandwidth of 150 kbps with 1 ms delay. The link bandwidth and link delay between routers are specified in Table I. 22 TCP flows run between each pair of source and destination nodes. TCP *NewReno* is used because NewReno and SACK are being deployed in the majority of web servers to provide better congestion control compared to other TCP versions [5]. We investigate the performance of the

TCP flow from $S4$ to $D4$, which runs across the same links and nodes as the PGM receiver with the longest RTT. One UDP flow sending Pareto traffic runs across "Link 4" with a 500 ms on/off interval. All the routers use simple drop tail queues with 120 packet slots each. The PGM sender and receivers are located in the nodes shown in Figure 8 and labeled "PS" and "PR*." All the simulations were run for 900 seconds.

In each of the following experiments, we measure the goodput (as defined in [6]) to show bandwidth achieved at the receiver excluding duplicate packets. In all the experiments, the goodput for TCP flows from $S7$ to $D7$, from $S11$ to $D11$, and from $S21$ to $D21$ are almost 2/3 of the link bandwidth because the RTT of each of them is fairly short [7].

A. *Experiment I: Highly Congested Network*

In this experiment, we use link bandwidths and delays as specified in Table I. The throughput (goodput) of the PGM session and the TCP session from $S4$ to $D4$ is shown in Figure 9, and the size of the congestion window for both PGM and TCP is shown in Figure 10.

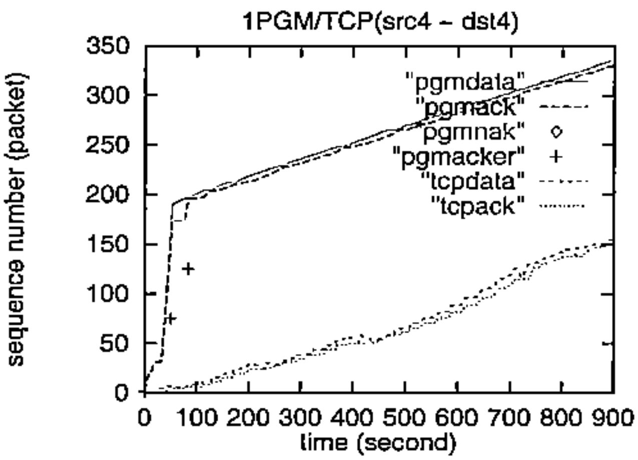


Fig. 9. PGM/TCP throughput in highly congested network

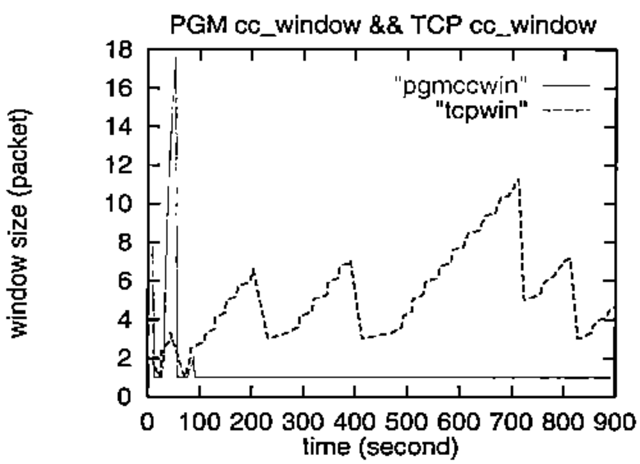


Fig. 10. PGM/TCP sender congestion window in highly congested network

Figure 9 shows that for the first (about) 50 seconds of the simulation, PGM has a much higher throughput than the TCP flow. After 50 seconds, the slopes of the PGM flow and the TCP flow are similar. Both of the flows have low throughput. This reason for this observation becomes obvious from the congestion windows for both PGM and TCP flows shown in Figure 10. The PGM sender window increases only in the first 50 seconds of the simulation. The size of the PGM sender window drops to one several times in the first 100 seconds, and it remains one till the end of the simulation. On the other hand, the congestion window at the TCP sender increases slowly due to the slow start at the beginning, but the TCP sender is able to send more data afterwards compared to the PGM sender due to a larger window throughout the rest of the simulation.

We first trace how the ackers are switched in this experiment. From the pgmcc implementation, the PGM sender always chooses the receiver who is closest to it as the acker at the beginning of the simulation (since it is the first receiver it receives feedback from). Hence, $PR1$ is elected as the acker at the beginning. Later, several packets are dropped at router 4 causing PR4 and PR5 to send NAKs for the same lost packets. The NAKs from PR5 are suppressed and only the NAKs sent from PR4 are forwarded to the sender. Hence, the acker is switched from PR1 to PR4 due to the higher loss rate of PR4 over PR1 perceived by the PGM sender. Finally, more packets are dropped at router 5 causing PR5 to send NAKs for the lost packets, so the last acker switch moves the acker from PR4 to PR5 since PR5 has the longest RTT and the highest loss rate. The receiving rate at each PGM receiver is shown in the first line (labeled "Highly Congested") in Table II.

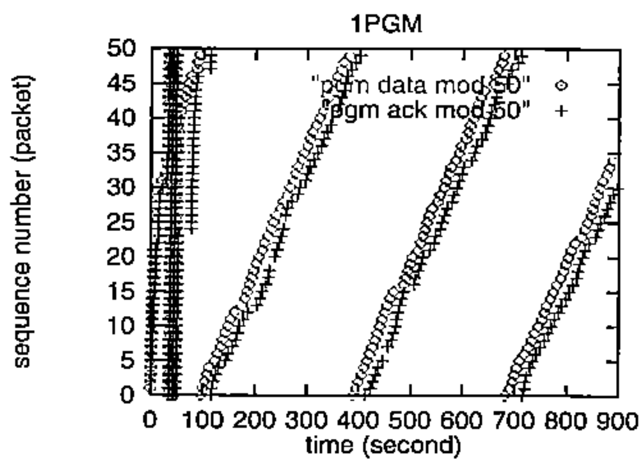


Fig. 11. PGM odata/ack sent/received at the sender in congested network

We also investigate why initial acker switches cause steep increase of the window at the PGM sender. Figure 11 illustrates the time that data packets (represented by the diamond shape) are sent and the acks are received by the PGM sender in this experiment. Each ack (sent from the current acker) is represented by a plus sign in the figure. The packet number (modulo 50 to make the figure more readable) is shown on the y-axis. At the beginning of the simulation, PR1 is selected as the acker because it is closest to the sender. After sending packet number 173, the acker switched from PR1 to PR4 at time 50.222 second (the overlapped diamond and plus sign in Figure 11 at time 76.9 seconds indicates the acker switch). Because the RTT of PR4 is much longer than PR1, it takes longer for PR4 to receive data packets sent from the sender than it does for PR1. Moreover, each data packet is marked with the current acker address

TABLE I

LINK BANDWIDTHS AND DELAYS BETWEEN ROUTERS

Link	*Link 0*	*Link 1*	*Link 2*	*Link 3*	*Link 4*	*Link 5*
Bandwidth (kbps)	50	100	50	150	150	50
Delay (ms)	20	10	5	5	5	10

TABLE II

RECEIVING RATE OF EACH PGM RECEIVER

Experiment	*PR1*	*PR2*	*PR3*	*PR4*	*PR5*
Highly Congested	100%	100%	100%	97%	92.5%
Medium Congestion	100%	100%	99.48%	100%	97.14%
Uncongested	100%	99.9%	98.39%	99.9%	98.03%

in pgmcc. Hence, even though an acker switch occurs early, the previous acker (PR1) continues sending ACKs to the sender until reception of packet number 173. As a result, the new acker (PR4) only sends ACKs after the reception of packet number 174. In this experiment, PR4 began sending ACKs after 76.9292 seconds even though the acker switch occurred at time 50.222 seconds.

There are two consequences of this behavior. First, when an acker switch occurs, it means there is a receiver with a lower throughput than the current acker. However, as discussed above, the previous acker keeps on sending ACKs till the packet number is equal to the trail of the sender window at the time of the acker switch. For each ACK received at the sender side, the sender increases the token count T by one and increases the window accordingly. Thus, more data packets are sent by the sender. This leads to the second consequence which is that the network becomes even more congested. We observed the same behavior when the acker switched from PR4 to PR5.

The delay of sending ACKs from the new acker observed in this experiment is one of the causes of the sudden drop of the window size to one. Because pgmcc uses a fixed timeout interval to detect congestion, if the sender does not receive an ACK from the acker within the timeout time specified, it drops the window size W to one and decreases the token count T to $\frac{-1}{2W}$ [1]. Revisiting Figure 11, we see that the distance between the last overlapped cross and diamond at time 76.9 and the first non-overlapping cross indicates the time that the sender waits for the ACK for packet number 174 from the new acker (PR4). If the distance is longer than the timeout interval, which is true in this experiment, one or more timeouts occur, degrading PGM performance.

Another reason for the sudden drop of the window size to one is that the RTT of the current acker itself is sometimes simply *longer than the timeout interval.* In this case, the PGM sender will never be able to receive an ACK within the timeout time and will keep timing out, as shown in Figure 10. This problem can be remedied by implementing a TCP-like retransmission timeout determination algorithm.

In terms of the goodput, the goodput for each of the PGM receivers ranges from 3.95 to 4.26 kbps and the goodput for TCP receiver is 1.39 kbps. It is not surprising that goodput for both PGM and TCP flows is quite low because the links are congested and shared among many TCP and UDP flows.

B. Experiment II: Medium Congestion

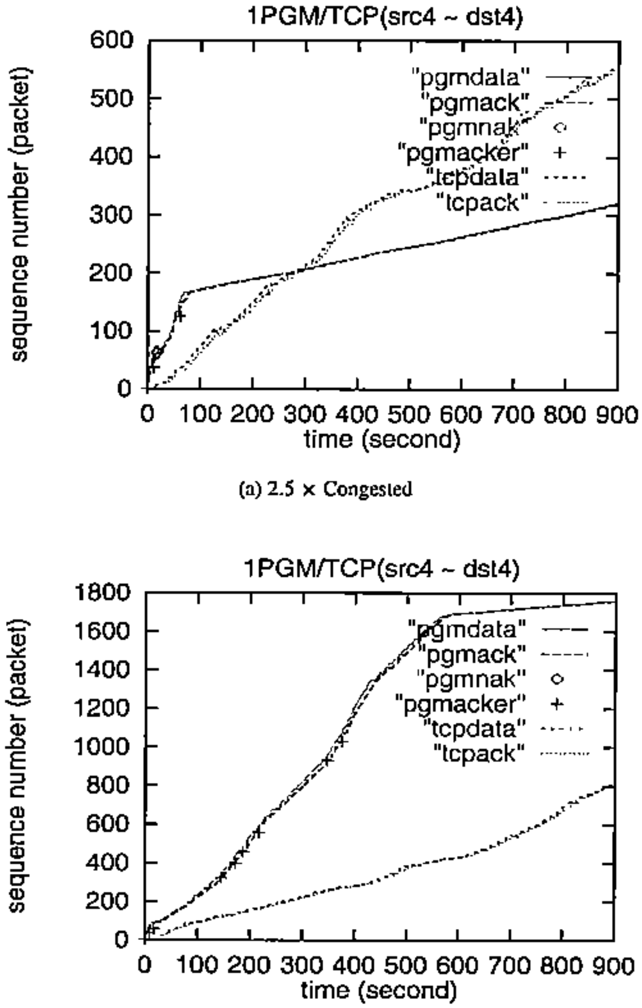
In this section, we keep all simulation parameters unchanged except that we increase the bandwidth of the links between routers. Many experiments were run with bottleneck link bandwidths ranging between 2.5 and 3.5 times the original bandwidths shown in Table I. The results are similar, so we only discuss the results using bandwidth of 2.5 times, and 3.5 times the original bandwidth.

The throughput of both PGM and TCP flows and their windows sizes are shown in Figures 12 and 13 respectively. The receiving rate of each PGM receiver is shown in the second line of Table II.

We observe similar behavior to Figure 9 in Figure 12(a). Fewer timeouts occur (shown in Figure 13(b) in this experiment compared to the highly congested network in the previous subsection. This is because we have increased the link bandwidths, so the time it takes for transmission of ACKs from the acker to the sender is shorter than that in the previous setting. However, the timeouts are still frequent, causing the window to drop to 1 because the RTT of PR5 is greater than the PGM sender timeout interval. The throughput of the TCP flow, on the other hand, is better than the one in experiment I and has a higher slope. In terms of the goodput, the goodput of PGM receivers ranges from 4.77 to 4.91 kbps, and goodput of the TCP receiver is 4.97 kbps.

Figure 12(b) may appear different at first, but it is similar to Figure 12(a) if we had run the simulation longer. The goodput of each of the PGM receivers ranges between 22 and 22.4 kbps, and that of the TCP receiver is 7.24 kbps. The reason for the higher goodput for PGM receivers, in addition to the increase of the bandwidth, is that the acker was switched from PR3 to PR5 several times. As discussed above, acker switches take time and the sender window is increased meanwhile. Further, the two different branches have different throughputs. This effect is clearly shown in Figure 13(b).

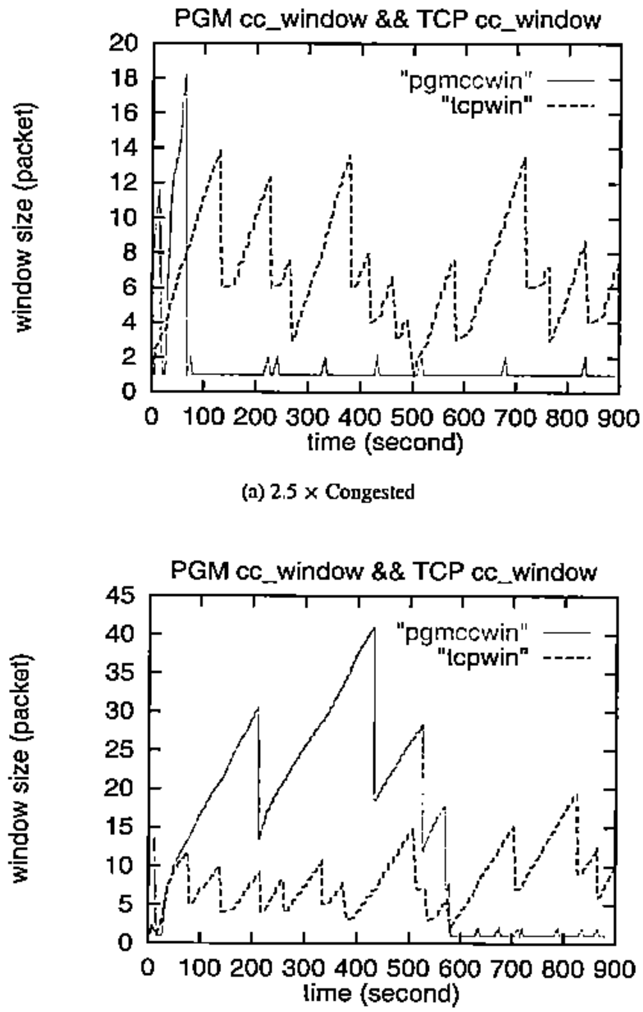
By increasing the bottleneck link bandwidth, the throughput of both the PGM and TCP flows increases. From this set of experiments, we conclude that the PGM flow outperforms the TCP flow during initial acker switching, but the TCP flow has a higher throughput if the timeout interval at the sender does not



(a) 2.5 × Congested

(b) 3.5 × Congested

Fig. 12. PGM/TCP throughput with medium congestion



(a) 2.5 × Congested

(b) 3.5 × Congested

Fig. 13. PGM/TCP sender window size with medium congestion

adapt to the increase of the acker RTT.

C. Experiment III: Uncongested Network

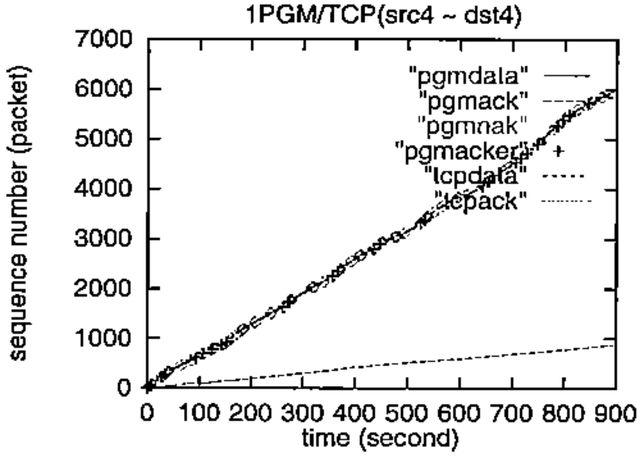
In this section, we retain all the parameter values of previous experiments but we increase the bandwidths of the links between routers. Many experiments were run with various bandwidths, but, since the results are similar, and we only show the results using bandwidths 10 times, and 80 times the original bandwidths in Table I. The throughput of both PGM and TCP flows and their windows sizes are shown in Figure 14 and Figure 15.

From the results in this set of experiments, we find that the acker switches back and forth between PR3 and PR5 due to the closeness of the throughputs of PR3 and PR5 (the receiving rate for each of the PGM receivers is shown in the third line (labeled "Uncongested") in Table II). Feedback suppression does not cause problems here because PR3 and PR5 lose different packets. As expected, the PGM sender window continues to increase some time after acker switches. In terms of goodput, the goodput for each of the PGM receivers ranges from 74.15 to 76.56 kbps, and for the TCP receivers it is around 7.8 kbps in both Figure 14(a) and (b).

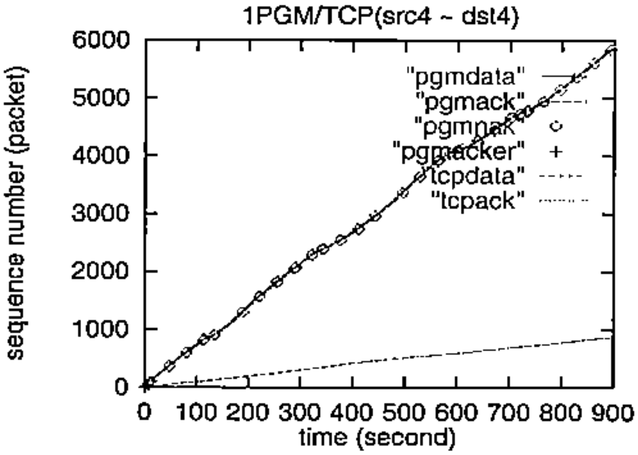
The reason why PGM outperforms TCP appears to be the selection of PR3 as the acker throughput most of the simulation. Both the TCP receiver and PGM receiver PR5 are connected to router 6. PR5 is the acker for only 173 seconds while PR3 is the acker for 736 seconds of the simulation. PR3 has a short RTT and reasonable loss rate. Therefore, even though increasing the link bandwidth will increases the throughput of both PGM and TCP flows, the PGM flow outperforms the TCP flow in this case.

V. Conclusions and Future Work

In this paper, we have investigated the fairness and dynamics of the pgmcc single-rate multicast congestion control protocol. Our simulation results show that a pgmcc flow initially sends more than a competing TCP flow due to the rapid opening of the PGM sender window between initial acker switches. If the acker selection process stabilizes and a PGM receiver with a very long RTT is selected to be the acker, timeouts severely degrade the performance of the PGM flow. A TCP-like retransmission timeout computation mechanism can remedy this problem. With the

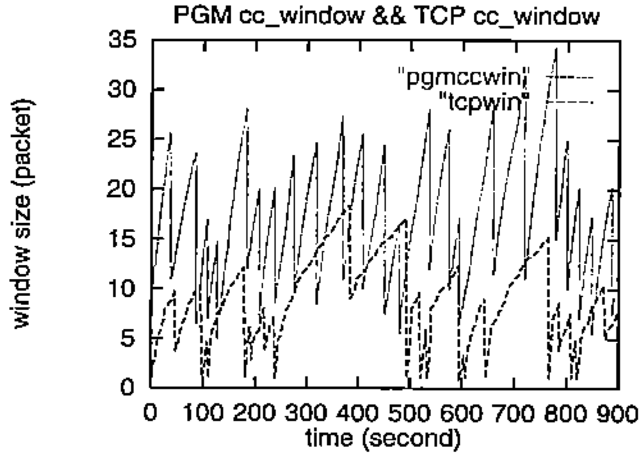


(a) 10 × Congested

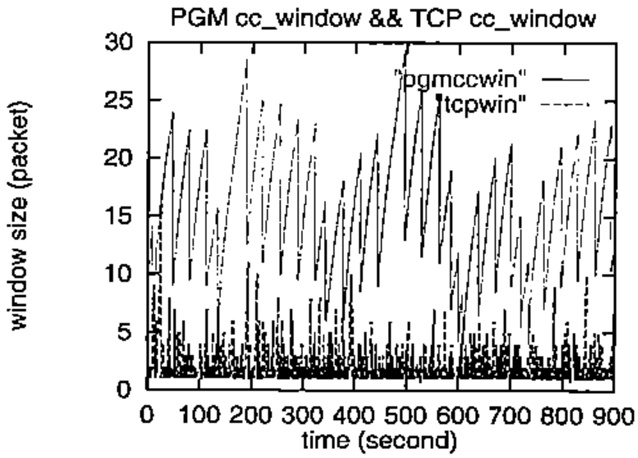


(b) 80 × Congested

Fig. 14. PGM/TCP throughput in an uncongested network



(a) 10 × Congested



(b) 80 × Congested

Fig. 15. PGM/TCP sender window size in an uncongested network

use of a timeout interval selected according to the acker RTT, the PGM sender can distinguish between situations of real congestion and late ACKs received from an acker with a long RTT. In an uncongested network, the PGM flow may outperform a competing TCP flow if frequent acker switches occur between ackers of different throughputs.

We plan to examine various application reliability semantics to see how the pgmcc scheme fits in the unreliable (or not fully reliable) multicast protocol context. In [1], the author states that the pgmcc can be safely deployed with both reliable and unreliable multicast applications. The PGM multicast protocol provides reliability in the transport layer as specified in [3] and illustrated in Figure 3. On examining the ns pgmcc implementation, we find that if a PGM receiver lost a data packet, it only sends a NAK back to the sender once. If the NAK get lost or corrupted before it gets to the sender or if the NCF sent to acknowledge the NAK reception is lost or corrupted before it gets to the receiver, the receiver which originally sent the NAK will wait for a retransmission timeout. Then, the receiver reschedules the retransmission timeout up to ten times waiting for the repair, instead of resending the NAK. If, after rescheduling the retransmission timeout for ten times, the repair is not received, the receiver treats the packet as unrecoverable. This essentially means that this flavor of PGM is not fully reliable. It is not completely unreliable either because NAKs are sent, and when a receiver receives the repair from the sender, it does not check whether the repair is needed or not (e.g., if the repair must be received within a certain amount of time and it is useless otherwise). We plan to experiment with various reliability semantics, and examine their effect on the pgmcc congestion control algorithm, especially on acker selection with insufficient NAKs.

ACKNOWLEDGMENTS

The authors would like to thank Gianluca Iannaccone and Luigi Rizzo for providing us with their pgmcc implementation and their help with configuring it.

REFERENCES

[1] L Rizzo, "pgmcc: a tcp-friendly single-rate multicast congestion control scheme," in *Proceedings of the ACM SIGCOMM*, August 2000, http://www.iet.unipi.it/ luigi.

[2] L. Rizzo, L. Vicisano, M. Handley, and G. Iannaccone, "PGMCC single rate multicast congestion control: Protocol specification," Internet Draft, Internet Engineering Task Force, Feb. 2001, Work in progress.
[3] D. Farinacci, A. Lin, T. Speakman, and A. Tweedly, "PGM reliable transport protocol specification," Internet draft, March 2000, Work in progress.
[4] K. Miller, "Multicast networking and applications," Addison Wesley Longman, Inc., 1999.
[5] K. Fall and S. Floyd, "Simulation-based comparisons of Tahoe, Reno, and SACK TCP," in *ACM Computer Communication Review*, July 1996, vol. 26, pp. 5–21, ftp://ftp.ee.lbl.gov/papers/sacks.ps.Z.
[6] S. Floyd and K. Fall, "Promoting the use of end-to-end congestion control in the internet," *IEEE/ACM Transactions on Networking*, August 1999.
[7] J. Padhye, V. Firoiu, D. Towsley, and J. Kurose, "Moldeling TCP throughput: A simple model and its empirical validation," in *Proceedings of the ACM SIGCOMM*, September 1998, vol. 28, pp. 303–314, http://gaia.cs.umass.edu/.