Technical Report: *LE-SBCC: Loss-Event oriented Source-based Multicast Congestion Control*

Puneet Thapliyal, Sidhartha, Jiang Li, Shivkumar Kalyanaraman[1]
Rensselaer Polytechnic Institute, Troy, NY 12180
Contact Email: shivkuma@ecse.rpi.edu

Abstract

We propose a *loss-event oriented purely source-based single-rate* multicast congestion control scheme (LE-SBCC) for reliable or unreliable multicast transport protocols. The scheme is easy to deploy and addresses the *drop-to-zero* and *TCP friendliness* problems [15, 18]. It consists of a simple cascaded set of filters and a rate-based AIMD module. Loss indications (LIs) from receivers are converted into loss events (LEs) [6]; the maximum LEs from any path are chosen, and at most one rate-reduction per round trip time (RTT) is effected. The scheme performs well in the common case of *partial LI aggregation*, though the worst case in the rare instance of *complete LI aggregation* is aggressive towards TCP. A range of results (numerical, simulation and experimental) is presented. The scheme has been implemented using PGM as the underlying RMT protocol over a Redhat 7.0 Linux testbed.

1 Introduction

This paper presents a single-rate loss-event oriented *purely source-based* congestion control scheme (LE-SBCC) for reliable or unreliable multicast transport protocols. The scheme can function without any *new* support from receivers, network elements or packet-header support and can leverage the underlying loss indications (LI) provided by protocols like PGM or RMTP [16, 19]. LE-SBCC is a single-rate scheme like PGMCC [15], LPRF [2, 3], Equation-based TFMCC [9, 18], MTCP [14] and Golestani [7]. In contrast to LE-SBCC, these schemes require *new* support at the source, receivers and transport packet-format level. Examples of multi-rate schemes (which are not source-based) include RLM, RLC, FLID-DL etc [11, 17, 5, 10].

Our scheme consists of a *purely source-based* cascaded set of filters and RTT estimation modules feeding into a rate-based additive increase/multiplicative decrease (AIMD) module (Figure 1). The filters are designed to address the following key problems:

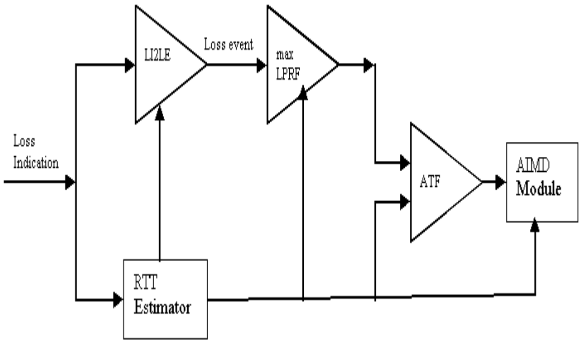


Figure 1: Cascaded Filter Model of LE-SBCC

Drop-to-Zero: is the problem of reacting to *more loss indications (LIs) than necessary* leading to a beat-down of the multicast flow's rate[18, 3]. This occurs because the multicast flow receives LIs from multiple paths and may not filter LIs sufficiently. In an earlier work [12], we showed that *without* flow multiplexing on paths, the problem of independent multi-path loss can be addressed through a simple adaptive time filter and a TCP-like RTT/RTO estimator. A single flow's congestion control actions directly impacts the loss rate and loss patterns in this case. However, this solution is insufficient if paths display a high degree of flow multiplexing, because then any *single* flow's congestion actions do not appreciably impact the path loss rates.

TCP-unfriendliness: is the problem of reacting to *less LIs than a hypothetical TCP flow would on the worst loss path* [4, 18, 9]. TCP-friendliness is a subject of current debate[6, 15, 9]. In particular, Padhye et al's TCP throughput formula [13] requires the definition and estimation of "loss rate" and "RTT" of the paths. The unicast scheme, TFRC introduced the concept of loss events (LEs) and uses the LE rate instead of the LI rate as the "loss rate." *An LE is a binary number defined every RTT per receiver which is 1 when one or more LIs are generated in that RTT by the receiver, and is 0 otherwise.* Like TFRC, this paper will also first derive LEs from LIs before responding to congestion.

On the subject of RTT estimation, PGMCC and TFMCC [15, 9] have receivers estimate RTT and drive the control based upon paths having the maximum value of ($RTT\sqrt{q}$), where q is the *per-packet LE probability.* Our proposed scheme LE-SBCC computes $\max RTT$ and $\max q$ concurently and uses both these estimates to drive the AIMD module. This decoupled use of $\max RTT$ and $\max q$ represents a sub-optimal approach in general[15, 9, 18].

[1]We acknowledge the help of Jye-Young Song to implement this scheme

2 Cascaded Filter Model

Figure 1 illustrates the outline of the building blocks of the scheme implemented at the multicast source. The cascade of filters convert a stream of loss indications (LIs) from receivers into per-receiver LEs, filter the LEs further, and finally output a stream of rate-reduction indications (RRs) to the rate-based AIMD module (at most one RR per RTT). The AIMD module performs multiplicative rate decrease upon receipt of an RR and performs rate increase each time an RR is not received within an increase interval. Observe that this closely models the behavior of AIMD control on a *single path* where RRs are triggered by LEs [6]. The modules work as follows:

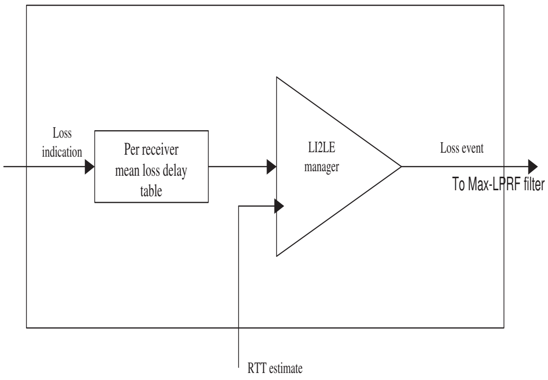


Figure 2: Loss Indication to Loss Event Filter (LI2LE)

LI2LE filter: This filter converts per-receiver loss indications (LIs) into per-receiver loss events (LEs). Recall that an LE is a per-receiver binary number which is 1 when one or more LIs are generated per RTT per receiver, and 0 otherwise. It stores a *per-receiver timestamp* ($T_{LastPassed}$) which records the time when the last LI was converted and passed as an LE. If a new LI arrives from the receiver *after* a period $SRTT + 2\sigma$ of[2] $T_{LastPassed}$, it is converted into an LE and passed, and the timestamp $T_{LastPassed}$ is updated to the current time. Else the LIs are filtered. This filter is critical because the rate-based AIMD module is not self-clocked (no acks like TCP, PGMCC [15] or Golestani [7]); which leads to large burst losses with drop-tail queues[3]

Max-LPRF: This filter is a extension of Bhattacharya et al's Linear Proportional Response Filter[2]. Assuming X_i is the count of LEs from receiver i, this

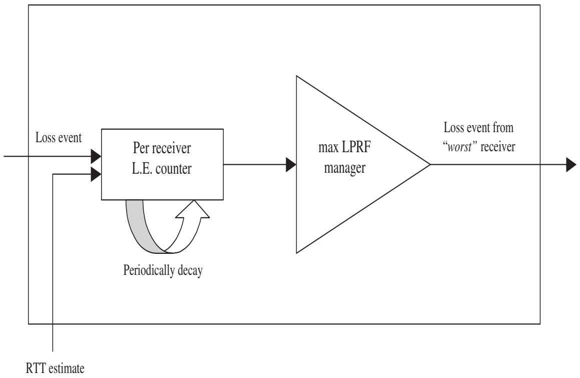


Figure 3: Max-LPRF: max-Linear Proportional Rate Filter

probabilistic filter takes as input all the LEs from receivers ($\Sigma_i X_i$) and on the average passes the maximum number of LEs from any one receiver (i.e. $\max_i X_i$). In particular, it *passes each LE with a probability* $\frac{\max_i X_i}{\Sigma_i X_i}$. The LE counts per receiver (X_i) are decayed periodically by 10% every 100 $SRTTs$. The O(N) state requirements of both LI2LE and Max-LPRF are not a big issue because single-rate schemes are typically targeted for a small-medium scale (< 10000 receivers). *Max-LPRF tracks the worst path better than LPRF and is the crucial building block for drop-to-zero avoidance.* It operates on per-receiver LE counts since they differ dramatically from LI counts in drop-tail networks with no self-clocking.

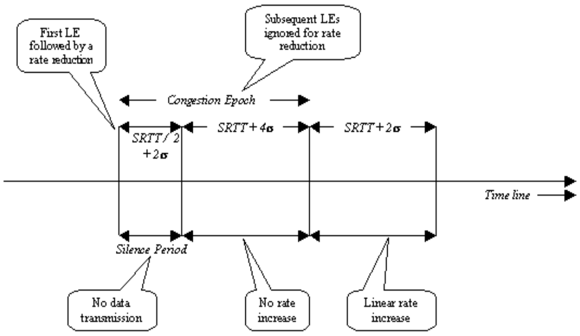


Figure 4: Adaptive Time Filter (ATF)

Adaptive Time Filter (ATF): This filter shown in Figure 4 is a modification of the time-filter introduced in our earlier work [12]. This filter simply drops excess LEs passed by Max-LPRF in any RTT to enforce *at most one rate reduction (RR) per* $SRTT + 4\sigma$. In addition, the filter also imposes an optional *"silence period"* of $SRTT/2 + 2\sigma$ when no packets are sent. As suggested in [12], the goal is to

[2] $SRTT =$ average of RTT samples similar to the variable used in TCP

[3] During overload the queue increases quadratically with time in rate-based AIMD.

reduce the probability of losing any control traffic or retransmissions during this phase.

All filters and the AIMD module need RTT estimates which is fed by the RTT estimation module developed in our earlier work [12]. It works similar to the TCP timeout procedure i.e. it calculates a smoothed RTT ($SRTT$) and a mean deviation which approximates the standard deviation σ. However the set of samples is pruned to exclude a large fraction of samples which are smaller than $SRTT/2$ (i.e. smaller by an order of magnitude) to bias the average RTT higher. The rate increase uses intervals of length $SRTT + 2\sigma$ (see [12]) while other functions differ in their use of σ as described earlier.

In summary, the scheme expects: (a) at least one LI per receiver per RTT if packets are lost in that RTT and (b) timely generation of feedback by receivers/network elements to allow reliable RTT sampling[4]. The scheme is sensitive to arbitrary delays introduced by receivers/network-elements in generating an LI corresponding to a lost packet. Further *LI aggregation* like in PGM [16] itself is a form of filtering which suppresses receiver IDs, timing information and reduces LI or LE counts, thus conflicting with (a) and (b) above[5]. We examine the performance effects in the following sections.

3 Performance Evaluation

We have evaluated the performance of LE-SBCC using simple modeling, numerical simulation (upto hundreds of receivers), *ns-2* simulation, linux-based implementation/experimentation (high multiplexing degrees and upto 10s of receivers).

3.1 Drop-to-Zero Avoidance

We have analyzed drop-to-zero (DTZ) problem using *ns-2*, numerical simulations and Markov Chain based theoretical analyses. We show that even in heavily multiplexed scenarios, we acheive DTZ avoidance.

3.1.1 NS-2 simulations illustrating DTZ avoidance

Consider a multicast flow to 4 receivers as shown in Figure 5. The links from "Router 2" to each of the receivers are bottlenecks at 5 Mbps each. A set of 4

[4]Within these constraints the scheme may be applied to unreliable multicast transport protocols

[5]LIs carry RTT information which affects the RTT estimation, and LEs carry receiver IDs which affects the max-LPRF filter.

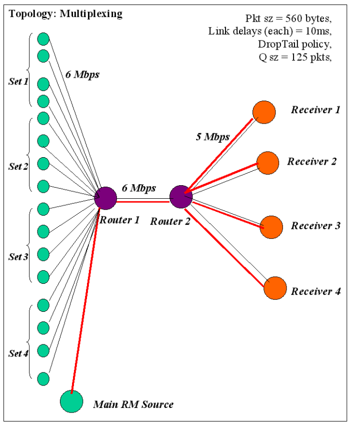


Figure 5: Topology to Test Drop-to-Zero Avoidance and TCP Friendliness

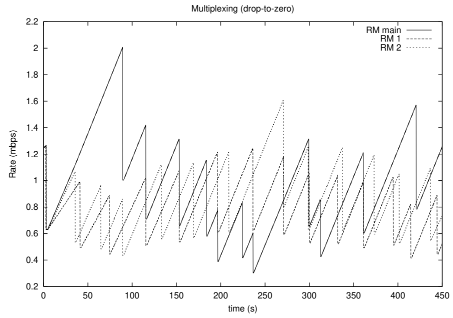


Figure 6: Drop-to-zero Avoidance Results

background unicast flows implementing the LE-SBCC algorithm compete on each of the 4 bottlenecks links. In Figure 5, the *unicast* flows in "Set 1" through "Set 4" compete on the bottleneck links to "Router 2-Receiver 1" through "Router 2-Receiver 4" respectively. The fair rate for all flows is therefore 1 Mbps. The buffer size (125 pkts) is roughly twice the bandwidth times fixed delays, and packet size is 560 bytes. Figure 6 shows that the multicast flow (solid line) competes fairly with a selected subset of unicast background flows (dotted lines) and the average rate is close to 1 Mbps, which is a non-trivial illustration of drop-to-zero avoidance.

The main RM flow transmitted a total of 167762 pkts in 450s and received 949 LIs (i.e. a low loss rate). After LI2LE filtering, there were only 32 LEs, a dramatic reduction! As discussed earlier, the bursty losses are caused by the lack of AIMD self-clocking in combination with drop-tail buffers. The Max-LPRF and ATF further filter LEs to result in a total of 12 rate-reductions which is the same as a unicast flow as shown in Figure 6. This illustrates the important role of the LI2LE and Max-LPRF filters in the DTZ avoidance process.

3.1.2 DTZ avoidance in medium to large scale trees

Scalability of DTZ avoidance in non-aggregated scenarios is numerically analyzed. We use a fanout topology where all the receivers and the single source are connected to the same router. The number of receivers in the multicast tree is increased from 1 to 10001 in a gradual manner. Number of LE's passed should be roughly equal to the LE's produced by the Max loss receiver. We assume one receiver has 5% loss, all others have 2% loss and study the effect of increasing the number of 2% receivers. The number of LIs reaching the source is plotted against the number of receivers in Figure 7. Under no aggregation, the number of LE's passed by source remains fairly constant(240-250). This indicates the scalability of our scheme in unaggregated scenarios.

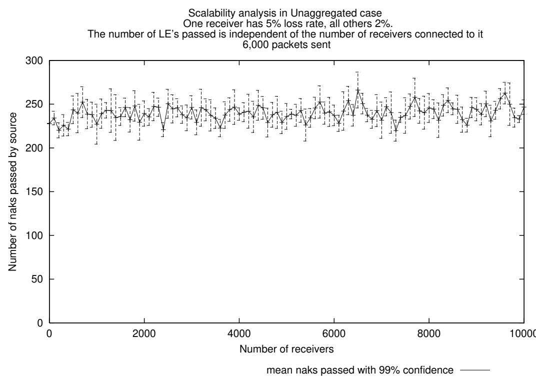


Figure 7: Numerical analysis: Scalability

3.1.3 Theoretical analysis of LE vs LI probability

We use LEs instead of LIs for rate reduction decisions. We do a theoretical analyses to show that this choice has less probability of DTZ. We have proposed a Markov Chain model for this process. We assume constant RTTs for this analysis.

Our model assumes that at any point in time, the source may be in any discrete state from 1 to N. The state x specifies the number of packets sent during that RTT, $x = Rate(x) \times RTT$. We start in state 1, and may go upto the maximum allowable rate, which translates to N. In general, whenever a loss event is encountered, we do a transition to state $\lfloor x/2 \rfloor$. In case of no loss we move to state $x + 1$. (See Figure 8) Let p be the LI probability. Let $a = 1 - p$. The probability of having at least one loss in state x (i.e probability of having an LE) is $1 - a^x$. For simplicity, assume $N = 2 \times m$. Let $p(x)$, be the probability of being in state x.We have five cases to consider :

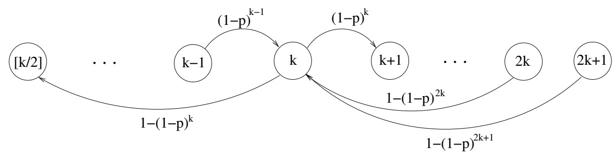


Figure 8: State transition of packets sent per RTT

1. We are in state 1. We could increase the rate and move to 2. We could also get losses at states 2 or 3, and move to 1. We could also get a loss, and stay in 1. In steady state,
$$p(1) \times a = p(2) \times (1 - a^2) + p(3) \times (1 - a^3)$$
2. We are in any state x between 2 to $m - 1$. This is the general case mentioned above. In steady state,
$$p(k) = p(k-1) \times a^{k-1} + p(2k) \times (1 - a^{2k}) + p(2k+1)(1 - a^{2k+1})$$
3. We are in state m. We could increase the rate and move to $m + 1$. We could also get loss at state N only, and move to m. In steady state,
$$p(m) = p(m-1) \times (a^{m-1}) + p(N) \times (1 - a^N)$$
4. We are in any state x between $m+1$ to $N-1$. The only difference between this state and 3 is that we cannot enter this state by a rate reduction, as the $N = \max(rate \times RTT)$. In steady state,
$$p(k) = p(k-1) \times a^{k-1}$$
5. We are in state N. We cannot increase the rate further. The only way to leave this stae is by cutting its rate, and moving to state m. In steady state,
$$p(N) \times (1 - a^N) = p(N-1) \times a^{N-1}$$

We did a Matlab analysis to solve the above set of equations for different large N's, final plots of which are similar. One plot of proportion between LI probability p and LE probability q is shown below (Figure 9).

Note that we did not consider bursty loss in the situation above. But in the real world, packet losses are mostly bursty because of buffer queues. To approximate the bursty behavior we modified our model. We assume that if a packet is lost during any instance within the RTT, all remaining packets transmitted during that RTT are also lost. As a result, ratio of LI(p) and LE(q) probabilities is changed (Figure 10). In all cases we see that q is lower than p. This suggests fewer rate cuts and thus less likelihood of DTZ.

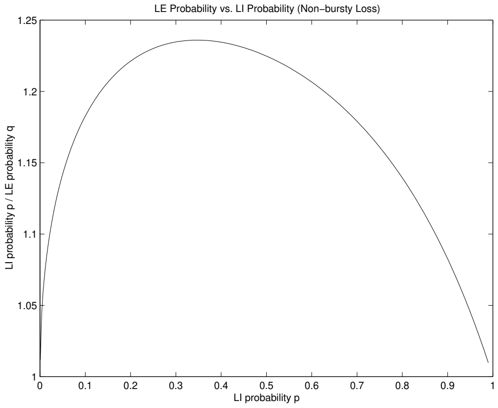


Figure 9: Ratio of LI(p) and LE(q) probabilities (Non-bursty loss)

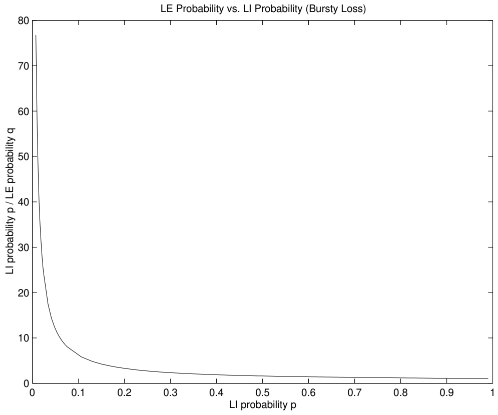


Figure 10: Ratio of LI(p) and LE(q) probabilities (Bursty loss)

Usually the packet loss rates observed in real networks are lower than 1%. To see the relation between LI and LE more clearly, we have another graph for low LI probability (Figure 11), which fits well with our simulation results. We observe high p/q ratios (10-90) which indicates our choice of LEs as being more effective for DTZ avoidance.

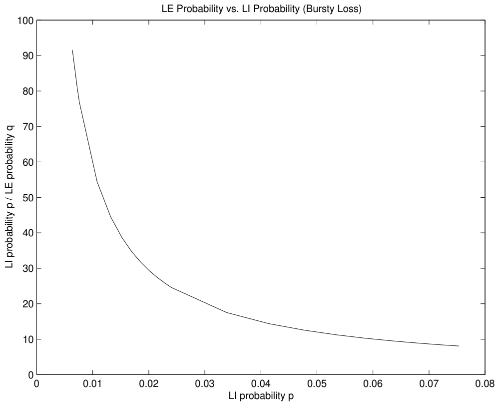


Figure 11: Ratio of LI(p) and LE(q) probabilities (Low loss probabilities)

3.2 TCP-friendliness

In the topology of Figure 5, consider two cases: case (a) when *one* of the flows of "Set 4" is replaced by a TCP Reno flow while the rest are LE-SBCC flows; and case (b) when *all* the flows in "Set 1" through "Set 4") are replaced by TCP flows. This topology is used to illustrate TCP friendliness.

NS simulation results: The simulation results for these two cases is shown in Figures 12(a) and 12(b) [6]. The dips by TCP close to zero represent timeouts. Again, the main RM flow (solid line in both graphs) shares the bottleneck(s) fairly with the competing TCP flow(s). The horizontal dotted line is the expected TCP throughput plotted post-simulation using the simplified TCP equation ($1.22MSS/RTT\sqrt{q}$).

Experimentation results: The scheme is implemented in a Linux 7.0 testbed. The topology used is identical to the one used for the NS simulations as described above. Again the RM flows are fair to the competing TCP flow (Figure 12(c)). More details on this are presented in the section on implementation issues.

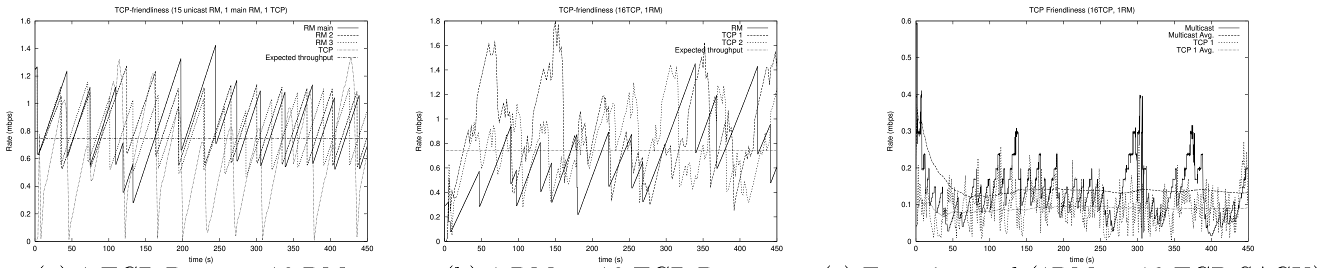
3.2.1 Expected RM throughput using LE probabilities – a measure of TCP friendliness

We treat our multicast problem as unicast one with the "worst" receiver being the only receiver with loss probability p and RTT T. Then using the simple throughput formula for unicast proposed by Mathis et al ($1.22MSS/RTT\sqrt{q}$), we do a post processing of the simulation data to verify the correctness of the throughput of our scheme. Clearely from the Figures 12(a) and 12(b), the AIMD rate graphs oscillate with the mean equal to the theoretical mean. The fact that the average throughput of the RM flows is equal to the unicast flows is a good measure of the TCP friendliness of our scheme. This further demonstrates the validty of choosing LE probabilities rather than LI probabilities in our scheme.

3.2.2 DropTail vs. RED queues

We evaluated the TCP-friendliness performance of our scheme with the two queue management policies - *Tail Drop* and *Random Early Detection (RED)*. The toplogy

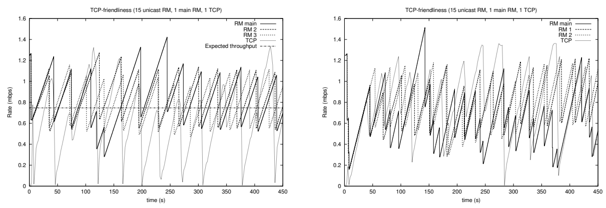
[6]To avoid cluttering, the TCP thruput is sampled every 2s



(a) 1 TCP Reno vs 16 RM (b) 1 RM vs 16 TCP Reno (c) Experimental (1RM vs 16 TCP SACK)

Figure 12: TCP Friendliness (Simulation and Experiment)

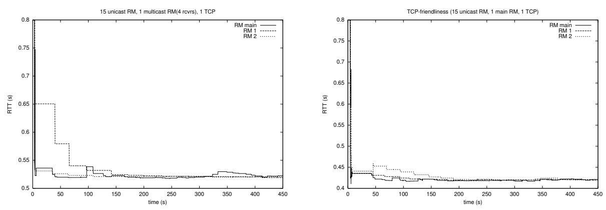
is the same as above and the other relevant parameters are as follows: TCP version: *Reno*, Q sz = *125 pkts*, Pkt sz = *560 bytes*.



(a) DropTail (b) RED

Figure 13: Droptail Rate vs. RED Rate

As we can see from Figure 13, the throughput is marginally lower with RED than with DropTail and so is the measured RTT (Figure 14). Also observe that the RTT converges much faster with RED than DropTail. Still, the scheme remains absolutely fair towards the competing TCP flow in both the cases and we do not see any DTZ with the RM flows.

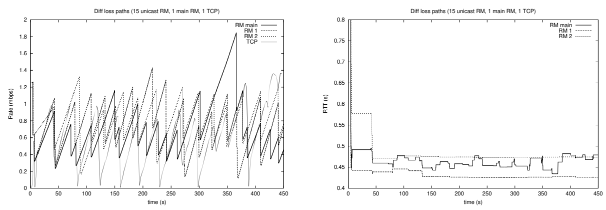


(a) DropTail (b) RED

Figure 14: Droptail vs. RED RTT convergence

3.2.3 Different loss paths

It was speculated that the scheme would not perform as well with the multicast tree having paths with varying degree of loss probailities and RTTs. We perfomed the following simulation to test the robustness of our scheme under these hetrogeneous conditions. Again the toplogy remains the same with modifications in follwing key parameter: The **Buffer** is 125 packets on all links except that it increases on the four links from Rondezvous Point (RP) to Rcvr1, Rcvr2, Rcvr3 and Rcv4 with the values 25, 50, 75 and 100 respectively.



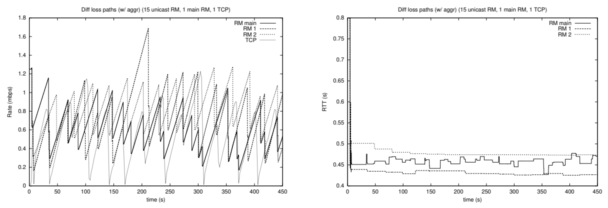
(a) Rate (b) RTTs

Figure 15: Rate and RTT graphs for topology with different buffers (loss rate and RTTs)

The link from RP to Rcvr1 has the smallest buffer(25). Therefore, it has the smallest RTT and the largest loss rate (numbers.txt). On the other hand, the link from RP to Rcvr4 has the largest buffer(100). So it has the largest RTT and smallest loss rate. For the *Main Source*, MaxL-PRF will respond to LEs from Rcvr1, whereas its RTT would be biased higher (avg of the 4 diff RTTS). The simulation is performed for both non-aggregation (Figure 15) and aggregation case (Figure 16). The graphs clearely show that the algorithm works well with aggregation enabled at the router too.

3.3 LI Aggregation Effects

As explained earlier, LI aggregation has an effect on the number of LEs and timing information reaching the source. To evaluate these effects, consider the simple multi-level topology in Figure 17. Observe that the



(a) Rate (b) RTTs

Figure 16: (With aggr) Rate and RTT graphs for topology with different buffers (loss rate and RTTs)

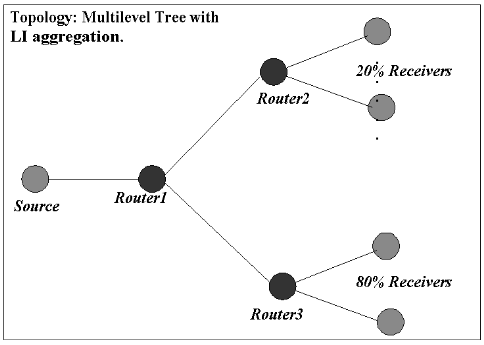


Figure 17: Topology to Illustrate LI Aggregation Effects

number of receivers and their loss rates can be varied. Consider three cases: (a) *no LI aggregation* performed at the routers (b) LI aggregation performed at router 2 and router 3 only (i.e. *partial LI aggregation*) (c) LI aggregation performed at all routers (i.e. *complete LI aggregation*).

Since *ns-2* simulation does not scale beyond tens of nodes, we wrote a C program to numerically simulate the scheme using a model based on fixed loss probabilities of receivers, fixed RTT and packets assumed to be sent in rounds of length RTT. The LIs for a round are computed and informed to the source at the end of the round, and the control decision is enforced in the next round. In Figure 17, we had 20% of receivers in the subtree headed by router 2, and 80% of receivers in the subtree headed by router 3. One receiver in the first subtree has a loss (LI) rate of 4% and all other receivers (in both subtrees) have a loss rate of 2%. A total of 6000 packets were transmitted which translates to a max-LI count of 240 and the observed max-LE count is roughly 150-160 (without aggregation).

The numerical simulation for cases (a), (b), (c) above are shown in Figure 18. The figure illustrates the LEs passed by the cascade of filters (one curve for each case), plotted against the number of receivers (upto 400 receivers). Except for the completely aggregated cases, each curve

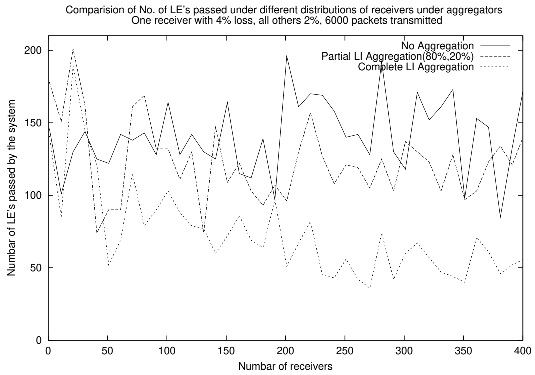


Figure 18: LI Aggregation Effects: LEs Passed by Cascade of Filters

is roughly flat which means that the Max-LPRF passes roughly the same number of LEs irrespective the number of receivers or the total number of LEs generated. Two curves (no aggregation and partial aggregation) are nearly identical which shows that the scheme works well even with non-trivial partial aggregation. We have verified these results for confidence with many randomized variations of the simulation discussed later in the section. This is because the max-LPRF scheme is *more sensitive to LE counts and receiver IDs rather than LI counts/IDs.* However, for the case of complete aggregation, the number of LEs passed is lower, which implies that the scheme will be more aggressive than an equivalent TCP. *In all cases, it does not incur the drop-to-zero problem*, i.e. passing more than 240 LEs.

3.3.1 Worst case aggregation analysis(Numerical)

We analyze the worst case for aggregation, in terms of LEs reaching the source. The topology used is a fanout one, where the source and all receivers are connected to the aggregator. Intitutively, this topology provides for total aggregation, as only one LI among the posible N will reach the source for a particular sequence number. As an example of worst case of aggregation : Suppose one reciver X has a fairly high loss rate, and all others have very low loss rates. Ideally, the number of LE's reaching through the source(after filtering) should be equal to the number of LE's generated by X. Assume that X also has the longest RTT. This implies that X's LI would be supressed, in presence of a LI from any other reciver. We call this situation "Biased Aggregation". As the number of receivers increase, we expect more overlaps between sequence numbers lost by X and other receivers. Lesser and lesser LI's generated by X might reach the source. In the worst case this might reach zero. The source might become more aggresive, as it will get lesser number of LIs after filtering. We use the topology mentioned above for this analysis. We assume one receiver has 20% loss rate, and all others 2%. we study the effect of increasing the number of 2% recivers under the aggregator. The number of LEs reaching the source vs the number of receivers is plotted. Under "biased" aggregation, the number of LEs passed by source drops sharply by increasing the number of 2% recevers. Finally, the source only reacts to 2% loss, instead of 20%. Our scheme becomes overly aggresive in the worst case scenario (Figure 19).

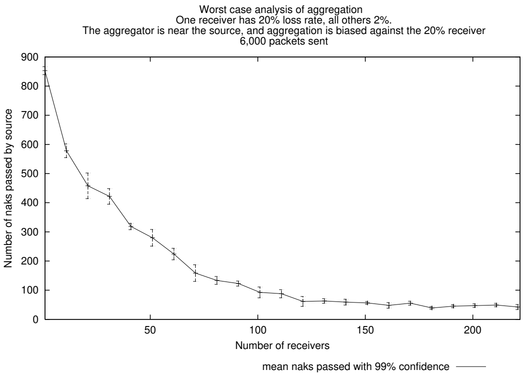


Figure 19: Numerical analysis: Worst case aggregation

3.3.2 Partial Aggregation Numerical Analysis

In the following graphs (Figures 20, 21 and 22) we show a complete range of results in case of partial aggregation. The topology is same as in Figure 17. One receiver has loss probability of 4% while all others have 2%. We vary the distribution of receivers under the two aggregators in the multicast tree. The convention followed is $x - y$, where x% receivers (including the 4% receiever) are on Router2 and remaining y% are under Router3. The following plots show the comparison between no aggregation, full aggregation and varying degrees of partial aggregation. In many cases, even under partial aggregation the performance is close to no aggregation case. The number of LEs passed never drops below the worst case.

4 Linux Implementation and Experiments

We implemented the scheme and performed expriments to verify the correctness of our scheme on a real testbed. Specifically, we tested TCP friendliness and drop-to-zero problems. Implementation differs from simulation as we had to deal with lots of real world issues discussed later.

4.1 Implemenation

The scheme is based on a simplified version of PGM. It has been implemented on top of UDP using RedHat Linux 7.0. The following aspects of PGM have been realized:

- ODATA (original data packet), RDATA (retransmitted data packet), NAK (negative acknowledgement packte), NCF (NAK confirmation packet) packet formats specified in PGM.
- Transmit window at source.
- Receive window at receiver.

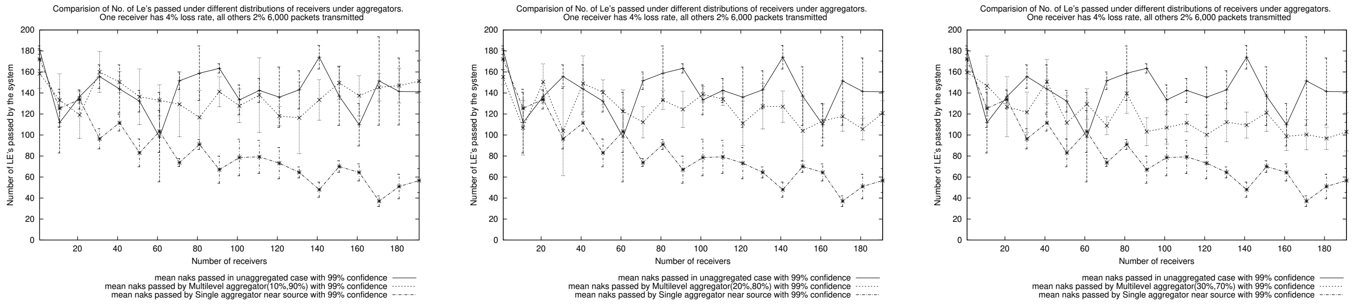
By implementing the features mentioned above, we achieved a reliable transport layer. On the other hand, for the sake of simplicity, we omitted certain aspects of PGM:

- SPM (source path message) packets.
- Network elements support.

SPM packets are sent by source periodically to receivers. They carry information about transmit window which affects the generation of NAK's. To compensate this, each packet from source carries the information about transmit window. Whenever a receiver gets a packet (ODATA, RDATA or NCF), it checks the sequence number continuity and sends NAK's if needed.

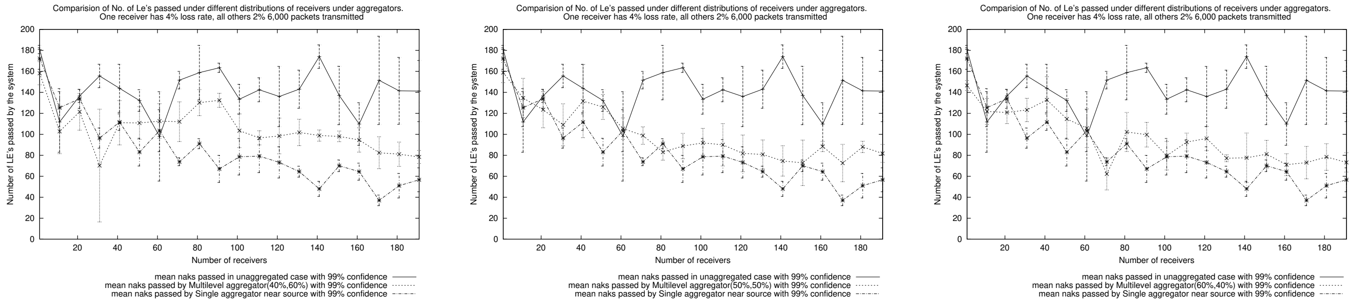
The program runs as a user process without requiring any root privileges. As a result, the granularity of the timer used is 10 it ms which is quite coarse. To reduce negative effects like unnecessary delays, the following mechanisms are used:

- After processing any timeout event, the next event is checked against the current time. If the timeout should actually happen, it will be processed. This process continues until no more timeout events are available. Since the API *gettimeofday()* can return the current time with the μs resolution, the interval between two successive events can be less than 10 it ms, provided events are processed sufficiently fast.
- When a timeout event is scheduled, its timeout value is checked. If its timeout is within 5 it ms (half of the timer granularity) from now, it is triggered. This introduces a random factor into event scheduling, but reduces many unnecessary delays.



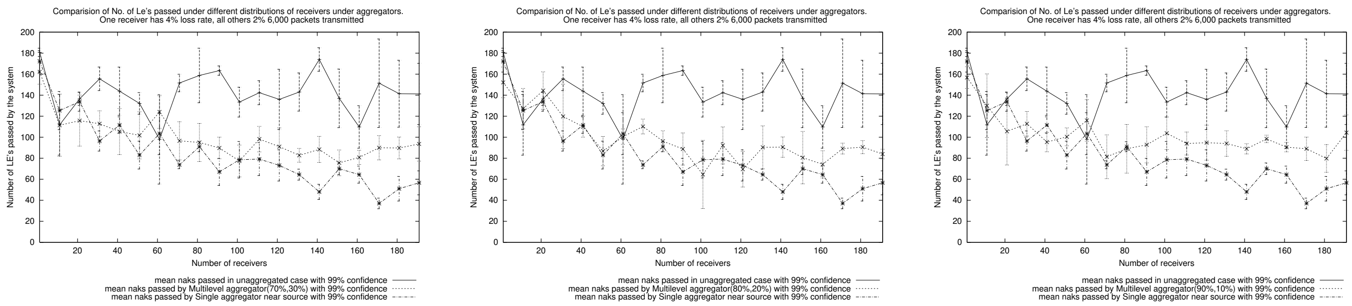
(a) 10-90 (b) 20-80 (c) 30-70

Figure 20: Partial Aggregation Effects



(a) 40-60 (b) 50-50 (c) 60-40

Figure 21: Partial Aggregation Effects (contd.)



(a) 70-30 (b) 80-20 (c) 90-10

Figure 22: Partial Aggregation Effects (contd.)

Using the standard 10 it ms timer introduces accumulative delays. For example, assume a event that should occur 1 it ms after the timer is set, it will actually occur after 10 it ms, which is 9 it ms more than requested, leading to accumulative delays. Our method helps reduce such unnecessary delays.

- We introduces a variant of TBF to control the packet transmission rate at source. Upon arrival of a token, if there are enough packets, the source will transmit as many packets as needed to maintain the required transmission rate. If there are not enough packets, all of them would be transmitted.

By inspecting the time out information, we have obsservered that the mechanisms described above worked as expected.

4.2 Experiments

The experiments were conducted on our testbed consisting of RedHat Linux 7.0 i586/i686 systems. During the experiments, the diff-serv package *tc* in RedHat Linux 7.0 were used to restrict bandwidth and link latency. Also, a multicast router daemon *mrouted* was used to route multicast packets from one LAN to others. Source code and binaries of *mrouted* can be found online at *http://www.vcas.video.ja.net/mice.*

4.2.1 Basic experiments

Some basic experiments, like one source to two receivers transmission, have been done to ensure correctness of the rate control module. The results agree with *ns* simulation. (Graphs are omitted).

4.2.2 TCP friendliness experiment

During the TCP friendliness expriment, TCP SACK was enabled. We used TBF queue discipline of *tc* to restrict the bottleneck bandwidth. The LE-SBCC program runs in user mode subject to priority and timer constraints. At higher link speeds, TCP which is implemented in kernel, might grab a bigger share of the bandwidth. Due to this reason, we set the bandwidth to a lower value than that in simlulation. Details of the configuration are listed as follows:

- Round trip link latency (set by *tc*): 500 it ms
- Bottleneck bandwidth (set by *tc*): 1Mbps
- Source initial transmission rate: 200Kbps
- Source initial RTT estimate: 200 it ms
- Receiver maximum random backoff before sending NAK's: 100 it ms

The fixed part of RTT is the round trip link latency of 500 it ms. Therefore, any RTT sample is at least of that value.

Note that the backoff before sending NAK's increases the RTT. Receivers will wait for that amount of time before sending NAK's, as specified by PGM.

For this experiment, all TCP sources and the multicast source ran on one machine, as in *ns* simulation. Furthermore, behind the bottleneck, there were four receiver machines. To achieve load balancing, each of the receiver machine had four TCP receivers and one multicast receiver. The result, shown in Figure 23, roughly agrees with that of *ns* simulation.

In the graph, the average rate of multicast is higher than that of TCP. The reasons might be:

1. The implementation of our scheme is in user space, thus has lower priority and is less reactive to the network situation than TCP which runs in kernel and has higher priority.

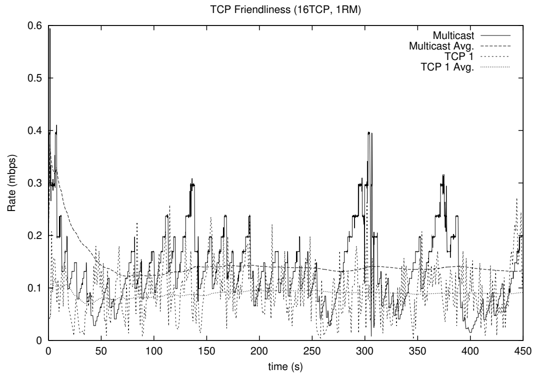


Figure 23: TCP friendliness expriment result (Linux implementation)

2. The timer of our program is coarse, which could make our program less reactive.
3. TCP timeouts are triggered due to small window size, while the multicast scheme does not have such rate-cutting mechanisms.

The TCP flow in the graph was randomly chosen from 16 TCP flows. The results indicate TCP friendliness and drop-to-zero avoidance of our scheme.

5 Summary and Conclusions

We have presented an *LE-oriented, purely source-based multicast congestion control scheme (LE-SBCC)* for reliable and unreliable multicast transport . The model is a simple cascade of filters followed by an AIMD module. In general the AIMD module may also be replaced by a source-based TFRC scheme [6] or rate-based binomial scheme [1]. With partial or no LI aggregation, the scheme correctly filters the LIs for AIMD; avoids the drop-to-zero problem while being TCP-friendly. With substantial aggregation, the scheme is likely to be more aggressive towards TCP, but never incurs the drop-to-zero problem. Like all single-rate schemes, its scalability is limited by the heterogeneity of the tree which may depress the source rate. The scheme is simple to implement and deploy since it is completely contained at the source[7].

[7]Thanks to Supratik Bhattacharya, Sonia Fahmy, Luigi Rizzo, Mark Handley, Lorenzo Vicisano, Jon Crowcroft and Brian Whetten for extensive comments on various versions of this work.

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APPENDIX

A Pseudo Code

event: Send packet:
  Record $T_{send}[j]$ for seq j
event: LI[i][j] received for from receiver i for seq j:
  call **estimateRTT()**
  call **LI2LEfilter()**
  if passed
    call **maxLPRFilter()**
    if passed
      call **ATFilter()**
      if passed
        $rate = rate/2$
      endif
    endif
  endif

estimateRTT() {
  $RTT_{current} = T_{current} - T_{send}[j]$
  $\delta = SRTT - RTT_{current}$
  $SRTT = RTT_{current} + 0.125 * \delta$
  $\sigma = \sigma + (0.125 * (|\delta| - \sigma)))$
}

LI2LEfilter() {
  if $((t - T_{lastPass}[i]) > SRTT + 2\sigma)$
    $T_{lastPass}[i] = t$
    pass LI as LE
  else
    filter LI
  endif
}

maxLPRFilter() {
  update X_i, $\max X_i$, ΣX_i
  $P(accept) = \frac{\max X_i}{\Sigma X_i}$
  Accept LE w/ probability $P(accept)$

```
}

ATFilter() {
  if (CongestionFlag == TRUE)
    filter LE
  else {
    SilenceFlag = TRUE
    SilencePeriodTimer = RTT/2 + 2σ
    CongestionFlag = TRUE
    CongestionEpochTimer = SilencePeriod + SRTT + 4σ
    Accept LE
  }
}

event: CongestionEpochTimer expires:
         CongestionFlag = FALSE

event: SilencePeriodTimer expires:
         SilenceFlag = FALSE

event: RateIncreaseTimer expires:
         if (CongestionFlag == FALSE)
           rate+ = MSS/(SRTT+2σ)
         else {
           if (SilenceFlag == TRUE)
             no data transfer
           else
             no increase in rate
         }
         RateIncreaseTimer = RTT + 2σ
```