PERFORMANCE ANALYSIS OF A SPLIT-LAYER MULTICAST MECHANISM WITH H.26L VIDEO CODING SCHEME

Naveen K Chilamkurti

Ben Soh

Sri Vijaya Gutala
Applied Computing Research Institute
La Trobe University, Melbourne
Victoria, Australia-3086

Abstract Support for video transmission is rapidly becoming a common requirement. Video coding schemes such as H.26L are combined with multilayer multicast protocols such as SPLIT to improve the quality of video received at the receiver. In this paper, we built a simulation system using a modified version of JVT (Joint Video Team) encoding / decoding software package and Network Simulator NS-2, to evaluate H.26L video transmission over SPLIT. System performance was observed in terms of Loss Ratio, Video Jitter, throughput and PSNR for quality of the transmitted video.

1 INTRODUCTION

In an era of proliferating multimedia applications, support for video transmission is rapidly becoming a basic requirement of network architectures. Furthermore, since most video applications (e.g., teleconferencing, television broadcast, and video surveillance) are inherently multicast in nature, support for point-to-point video communication is not sufficient. Unfortunately, multicast video transport is severely complicated by variation in the amount of bandwidth available throughout the network (Xue, 1999).
A scalable solution to the problem of available bandwidth variation is to use multi-layered video. A multi-layered video encoder encodes raw video data into one or more streams, or layers, of differing priority. However, multi-layered video is not by itself sufficient to provide ideal network bandwidth utilization or video quality (Mccanne, 1996).
The SPLIT-Layer Video Multicast Protocol is a receiver based rate adaptation scheme solely intended for single source video transmission (Chilamkurti, 2003). 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.

In this paper, we simulate the transmission of H.26L encoded streams over SPLIT. H.26L is the newest and most efficient video coding scheme developed by the International Telecommunication Union (ITU) (H.26L, 2003). It uses a number of tools that allow it to deliver much more efficient video coding in low bit-rate applications than any MPEG standard.

2 H.26L – A VIDEO COMPRESSION STANDARD

2.1 H.26L: Overview

The main objective behind the H.26L project is to develop a high-performance video-coding standard by adopting an approach where simple and straightforward design using well-known building blocks are used. The ITU-T Video Coding Experts Group (VCEG) has initiated the work on the standard in 1997. The emerging H.26L standard has a number of features that distinguish it from existing standards, while at the same time, sharing common features with other existing standards (Greenbaun, 1999).
Some of the key features of H.26L are (1) Saves up to 50% in bit rate savings (2) High quality video (3)

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Adaptation to delay constraints (4) Error Resilience and (5) Network friendliness.

3 SPLIT-A RECEIVER-ORIENTED VIDEO MULTICAST PROTOCOL

3.1 Overview

There are many receiver based rate adaptation protocols capable of providing scalable rate adaptation of multicast video traffic to heterogeneous receivers. SPLIT-Layer Video Multicast Protocol (SPLIT) however is specifically designed to take advantage of existing encoding techniques to provide the end user with an increased perceived video quality.

SPLIT works by having the source S encode n (n > 1) video layers (V) where V1 is the base layer and every additional layer V2,..,Vn is enhancement layers. Each layer is then 'split' into two streams VnHP and VnLP where VnLP contains aprox. 1/n-1 of Vn. VnHP and VnLP are then transmitted to separate multicast address at a high and low priority respectively (IPv6 Priority field). Each destination wishing to join a video session will begin by subscribing to the base layer (V1HP and V1LP) after t time intervals if there is no congestion the receiver will add V2HP and V2LP and again wait t time intervals 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.If destination D has subscribed to m video layers (that is V1HP and V1LP to VmHP and VmLP) detects congestion (determined by packet loss rate) and has not recently received a join experiment message the destination will drop VmLP and begin using the hybrid loss concealment to estimate the data lost from VmLP if packet loss rate is still to high the layer containing Vm-1LP will be dropped and estimated and so on. If after dropping layer V1LP congestion remains a problem layer VmHP is dropped, layer VmLP and Vm-1LP remain dropped and layers V1LP to Vm-2LP are reinstated. The sequence is repeated with m now equal to m-1 until acceptable packet loss is obtained or only layer V1HP remains.

If receiver R1 is currently subscribed to layer n and receiver R2 (who shares a bottleneck point with R1) is subscribed to layer n + 1 and both suffer congestion from the same cause (i.e. at the shared bottleneck) and receiver R1 drops layer n this will have no effect on the congestion unless R2 drops layers n + 1 and n. Therefore the acceptable length of time for a receiver to be congested (i.e. wait to drop a layer) will be a function of the number of layers the receiver is currently subscribed to. After dropping a layer a receiver will send a drop layer message stating the layer that has been dropped so 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 one the receiver is currently subscribed to. After a successful layer drop (i.e. no more congestion) the receiver will send an end congestion message to surrounding receivers so that any lower layer 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 a period can drop the appropriate layers.

Before beginning a join experiment a receiver sends a join experiment notification message addressed to the base layer in the local area (determined by IP multicast scope) with IPv6 priority set to 7 (internet control traffic) (Hinden, 1995). If the experiment fails (causes congestion) the layer is dropped and any other receivers in the area that were affected by the congestion do not drop layers until some time after the experiment. If congestion remains this may be improved by a layer dropped message. If the experiment is a success a join success message is sent so that any receiver who suffered congestion shortly after receiving a join message can begin to drop layers as the congestion was caused by an external event. (i.e. not the join experiment)

4 EXPERIMENTATION SETUP

4.1 Overview

In order to evaluate the video transmission over SPLIT, we set up the simulation system, which is illustrated in Figure 1 to gather data for our analysis. The data and process point of views of the simulation set-up is demonstrated in the Figure 2.

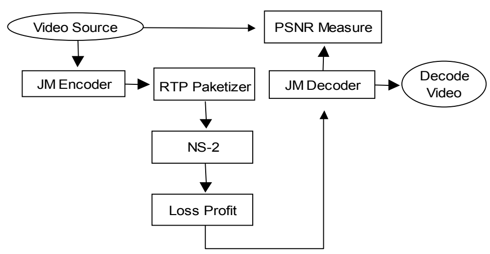

Figure 1: Simulation system block Diagram

We first feed a sample video source (Foreman. CIF), a raw video sequence in QCIF (Quarter Common Image Format) [4:2:0] format to the encoder.

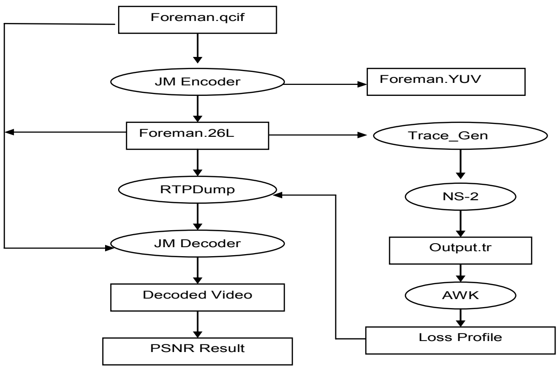

Figure 2: Data and Control Flow of the Simulation System

5 SIMULATING SPLIT

SPLIT is simulated using (NS-2, 1999) and all the necessary information to configure and control the simulation is stored in a file using Otcl script. The simulation objects are instantiated with the script, and immediately mirrored in the compiled hierarchy. The input script defines the topology, builds the agents, sets the trace files and sets the start times for the initial events in the simulation.

5.1 Topology

The following network topology was defined to run the simulations.

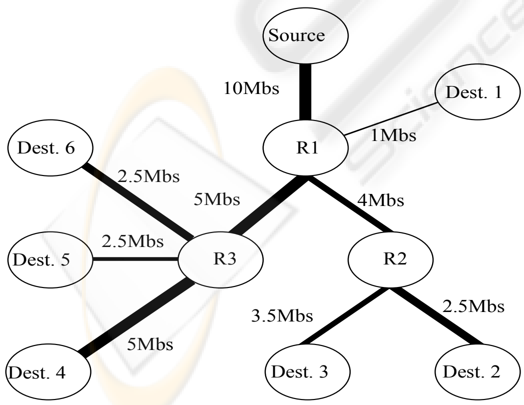

Figure 3: Topology of the Simulation

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. We generate a video trace and attach it to the source. At t= 0.01 Sec source transmits the CBR data stream to all the receivers. At t=1 Seconds both TCP and trace data applications begin to transmit frames with the interval of 10ms.

5.2 Routing and Queuing

A dense mode of multicast routing algorithm is used during simulations with the prune time-out set to 30 seconds. This was to ensure that the operation of SPLIT could be fully examined without any interference from the underlying routing protocol. Each router was implemented using the RED (Random Early Detection) (Floyd, 1993) queue so that optimal transmission rates of the base layers could be achieved through RED queue and its priority stream.

6 EXPERIMENTATION RESULTS AND ANALYSIS

To evaluate the visual quality of the video, the simulations are run under two different scenarios. The first scenario (UDP-H26L) provides the baseline for comparing our results of visual quality. The application simply hands down each video frame to the UDP layer. In the second scenario, the frames are handed to the SPLIT Source (SPLIT-H26L).

6.1 Throughput at Receivers:

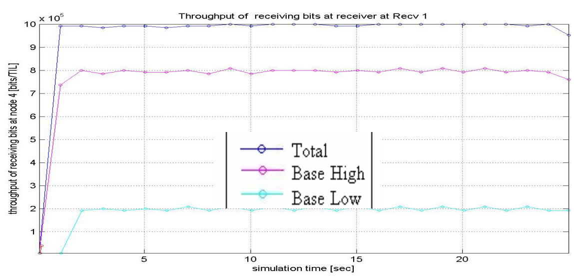

Figure 4: Throughput of Receiving bits at Receiver1

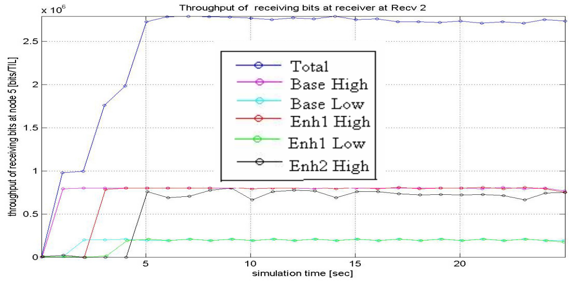

Figure 5: Throughput of Receiving bits at Receiver2

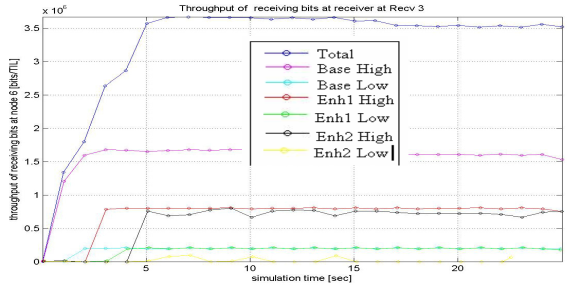

Figure 6: Throughput of Receiving bits at Receiver3

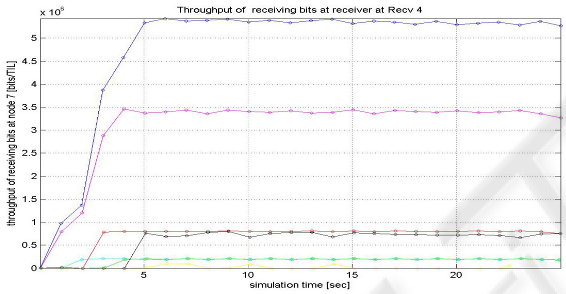

Figure 7: Throughput of Receiving bits at Receiver4

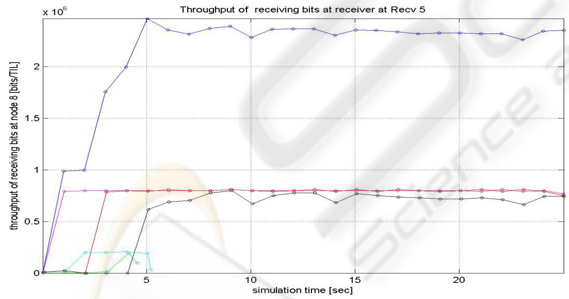

Figure 8: Throughput of Receiving bits at Receiver5

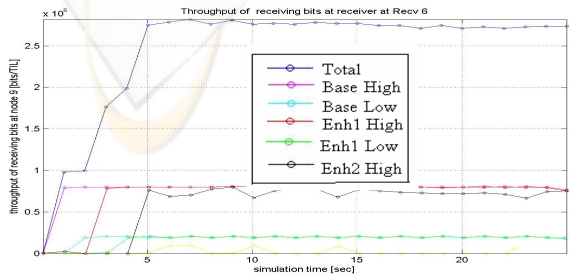

Figure 9: Throughput of Receiving bits at Receiver6

Discussion:
Throughput is defined as the maximum rate at which the switch can forward packets without packet loss. Figures 4, 5, 6, 7, 8, and 9 represent throughput at the receivers.

In this experiment Receiver 1 is connected to the source at a bottleneck speed of 1Mbs.The available bandwidth was fully utilized by SPLIT protocol and Receiver1 being able to subscribe only the base layer. This occurred because the 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.

The second set of results was taken from Receiver 2, which is connected at a bottleneck speed of 2.5Mbs to the source. In this experiment SPLIT was able to fully and effectively utilise the available bandwidth and was able to subscribe to the available video layers.

The results shown in Figure 6 are taken from Receiver 3, which is connected to the source at a bottleneck speed of 3.5Mbs. In this instance the SPLIT receiver was able to subscribe to three high priority streams as well as the three low priority streams. By sacrificing portions of each enhancement layer with the view that packet loss concealment mechanisms would be able to reproduce the missing data SPLIT was able to receive an extra enhancement layer.

Receiver 4 is connected to the source at a speed of 5Mbs, which is clearly sufficient to receive the five 1Mbs video layers being transmitted by the source. As expected, Figure 7 shows that both the SPLIT receivers had no problems in receiving the five layers.

Receiver 5 was connected to the source at a bottleneck speed of 2.5Mbs. As shown in Figure 8 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.

In this final experiment, Receiver 6 was connected to the source at a speed of 2.5Mbps. As shown in Figure 9 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.
The overall performance of SPLIT was quite good. It was able to subscribe to all the available layers and this leads to the increase in throughput at the receiver. By being able to make a more effective use

of the available bandwidth the SPLIT mechanism ensures better video quality.

6.2 Loss Ratio

Loss ratio for a particular flow I is defined to be:

$$\text{Loss Ratio} = \frac{\text{Number of packets dropped in flow I}}{\text{Total packets received in flow}}$$

In this experiment, there were five layers of high priority and low priority stream. The loss ratio is computed for each flow from the trace file obtained after the simulation and tabulated as follows:

Table 1: Loss Ratio in High Priority Layers

	Number of packets received	Number of packets dropped	Loss Ratio
Base High Layer	67419	10	0.014832
Enhancement High Layer1	55072	18	0.032684
Enhancement High Layer2	50665	94	0.185532
Enhancement High Layer3	37981	243	0.639793

The number of packets dropped was very low compared to the number of packets received in all the high priority layers. The mean loss ratio across all the high priority layers subscribed is found to be 0.0021821025.

Loss Ratio in the lower priority layers:

Table 2: Loss Ratio in Low Priority Layers

	Number of packets received	Number of packets dropped	Loss Ratio
Base Low Layer	14379	31	0.215592
Enhancement Low Layer1	2625	124	4.723
Enhancement Low Layer2	236	146	61.8644
Enhancement Low Layer3	207	154	74.3961

The mean loss ratio in low priority layers is found to be 0.240133 or 24%. But in SPLIT mechanism if a destination D has subscribed to m video layers (that is V1HP and V1LP to VmHP and VmLP) and detects congestion then the destination will drop VmLP. If packet loss rate is still too high then layer-containing Vm-1LP will be dropped and so on. Hence the loss rate in lower priority layers is found to be high.

6.3 Video Packet Jitter:

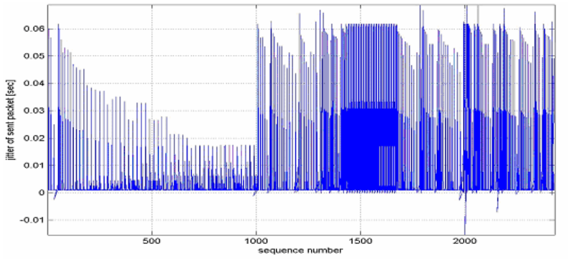

Figure 10: Packet Jitter for video flow – UDP-H26L

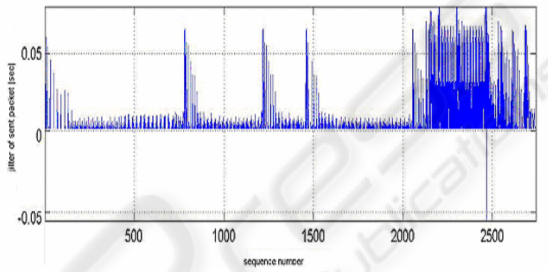

Figure 11: Packet Jitter for video flow – SPLIT-H26L

Discussion:

Figure 10 and Figure 11 depict the video packet jitters. The delay jitter values are taken for video over UDP and SPLIT at Dest4. Packet jitter experienced by UDP-H26L is more than SPLIT. While in Video over SPLIT the jitter values were ranging from 0.1 to 0.2. The reason for increase in jitter for the first scheme is that UDP bursty nature induces more jitter to the other competing flows. Thus video application based on SPLIT will require low play out buffers to absorb jitter than applications based on UDP. Hence we can say from the results obtained the SPLIT rate adaptation scheme helps in reducing jitter effects.

6.4 Quality Measure: PSNR (Peak Signal to Noise Ratio)

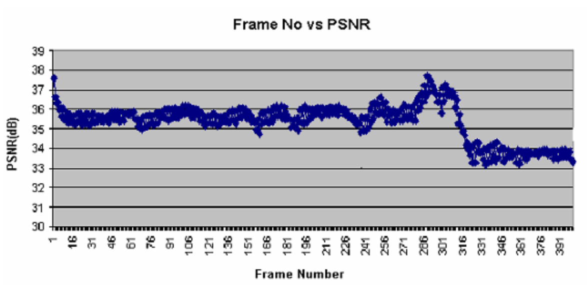

Figure 12: PSNR of UDP-H26L

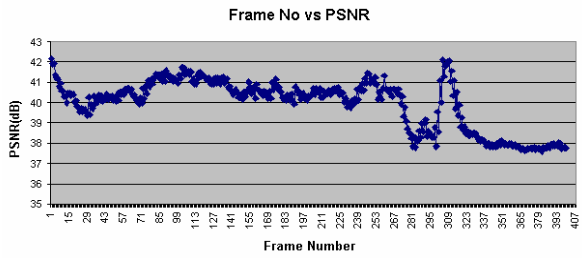

Figure 13: PSNR of SPLIT-H26L

Discussion:
PSNR values are compared by UDP-H26L and SPLIT-H26L schemes by streaming a QCIF foreman sequence encoded with H26L. Figures 12 and 13 compares the PSNR values of the foreman sequence encoded using the H26L codec with and without SPLIT protocol. Figure 12 corresponds to the simulations with UDP source, 5Mbps bottleneck bandwidth. In this case, for the foreman sequence, an average PSNR of 34.44 was obtained. The average PSNR obtained was 37.25. We can see from the graphs that under similar network conditions, the source with SPLIT gave better PSNR results. On average a gain of 3dB is found.

Figure 14: Foreman sequence used in simulations

Besides the subjective performance of the two schemes, snapshots of the decoded Foreman sequence are shown in Figure 14. The visual quality of the video transmitted with SPLIT is found to be better.

7 CONCLUSIONS AND FURTHER RESEARCH

In this paper, we built a simulation system using Network Simulator NS-2, to evaluate H.26L video transmission over SPLIT. System performance was observed in terms of Loss Ratio, Video Jitter, throughput and PSNR for quality of the transmitted video.

It was observed that, in the proposed system, video layers of high priority do not experience much loss, because the SPLIT mechanism makes all lost packets concentrated in video layers of lower priority.

To evaluate the visual quality of the video sequence, two scenarios were considered. In the first case, video traffic was transferred from source to the destination using UDP. The simulation results showed a low jitter value and a high quality image at the decoder under congestion by being able to subscribe extra enhancement video layers, whilst keeping packet loss under a threshold.

Our future research is to establish a mapping between the packet level loss pattern and loss pattern on a video level. This is of utmost importance since it will enable relating the end-user perception to the packet level loss, which might provide a reference basis for effective error correction or error concealment techniques at the end hosts.

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