SPLIT: A Priority Based TCP-Friendly Video Protocol

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1. Introduction

Multicast distribution of video is one of the most important emerging Internet applications. The heterogeneity and scalability of the Internet makes video multicast a challenging problem. In particular, for IP-based networks, there are some shortcomings for real-time video transmission. The ability to transmit live video, such as video conferencing and telecommuting will open up new opportunities for the Internet technologies.

There are two different approaches to achieve multirate transmission, i.e. the sender-driven and the receiver-driven multicast [1]. Sender-driven transmission relies on feedback sent by the receiver. This may cause feedback implosion, which can deteriorate the sender's performance [3]. On the other hand, layered video can be transmitted via multi-session multicast in the receiver-driven approach [2, 4, 5 and 6]. Compared with the sender-driven approach, receiver-based approach provides a better solution for heterogeneity and scalability [7].

A study by Bajaj et al. in [9] found that the performance benefit of priority dropping for layered video is lower than expected. Recent research work by Young-Gook Kim et al in [10] advocated a new packet classification scheme and shows that this mechanism is used to isolate the packet loss event to the lowest priority layer.

In this paper, we propose a TCP-friendly layered video multicast algorithm which provides a feasible solution to both congestion control and error control in the Internet environment. We first introduce a new receiver oriented multicast congestion control algorithm SPLIT [12]. Secondly, we apply the priority dropping mechanism (PDM) advocated by [10] to SPLIT using RED (Random Early Detection) queue. Since the proposed PDM along with packet classification scheme can distinguish the priority at the packet level within one layer, it can be applied to any type of video transmission over the Internet.

The rest of the paper is organized as follows. In section 2, we introduce SPLIT receiver oriented multicast algorithm, SPLIT packet format, SPLIT mechanism. Packet classification scheme with video layering is described in section 3. RED queue with PDM is described in section 4. The NS-2 simulation topology and experimental results are discussed in Section 5. Finally, concluding remarks and future work are given in Section 6.

2. Proposed Split-Layer Video Multicast Protocol

By 'splitting' each encoded video layer into two streams, SPLIT is able to provide an end-receiver with the most relevant video data so that the error concealment techniques can better reproduce the encoded video under lossy conditions.

SPLIT works by having the source S encode n ($n > 1$) video layers (V) with V_1 as the base layer and every additional layer V_2... V_n as enhancement layers. Each layer is then 'split' into two streams V_nHP and V_nLP, where V_nLP contains approximate 1/n-1 of Vn, for n = 2, 3,.

V_nHP and V_nLP are then transmitted to separate multicast address at a high and low IPv6 priority field respectively. The delivery of V_nHP can also be optionally enhanced with the use of FEC (Forward Error Correction).

Each destination wishing to join a video session will begin by subscribing to the base layer (V_1HP and V_1LP). If there is no congestion after time t, the receiver will add V_2HP and V_2LP. Again, after time t and if there is still no congestion, the process will be repeated until either the receiver has joined all 2n multicast sessions or congestion is detected.

Destination D that has subscribed to m video layers (i.e. from layers V_1HP and layers V_1LP to V_mHP and V_mLP) will drop V_mLP if:

(i) congestion (determined by packet loss rate) is detected; and
(ii) D has not recently received a join experiment message.

Subsequently D will begin using the hybrid loss concealment to estimate the data lost from V_mLP. If packet loss rate is still too high, the layer containing V_{m-1}LP will be dropped and the amount of data lost is estimated, and so on. If after dropping layers V_mLP ... V_1LP, congestion still remains a problem, then layer V_mHP is dropped; but layers V_{m-2}LP ... V_1LP are reinstated. The sequence is repeated until acceptable packet loss is obtained or only layer V_1HP remains.

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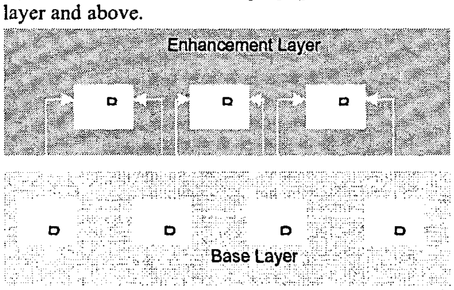
Let receiver R1 currently subscribes to layer n and receiver R2 (sharing a bottleneck point with R1) subscribes to layer n + 1. If both receivers suffer congestion from the same cause (i.e. at the shared bottleneck) and receiver R1 drops layer n, then this will have no effect on the congestion (unless R2 drops layers n + 1 and n). Therefore, the acceptable length of waiting time for a receiver to drop a layer will be a function of the number of layers the receiver currently subscribes to. After dropping a layer, a receiver will send a drop layer message stating the layer number that has been dropped. This is to ensure that receivers in the area receiving a lower layer can hold off from dropping a lower layer until surrounding receivers drop layers higher than the layer the receiver currently subscribes to. After a successful layer drop, the receiver will send a no-congestion message to surrounding receivers so that any lower-layer-subscribing receiver that is still congested can proceed to drop appropriate layers. Any receiver that feels it has been in a state of congestion for too long can drop the appropriate layers.

If the experiment fails, the layer is dropped and any other receivers in the area that are affected by the congestion do not drop layers until some time after the experiment. After the first failed join experiment, a receiver will wait s seconds before attempting another join. If the second experiment is also a failure, the receiver will wait 2 * s seconds. If there is a third consecutive failed join experiment, the receiver will wait 2 * (2 * s) and so on.

If the experiment is a success a join success message is sent so that any receiver, which suffers congestion shortly after receiving a join message, can begin to drop layers.

3. Video Layering and Priority Dropping

This scalability is achieved by using intraframe coding (I – frame), predictive coding (P-frame), and bidirectional predictive coding (B-frame). The dependency of these frames is shown in fig 1. As shown in the figure, each layer is dependent on the next lower layer. Thus, the packet loss from a lower layer results in severe error propagation over the same layer and above.



As shown in fig 1, the base layer (lowest layer) is the most important one and so it gets the highest priority. Given the priority to each layer or each packet in a layer, a router can selectively drop packets from lower priority layers or packets of lower priority. This will isolate loss region to the enhance layers of the current layer subscription. This will also minimize error propagation and also reduces redundancy due to proactive error control such as FEC [13].

In our experiments we use priority classification scheme (PCS) [10] to assign priority to each packet generated by SPLIT. We assign priority zero (0) to TCP packets to make SPLIT TCP friendly. It is assumed that each layer has a Constant Bit Rate (CBR). For a cumulative rate layer, if there are L layers then the total target rate can be written as

$$R = \sum_{i=1}^{L} r_i \qquad (1)$$

Where R is the total target rate and r_i is the target rate of layer *i*. In our experiments layers are numbered from 1, which starts with base layer.

$$P_{\min}(l) = P_{\max}(l-1) + 1 \qquad (2)$$

$$\mathrm{Pmax}(l) = \frac{\sum_{i=1}^{l} \mathrm{r}_i}{\mathrm{rbasic}} \qquad (3)$$

$$P_{avg}(l) = \frac{P\min(l) + P\max(l)}{2} \qquad (4)$$

$$\sum_t P_l(t) \le \frac{r_l}{r_{basic}} * t \qquad (5)$$

where r_{basic} is the minimum value of the target rate, P_{max}, P_{min}, P_{avg}, P_l are the maximum, minimum, average and total values of priority of layer *l* respectively. If n bits are used, then $2^n * r_{basic}$ is the maximum rate to assign the priority. In our experiments we used priorities between P_{min} and P_{max} assign to each SPLIT packet. We adapt the idea of sending video packets over various layers and combine it with priority dropping to improve video quality and achieve inter-section fairness. We assigned highest priority to base layer and lowest priority to higher layers using equations (2) and (3), and also bounding to equation (5).

4. RED with priority dropping

At the routers we used RED queue with priority dropping. We used similar model as described in [10]. We added priority mechanism to exciting RED queue so that it can drop from the lowest priority as shown in equation (6). In normal RED queue, when it drops a packet, it can select the packet to drop with three ways, from the tail, from the front or randomly. Among them, random selection is the best way to prevent global synchronization, and it distributes packet loss uniformly to existing flows [14]. In our RED with priority we select the packet to drop with the lowest priority as given by PCS explained in previous section.

Since TCP traffic does not have its priority, we set its priority value to 0. The ultimate goal of PCS is that the receiver does not experience any packet loss for higher priority packets when there is a lower priority packet.

```
If (enqueue) Update Queue Statistics
  If (Random Drop) then
    If (P==0) drop /* TCP packet*/
      else if (P<Max (P)) drop first packet among P=MAX (P)
  else drop the packet
Update Queue Statistics                                   (6)
```

As described in [14] the performance of the queue strongly dependent upon queueing delay and RED parameters. If the queue is largely enough, there is very less chance of losing higher priority packet.

5. Simulation Topology

The sample topology consists of a single source, three routers and six destination nodes. Each node is connected at a different bandwidth ranging from 1Mbs to 10Mbs. The source will be transmitting a scaled five-layer stream, consisting of 1Mbs per layer with a packet size of 1Kb. The SPLIT source will 'split' each layer into a high and low priority streams at a ratio of 4:1. (i.e. 80% for high priority and 20% for low priority streams). This will be simulated in Ns-2 as a ten (five high and five low priority) constant bit rate (cbr) flows. We use TCP as the back ground traffic. Each TCP frame has a unique flow-id so that it could be identified at the receiving end. Also each video has a unique flow-id so that it could be identified at the receiving end. At t= 0.00 Sec source transmits the CBR data stream to all the receivers. At t=0.02 Seconds TCP traffic begin to transmit to destination 4 with the interval of 10ms.

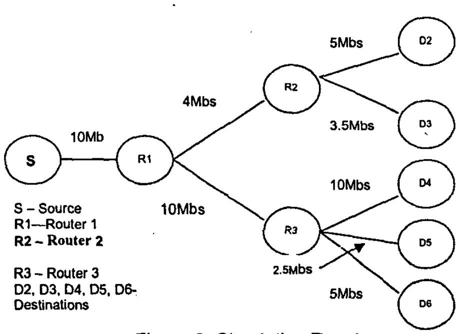


Figure 2: Simulation Topology.

6. Simulation of SPLIT and Discussion

6.1 Packet loss in the RED queue

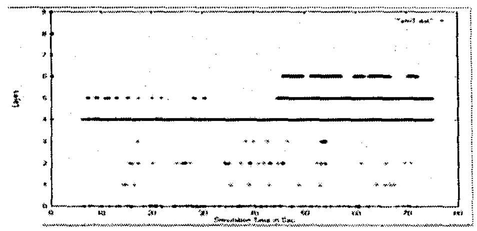


Figure 1: Packet Loss in different layers with PDM.

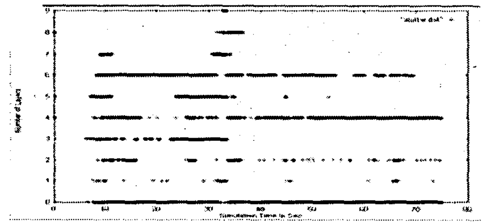


Figure 2: Packet loss in different layers without PDM

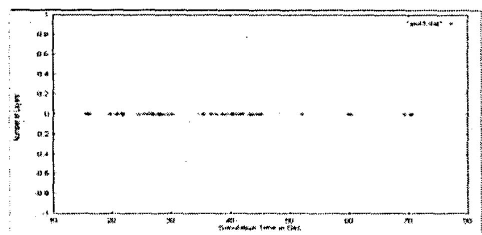


Figure 3: CBR packet loss in the base layer with PDM

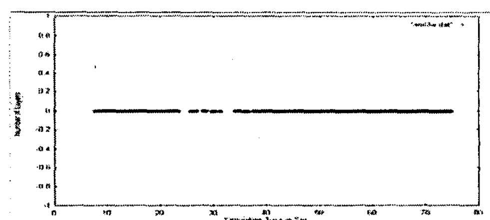


Figure 4: CBR packet loss in the base layer without PDM.

6.1.1 Discussion

We can observe the effect of priority dropping mechanism from the figures 1, 2, 3 and 4. In figure 1 number of packets dropped in the base layer is considerably small, as we move on to the upper layers packet drops are increasing. This effect is due to Priority Dropping Mechanism (PDM) which protects high priority packets dropping from the base layers. This drop in the base layer is due to the CBR traffic. As shown in the figure 3 CBR packet drop is very low. Hence, from our analysis, we can say that this CBR packet drop is due to frequent *join* and *leave* experiments employed by different receivers. Without priority drop mechanism (PDM), most of the CBR packets are lost in the RED queue due to default drop tail mechanism. Packet drops are isolated from high priority layers to low priority layers. Without Priority dropping mechanism, the loss (CBR Packet loss) in the base layers increases with the number of flows. SPLIT based application with PDM (Priority Drop Mechanism) will have better quality in event of congestion because whenever there is congestion in the network most of the packets which violate the available rate constraint are dropped in the lower priority layers.

6.2 Drop in the individual Layers

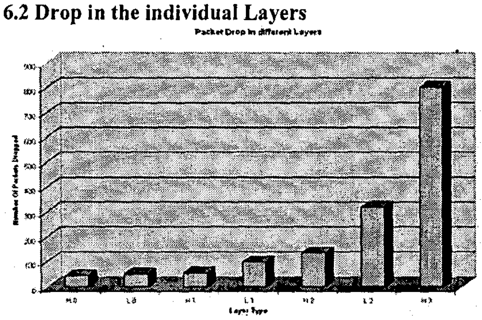


Figure: 6 Packet Drop in individual Layers.

6.2.1 Discussion: In the above graph all *Hi* depicts high priority layers and *Li* depicts low priority layers. From the above graphs, we can see that packets are lost in all layers regardless of their importance. If we are moving from Base layers to Enhancement layers (i.e. from H0-H3, L0-L3) the packet drop rate is increasing linearly. In the above figure 6, due to bursty TCP and CBR traffic base layer drops are noticed and H3 layer (Enhancement Layer3 High priority Stream) experiences huge packet loss. In order to avoid congestion SPLIT receiver is dropping packets in this H3 layer. However, in the proposed system, video layers of high priority do not experience much loss because SPLIT with PDM mechanism makes all the lost packets concentrated in lower priority layers. This mechanism drops all packets, which violate the rate constraint in the RED queue to avoid congestion. Also, each SPLIT receiver individually adjusts its reception bandwidth by selecting a suitable subset of layers in order to control packet loss over a congested link. This is clearly shown from figure 6 that high drops are occurring at low priority layers. This ensures a high quality video to the end user in case of packet loss.

6.3 Results Analysis

6.3.1 Receiver 1 (1 Mbs Link Bandwidth). The throughput of Receiver 1 is shown in Figure 7 and 8.

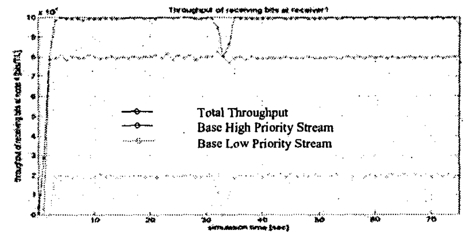


Figure 7: SPLIT receiver 1 Throughput without PDM

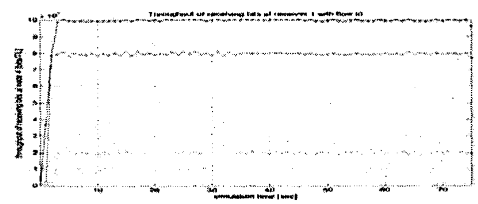


Figure 8: SPLIT receiver 1 throughput with PDM.

Discussion:

In this first experiment receiver 1 (node 4) is connected to the source at a bottleneck speed of 1Mbs. It can be seen from figures 7 and 8 that SPLIT with PDM and SPLIT without PDM are able to maintain the same level of throughput. In both situations SPLIT is able to subscribe to base layer and available bandwidths are fully utilised. From the figure 7 performance of SPLIT without PDM is reduced at 30-35 sec of the simulation because of more frequent join experiments.

After 4 secs receiver 1 was able to subscribe to the full base layer. This occurred because available bandwidth was not sufficient enough to enable the receiver to subscribe to the high priority streams of both the base and first enhancement layer.

Overall SPLIT was able to perform adequately in this experiment. The available bandwidth was fully and effectively used, and the end user was able to receive the highest possible level of subscription that the available network resources would allow.

6.3.2 Receiver 2 (5 Mbs Link Bandwidth). The throughput of receiver 2 is shown in Figure 9 and 10.

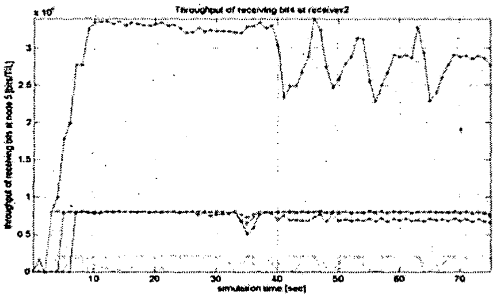


Figure 9: Throughput of receiving bits at receiver 2 without PDM.

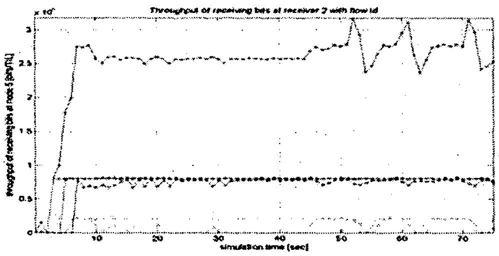


Figure 10: Throughput of receiving bits at receiver 2 with PDM.

Discussion:

As shown in figures 9 and 10, SPLIT converged to a stable level of subscription when compared to SPLIT without PDM. It can also be seen from the figure 10 that the subscribed bandwidth in SPLIT with PDM is less than SPLIT without PDM, where it is 2.5 Mb and 3.2Mb respectively.

SPLIT without PDM takes longer time to converge to a stable level of subscription than its counterpart. However in this instance join experiment failures after 40secs not only hampers SPLIT performance, but also adversely affect throughput during this stage. In this experiment SPLIT protocol with and without PDM were unable to utilize the available bandwidth. After 50 sec of simulation time SPLIT with PDM is trying to subscribe to higher layers to utilize the available bandwidth due to frequent join experiments failure, which resulted in the packet loss leading to performance degradation.

6.3.3 Receiver 5 (2.5 Mbs Link Bandwidth).

The throughput of receiver 5 is shown in Figure 15 and 16.

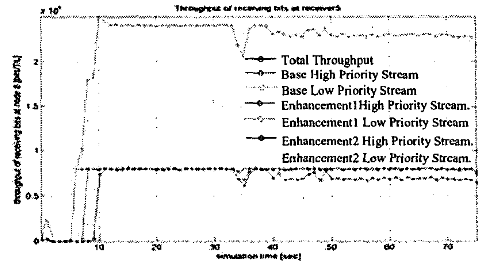


Figure 15: throughput of receiving bits at the receiver 5 without PDM.

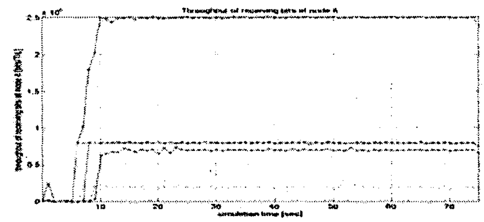


Figure 16: throughput of Receiving Bits at the receiver 5 with PDM.

Discussion:

As shown in figures 15 and 16 the SPLIT receiver was able to subscribe to both the high and low priority streams of the base layer as well as the high priority streams of the first two enhancement layers. This resulted in the observed increase in throughput at the SPLIT receiver.

In Figures 15 and 16 it can also be seen that, SPLIT receivers with PDM and without PDM, were able to converge to a stable state at a similar rate. But whenever congestion occurs SPLIT receiver is able to drop its low priority layers to avoid performance degradation.

Overall in this experiment SPLIT with PDM was able to use the available bandwidth much more efficiently than without PDM. By not subscribing to the low priority streams of the enhancement layers the SPLIT receiver is shown to be able to improve the preserved video quality to the end users.

6.4 Packet Drops with Varying Queue Length

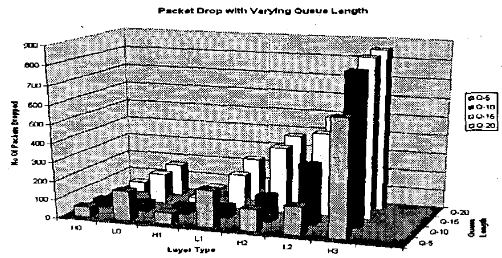


Figure 19: Packet drop in All Layers with Different Queue Lengths.

Simulation results shown in Figure 19 are obtained with different queue lengths keeping rest of the parameters untouched. In figure 19, layers numbers *hi* shows all high priority layers and *Li* depicts all low priority layers. From this graph we can say that PDM mechanism makes all lost packets concentrated in low priority layers. Figure 19 shows that with varying queue length packet loss in the Base high layer less when compared to the remaining enhancement high layers. As seen in figure 19 all packet drops are isolated from high priority layers.

7. Conclusions and Future Work

In this paper, we apply priority dropping mechanism (PDM) to RED algorithm to protect the video layers selectively from performance and throughput deterioration. Moreover, to achieve fairness among different type of network traffic with PDM, we implemented packet classification scheme (PCS) to assign the priority to each SPLIT layer or packet based upon the target rate.

Another interesting aspect of our research is regarding loss of packets in the individual layers. It was observed that, in the proposed system, video layers of high priority do not experience much loss due to the proposed PDM mechanism in the RED queue. Moreover, a new packet classification scheme implemented in SPLIT makes all lost packets concentrated in video layers of lower priority. Hence SPLIT based applications will have better quality in the events of congestion because most of the packets, which violate the available rate constraint, are dropped in the lower priority layers containing low priority traffic.

In the future we would also like to examine the effects of using different scalable video encoding techniques and packetization schemes to find out which are best suited for use with the SPLIT protocol. We also like to run the simulations on more complex network topologies involving multiple routers and studying the effect of varying the congestion link delay to the response time of the congestion control mechanism.

8. References

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