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Multi-rate Congestion Control using Packet-pair Bandwidth Detection with Session and Layer Changing Manager

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Abstract*

Multi-rate multicast is the most efficient way to support the future high speed, bandwidth varying and heterogeneous network. Past research into layered multicast protocols has mainly focused on the effectiveness of determining the maximal number of layers that can be subscribed to by each receiver, and the fairness issue with different sessions, especially for TCP. In this paper, we describe a new multi-rate multicast congestion control protocol called *P*acket-pair bandwidth detection with *S*ession and *L*ayer changing *M*anager (PSLM). PSLM can treat different sessions with different priorities and guarantees high transmission quality for important sessions. PSLM also addresses the layer stability issue to improve consistent quality requirement, as the support for stable network transmission. From our simulation results PSLM can not only achieve the above two requirements but also keeps all the advantages from the original PLM protocol.

1. Introduction

Multicast solves the inefficient use of bandwidth problem of unicast and also the insecurity problems of broadcast. It is an efficient way for group communications in IP networks. However, deployment of IP multicast on the Internet has not been as rapid as expected. Among the reasons behind this slow deployment is the lack of efficient multicast congestion control schemes [1]. According to [2], there exist two kinds of multicast congestion control. With a *S*ingle-*R*ate *M*ulticast *C*ongestion *C*ontrol (SR-MCC) scheme, all receivers in a multicast session receive the data at the same reception rate. The scheme picks one of the slowest receivers as representative. A single sluggish receiver can retard the reception rate of all other receivers that may be in better network conditions. For the *M*ulti-*R*ate *M*ulticast *C*ongestion *C*ontrol (MR-MCC) scheme, there is a layer with the highest importance, called a base layer. It contains the most important features of data for decoding by the receiver. Additional layers, called enhancement layers, provide increasingly better quality. The more layers receivers subscribe to, the more bandwidth is consumed, and the better quality received. MR-MCC is a very good solution for a complex heterogeneous network and also it is very good at supporting communications over wireless networks with their limited bandwidths and higher, varying, error rates [3]. All our following study focuses on the MR-MCC.

The technique of cumulative layered multi-rate multicast congestion control was first proposed by McCanne et al. [4] in the context of packet video transmission to large heterogeneous audiences. Their *R*eceiver-driven *L*ayered *M*ulticast (RLM) protocol achieves scalability by using a methodology, in which the hosts tune their subscription level by joining and leaving layers. In order to address the problems of RLM, Vicisano, Crowcroft and Rizzo developed the *R*eceiver-driven *L*ayered congestion *C*ontrol (RLC) protocol [5]. The use of bandwidth of each layer in RLC is increased or decreased exponentially according to the detecting of the available bandwidth. This achieves the same rate adjustment scheme as TCP. To prevent inefficient uncoordinated actions of receivers behind a common bottleneck, an implicit coordination is done using synchronisation points (SP). *F*air *L*ayer *I*ncrease *D*ecrease with *D*ynamic *L*ayering (FLID-DL) is a protocol for improving RLC which is presented in [6]. FLID-DL uses a Digital Fountain encoding [7], allowing a receiver to recover the original data upon reception of a fixed number of distinct packets, regardless of specific packet losses. FLID-DL introduces the concept of *dynamic layering* to reduce the IGMP leave latencies. *W*ave and *E*quation *B*ased *R*ate *C*ontrol (WEBRC) is the first multiple rate multicast congestion control protocol to be *equation based* [8][9][10]. It has two major innovations: MRTT and Wave. MRTT is a multicast analogue of the unicast round trip time (RTT). Wave is the transmission rate on a channel is periodic, with an exponentially decreasing form during an active period followed by a quiescent period.

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The *P*acket-pair receiver-driven cumulative *L*ayered *M*ulticast (PLM) congestion control protocol was proposed by Legout et al. [11] for audio/video or file transfer application. A fundamental different approach compared with above protocols is the use of two key mechanisms: *F*air *Q*ueueing (FQ) and receiver-side *P*acket-*P*air (PP). Networks with FQ have many characteristics that immensely facilitate the design of congestion control protocols and also improve TCP-friendliness. In the FQ network environment, the bandwidth available to a flow can be determined using the PP method. PP can sense the bandwidth change in the network before congestion happens, which the join experiment, used by other protocols does not. These let PLM have a good performance evaluation reputation compared with other MR-MCC protocols. Somnuk [11] and other researchers [1][2] have already done very good performance evaluation analysis for PLM.

All these above protocols treat each session independently without regard to other sessions even when they are strongly related to each other, such as audio/video traffic. Flexible bandwidth allocation algorithms for the sessions are required. In this paper, we describe a new multicast congestion control protocol -- PSLM, which is based on PLM (Figure 1). PSLM monitors the different priorities for different sessions and judges whether and how it can change the bandwidth allocation to guarantee the required transmission quality for the high priority sessions. Also in order to avoid the regular layer re-subscription and leaving, PSLM introduces a new layer changing control algorithm to avoid unnecessary layer changing when the network is in a stable situation, which means there is no significant bandwidth change for the bottleneck. Finally, our PSLM protocol requires no additional router support beyond basic multicast functionality, makes no new demands on any multicast protocols, and is suitable for mapping to all the other existing MR-MCC protocols.

The remainder of this paper is organized as follows. Section 2 presents the new PSLM protocol. Section 3 provides the simulation results. We conclude with directions for future work in Section 4.

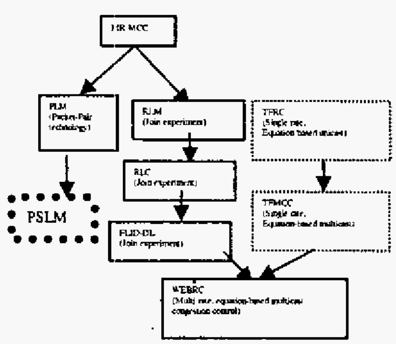

Figure 1. Classification of MR-MCC

2. PSLM

PSLM is designed to provide multi-session management and smooth layer change over time. It is a new protocol which is based on the PLM, continuing to use Packet-pair (PP) for bandwidth detection. There are two innovations for PSLM (Figure 2): *S*ession *M*anager (SM) and *L*ayer *M*anager (LM).

2.1. Session Manager (SM)

PLM is a protocol which simply requires all the sessions to be fairly treated, sharing the available bandwidth. This lets PLM be very TCP-friendly, but fairness is not everything. Assume we have an audio session and a video session respectively. Audio sessions do not have large bandwidth requirement but a small transmission break can lead to a noticeable loss of quality for the audiences. A video session needs a large bandwidth, but the loss of a few frames causes nothing serious. It is clear to see that, during a congested situation, if we simply decrease all the bandwidth allocations equally for all the sessions, it is unfair for the low bandwidth, audio, session and unnecessary for the large bandwidth, video, session. The *S*ession *M*anager (SM) is an extension function for the receiver which is based on the original PLM protocol. The main work of SM is to monitor the priorities of all the receiving sessions and control the bandwidth allocation for these sessions with respect to each other and guarantee the required transmission quality for the most important sessions.

In SM, it is necessary to set priorities for different sessions and the number of concurrent sessions from the sender point. All these are defined by the user. When the receiver receives sessions, it checks the number of concurrent sessions of various relative priorities first. If there are not relatively higher priority sessions, it treats them all fairly. Otherwise, according to their priorities, the receiver changes the bandwidth allocation for the different sessions. The higher priority session receives higher bandwidth. Also, during congestion, SM changes the session layer rate, decreasing it according to the priorities of the sessions. Higher priority sessions have a lower decrease in rate, in order to let the lower priority sessions give up their bandwidth first.

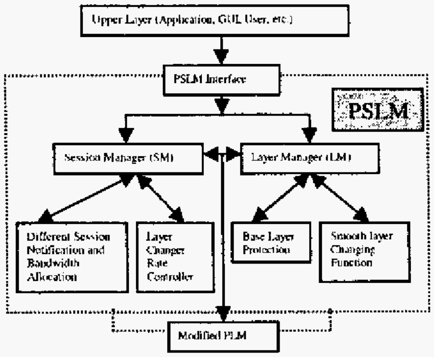

Figure 2. PSLM Architecture with Modified PLM

2.2. Layer Manager (LM)

An important metric for evaluating video/audio distribution protocols and end-user experience is the stability of the service quality. Unfortunately, adding or dropping layers to react to changes in network load in PLM depends only on the PP checking value. If the PP checking value is larger than the subscription layer value and more layers are available for subscription, PLM adds the layer immediately. Otherwise, if the PP value is less than the subscription value, PLM drops layers until the subscription value reaches the PP value. Due to the combined effect of dynamic variation in the capability of the network to carry traffic with the changes in traffic towards the receivers in response to the layers added, or removed, as the PLM entities converge to the PP checking value, there will be a tendency to shift the level of layer subscription continuously. The *L*ayer *M*anager (LM) is executed to address this problem. We first define this kind of layer adding and dropping as *U*nstably *C*hanging (UC), and then count the number of the UC adjustments. Every time such an adjustment happens the LM does not subscribe to an enhanced layer immediately, but increases the next PP checking time by 10 sec. If this persists for five cycles, we can assume that the network is in a relatively stable condition. PSLM then subscribes to an enhanced layer, increases the PP checking time by 60 sec and resets UC back to 0.

Depending by the different session priorities, LM also provides the function for protecting base layers. During congestion, the LM drops the lower session enhanced layers first, and then drop the higher session layers. Only after all the enhanced layers have been dropped will the LM start to drop the lower session base layer and then higher session base layer. This function ensures the important information has better protection.

3. Simulation and Evaluation

In this section we present simulation results of a simple topology to evaluate the operation of PSLM. We implemented the PSLM in network simulator ns2 version 2.27 [13], and ran simulation experiments in many network conditions. We compared the layer changing performance evaluation results of the two sessions PLM, with two sessions PSLM, first only including the SM, and then together with the LM. At last, we have done a general performance testing for PSLM.

3.1. PLM vs. PSLM

To make the experiment simple, we assume that original media data for both two sessions' base layer is encoded for 150Kbps, each enhanced layer rate is 50Kbps, packet size is 1024 byte, check value C = 1. The simulation topology is illustrated in Figure 3. It consists of two sources (S_1, S_2), one receiver (R) and the bottleneck between the two routers (B_0, B_1). The bottleneck is set to be 450Kbps. The bandwidth of other links is 10Mbps, which is rich enough not to cause congestion. Receiver (R) starts receiving S_1 at 5 sec and S_2 at 250 sec.

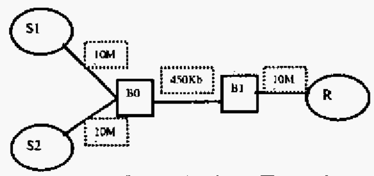

Figure 3. Simulation Topology

Figure 4(a) shows the bandwidth allocation when receiver runs PLM to receive each stream independently only considering the absolute fairness to other sessions. The PLM2 session joins at 250 sec. The PLM1 session quickly decreases its layer subscription to the layers 1 or 2 to share the bandwidth with the PLM2 session. We can see it very clearly: PLM1 and PLM2 share the bandwidth equally and cooperatively. Both of them can only get one enhancement layer as the maximum layer for subscription. Also, before PLM2 joins the bandwidth competition, there exists significant layer shifting for the PLM1 session between layers 5 and 7. After 250 sec, with both of these two sessions active, there is still layer shifting between layer 2 and 1, even under the relatively stable network situation. Figure 4(b) gives us the throughputs of theses two sessions. The layer encoding parameter settings and the bandwidths allocation can be clearly seen in Figure 4(a).

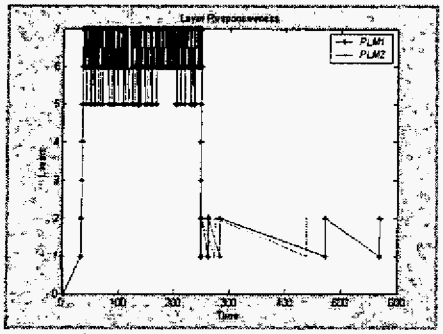

Figure 4(a). PLM in two sessions (layer)

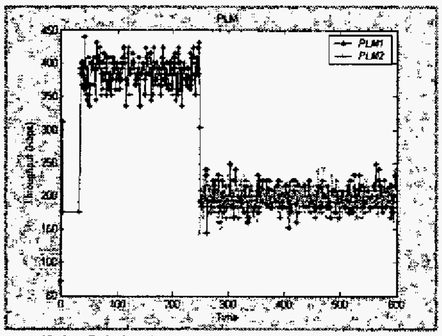

Figure 4(b). PLM in two sessions (throughput)

3.1.1. PSLM: Session Manager (SM) The *S*ession *M*anager (SM) of PSLM has been developed to address the absolute fairness problem in PLM. In our example, we use the same topology and parameter settings as in the PLM simulation, but use the PSLM protocol with SM instead of PLM. We set the PSLM2 session to have higher priority than the first PSLM1 session. From the simulation results (plotted in Figure 5), the two sessions still react quickly to the changing of network situation, only taking about 1 sec to respond, and both of these two sessions keep a "friendly" relationship to each other. However different sessions have different bandwidth allocations and the higher priority PLSM2 session has more layers to subscribe to. Note, also, from Figure 5, that there is a high rate of variation in the number of layers that each session subscribes to. This is mainly caused by our unfair bandwidth allocation algorithm.

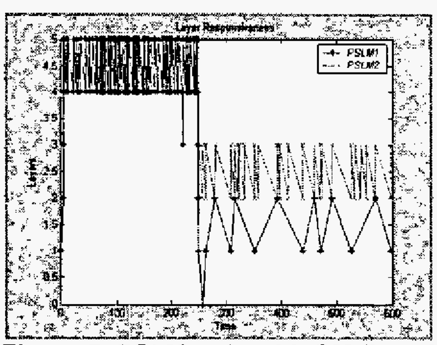

Figure 5. PSLM only with SM (layer)

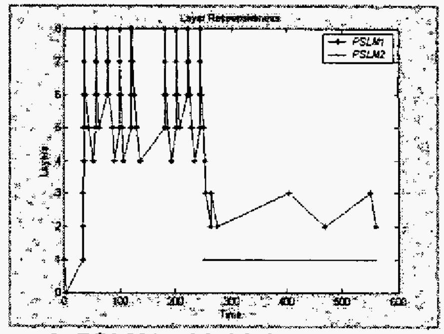

Figure 6. PSLM1 with higher priority (layer)

3.1.2. PSLM: Layer Manager (LM) The *L*ayer *M*anager (LM) is used to avoid unnecessary layer subscription when the network situation is relatively stable. The same topology and parameters setting as before are used, and in this testing we execute the entire PSLM protocol with both SM and LM. At the first experiment, we set PSLM1 at higher priority. From the simulation results shown in Figure 6, PSLM1 has more layers to subscribe to and PSLM2 can only get the base layer. Comparing with the SM test, there is a significant improvement in avoiding unnecessary layer shifting. Before 250 sec. we can clearly count the number of the layer shifts for the first session. When the second session comes at 250 sec, PSLM1 still keeps the advantages from PLM, decreasing its subscription very quickly in reaction to the bandwidth reduction and SM's actions, but still having larger bandwidth allocation. This time PSLM1 still gets 2 layers for subscription, but it shifts only rarely up to layer 3. The PSLM2 session can only get one base layer, with limited bandwidth because of its lower priority, but it keeps this one layer very stable, with no change during the whole simulation time. Figure 7(a) gives us the complementary simulation results, in which we give PSLM2 higher priority. Figure 7(b) is the throughput response for this second experiment. From Figure 7(a) and (b), the layer changing and throughput can be exactly matched with each other.

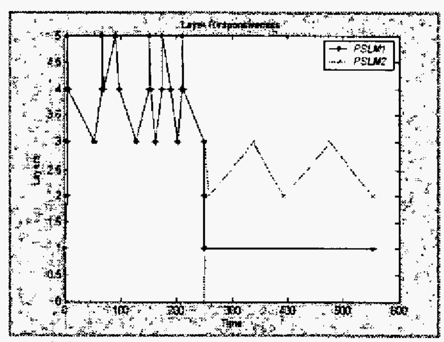

Figure 7(a). PSLM2 with higher priority (layer)

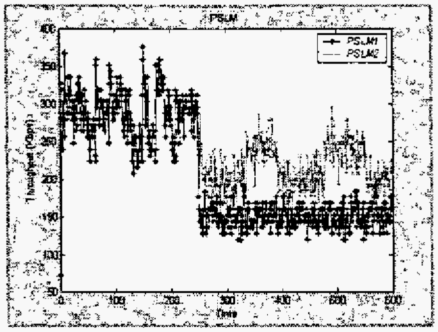

Figure 7(b). PSLM2 with higher priority (throughput)

3.2. PSLM General Testing

In this section we describe the general performance evaluation testing for PSLM. This includes responsiveness for network situation changing reaction testing and equal-sharing for TCP-friendly testing. All these experiments we assume they are equal important and haven't add any priority setting for different sessions.

3.2.1. Responsiveness We use constant bit rate (CBR) transmission as a simple check to see whether PSLM reacts to varying available network bottleneck bandwidth smoothly and efficiently. The topology is shown in Figure 8. In order to make sure the bandwidth can be totally subscribed to (in our example it is 1Mbps), we set the available largest PSLM server and host session rate to be 1.5Mbps.

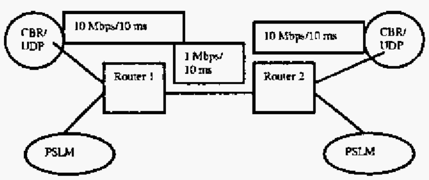

Figure 8. PSLM Responsiveness Test Topology

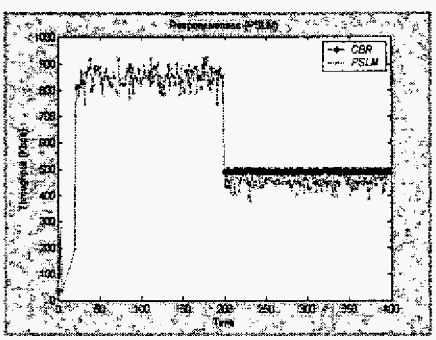

Figure 9. PSLM Responsiveness Testing

We use a single PSLM multicast session across the bottleneck, where the server starts at time 3. After 200 sec we add a CBR source over the bottleneck link at 500Kbps (half the bottleneck). We run the simulation for 400 seconds. From Figure 9 we can see that before 200 sec PSLM on its own can make best use of the bandwidth, about 900Kbps. When it detects the 500Kbps CBR stream starting at 200 sec. it only uses few seconds to reduce its rate to about 500Kbps to equally share the bandwidth with the stream.

3.2.2. Equal-sharing In the equal-sharing test, two TCP sessions share a common bottleneck with a single PSLM session. The goal is to see how PSLM competes with TCP traffic over a common bottleneck and demonstrate TFRC. In our simulation experiment, we set the TCP windows at 5000 bytes to remove any effect on the data rate generation from the influence of maximum window size. We also use the same PSLM session rate setting as in the responsiveness test, above, to ensure that the session bottleneck behaviour would be identical. The equal-sharing topology is showing in Figure 10.

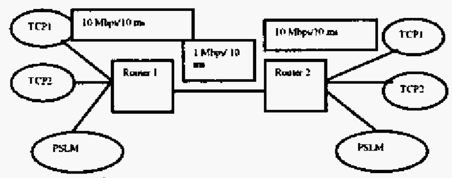

Figure 10. PSLM Equal-sharing Topology

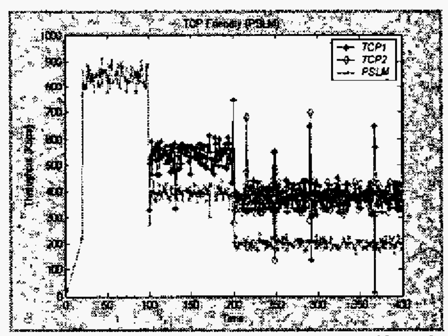

Figure 11. PSLM Equal-sharing Testing

The PSLM session starts at time 3 sec and quickly consumes the whole bandwidth. We start the TCP1 session at 100 sec and the TCP2 session at 200 sec. We run the simulation for 400 seconds. Figure 11 gives the throughput results for this experiment. At time 100 sec, PSLM responds to the new TCP1 very quickly. Both sessions share the bandwidth not equally but cooperatively with about 400Kbps and 550Kbps each. Then at 200 sec PSLM and TCP1 decrease their transmission rates in response to the new TCP2 session. At the end of the plot, the two TCP sessions' throughputs are about 400Kbps and the throughput of PSLM session has decreased from over 800Kbps to about 200Kbps.

4. Conclusion

A cumulative layered technique is the general recommended solution to the problem of heterogeneity of receivers and network links in multicast applications. In this paper, we first thoroughly investigate and classify the most important existing MR-MCC protocols. Then we analyze the problems which have not been addressed by theses previous proposals. After that, we present a novel packet-pair based layered multicast congestion control mechanism, PSLM, which is based on PLM. In PSLM we define two main functions *S*ession *M*anager (SM) and *L*ayer *M*anager (LM) to solve the priority bandwidth allocation problem and smooth layer changing algorithm for stable network states. Then, we give our experimental simulation results. From our results, we believe our SM and LM meet the requirements set for congestion management and layer subscription optimization, and are also suitable for transferring to other protocols which use different bandwidth detection methodologies.

The PSLM protocol assigns the priorities for different sessions at the parameter setting stage. This allocates the sessions' bandwidth before the start of transmission, so lower priority session cannot fully use the bandwidth when it starts first without other sessions. Also PSLM can avoid the unnecessary layer subscription, but it causes the layer shifting range to get large. These are topics for future research to improve the performance of PSLM, as are other approaches to reducing congestion.

5. References

[1] A. Matrawy and I. Lambadaris, "A Survey of Congestion Control Schemes for Multicast Video Applications", IEEE Communications Surveys & Tutorials, Fourth Quarter 2003.

[2] J. Widmer, R. Denda and M. Mauve, "A Survey on TCP-Friendly Congestion Control", IEEE Network May/June 2001.

[3] S. Puangpronpitag, "Design and Performance Evaluation of Multicast Congestion Control for the Internet". PhD thesis, The University of Leeds, November 2003.

[4] S. McCanne, V. Jacobson, and M. Vetterli, "Receiver-driven Layered Multicast", In Proceedings of ACM SIGCOMM, August 1996.

[5] L. Vicisano, L. Rizzo, and J. Crowcroft, "TCP-like Congestion Control for Layered multicast Data Transfer", In Proceedings of IEEE INFOCOM, April 1998.

[6] J. Byers, M. Frumin, G.Horn, M. Luby, M. Mitzenmacher, A. Roetter, and W. shaver, "FLID-DL: Congestion Control for Layered Multicast", In Proceedings of 2nd Workshop on Networked Group Communication (NGC2000), November. 2000.

[7] J. Byers, M. Luby, M. Mitzenmacher and A. Roetter, "A Digital Fountain Approach to Reliable Distribution of Bulk Data". In Proceedings of ACM SIGCOMM, September 1998.

[8] M. Luby, V. Goyal, S. Skaria, and G. Horn, "Wave and Equation Based Rate Control Using Multicast Round Trip Time", In Proceedings of SIGCOMM'02, August 2002.

[9] M. Luby, V. Goyal, "Wave and Equation Based Rate Control (WEBRC) Building Block", IETF Reliable Multicast Transport Working group Internet-Draft, February 2002.

[10] M. Luby, V. Goyal "Wave and Equation Based Rate Control Using Multicast Round Trip Time: Extended report", Technical Report DF2002-27-001, Digital Fountain, July 2002.

[11] A. Legout and E. W. Biersack, "PLM: Fast Convergence for Cumulative Layered Multicast Layered Multicast Transmission", In Proceedings of ACM SIGMETRICS, June 2000.

[12] S. Puangpronpitag, R.D. Boyle and K. Djemame, "Performance Evaluation of Layered Multicast Congestion Control Protocols: FLID-DL vs. PLM" , In Proceedings of International Symposium on Performance Evaluation of Computer and Telecommunication System, July 2003.

[13] NS. UCB/LBNL/VINT Network Simulator - ns (version 2), http://www-mash.cs.berkeley.edu/ns/.