MULTI-LAYERED VIDEO TRANSMISSION OVER INTERNET

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Abstract

Transmission of real-time video in heterogeneous environment poses a number of problems related to the scalability. Among the various solutions that have been proposed layered transmission of video seems quite promising. In this paper architecture for multi-layered transmission of video over Internet has been proposed. Work done for the implementation of the architecture has also been stated.

Introduction

The multicast distribution of real-time video is an emerging field. It has become more important since the development of video conferencing tools, distance education and media-on-demand. Running of this application on the current network becomes a challenging task. This is because, firstly, the real time applications in general are less tolerant to the packet delays and losses. Secondly, in the heterogeneous network different receivers may have different bandwidth capacities as a result of which some receivers may have congested link, which may pose problems.

The classical solution that has been forwarded is the Receiver-driven Layered Multicast (RLM) approach [1]. In this approach the video is transmitted in a number of layers and at the receiving end the receivers accept as many layers as their respective bandwidth allows. The quality of image formed is thus dependent upon the number of layers combined. Thus, while the sender sends the video at a constant rate and in layers, the receivers dynamically judge the number of layers they can receive and combine the layers. While RLM has been hailed as a milestone in improving the scalability of video transmission it suffers from certain disadvantages. These include additional encoding and decoding time, need for a bigger buffer at receiving end, congestion of link during join experiment and misunderstandings during the sharing of information between receivers.

Some of the problems faced by RLM have been solved in the Layered Video Multicast with Retransmission (LVMR) protocol [2]. The main features of this approach are (a) retransmission of lost packets given an upper bound on recovery time; (b) application of an adaptive playback point scheme to achieve more successful retransmission; and (c) adaptation to network congestion and heterogeneity using hierarchical rate control mechanism. In the hierarchical approach of LVMR the information is distributed between the sender, receiver, and the some agents in the network. This helps the receivers have minimal state information,

decrease control traffic on the multicast session, allow multiple experiments to be conducted simultaneously, and helps to drop the correct layer(s) during congestion in most cases.
The earlier mentioned approaches are receiver initiated wherein the sender sends fixed number of layers and the receiver adjusts dynamically. This approach alone is unable to provide scalability to a large extent because the key challenge of sending the video over the Internet is matching the transmission rate to the currently available bandwidth. In the sender-initiated approach the selection of number of video layers to be sent is dependent on the feedback from the receivers. To prevent feedback implosion the concept of feedback merger is used [3,4]. The adaptive multi-layered scheme uses the rate-based approach in which the sender sends a feedback packet with other video packets. This feedback packet goes through the intermediate nodes to the destination and returns back to the sender, thereby, forming a closed loop. The intermediate nodes and destination mark the feedback packet with the available bandwidth information and send it to the sender. Based on the information from the network and the receivers the sender can adjust the total number and rate at which to send the layers.

Both, the sender and the receiver-initiated approaches, work on the philosophy of best effort , wherein the delivery of packets is not guaranteed. Hence, due to congestion some packets may get dropped. This drop in packets is normally uniform which means that packets may get dropped from any layer. Since, base layer normally contains the more sensitive information it may not be a good idea to let the packet from the base layer drop to adjust to the available bandwidth. This can be achieved by using the priority-based approach wherein the priority is given to a layer depending upon its utility [5].

In another approach, to compensate for the lost packets some extra packets are sent along with the normal packets. Whenever there is a packet loss at the receiving end, the receiver uses these extra packets to reconstruct the lost data packets. Even though this technique uses more bandwidth it is considered to be a good solution for the recovery of lost data packet. An example of error correcting code is the Reed Solomon code. During the layered transmission of the video, one of the layers could be the Forward Error Correcting (FEC) code layer. In case there is a drop in packets the receiver may be able to regenerate the lost packet by taking data from the FEC layer.

Proposed Architecture

Figure 1 shows the proposed architecture for the transmission of video in multicast environment. The transmission shall be done in a number of layers. The first layer, the Base Layer, containing only the basic information of the image shall be given the highest priority. The second layer shall be the Enhancement Layer. The third layer shall contain extra packets for forward error correcting code (FEC) or additional enhancement layer. Feed back from the receivers shall be sent through successive Feedback Mergers.

The aim of the project is to develop a combined sender and receiver driven approach. In this system both sender and receiver shall actively take part in the adjustment of video quality. Based on the network condition the sender shall decide upon the optimum number of layers to be sent and based on the local network conditions the receivers shall decide upon the number of layers to accept.

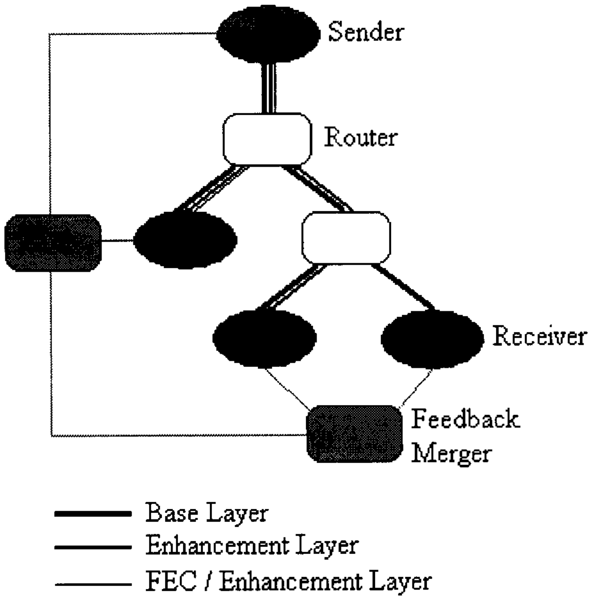

Figure 1: Architecture of the Proposed system for the improvement of quality of video

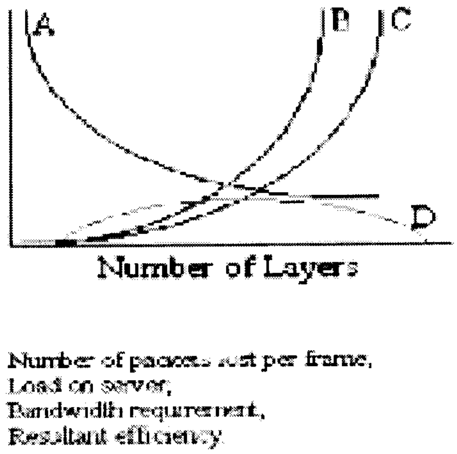

Figure 2: Effect of the three parameters on the optimum number of layers.

The optimum number of layers to be sent by the sender depends upon:

- Load on Server: The number of layers increases the load increases. This is because the server has to do more processing to send extra information with every layer.
- Load on the network: As the number of layers increases the load on the network increases because the layers have to be sent at different multicast addresses.
- Quality of video: As the number of layers increases the options for the receiver to increase the quality of video increases. With the increase in the number of layers, the receiver can get the finer details of the image at the macroblock level.

The sender should be intelligent enough to send the optimum number of layers. For that the sender has to have the information about the network conditions. One way could be that the receiver sends the feedback to the sender regarding the packet drop, rate at which packets are reaching at the receiver, etc. As a result of it the sender will not be sending unnecessarily extra layer. Hence some mathematical modelling has to be done in order to find out the optimum number of layers, which will suit the requirements of maximum number of receivers. Figure 2 shows the effect of the three parameters on the optimum number of layers.

Results and Discussion

Transmission of Video in Layers

As the first step towards the creation of the architecture the video being sent has been divided into two layers. Marratech Pro [11] (a video conferencing tool) has been used for sending and receiving video over the Internet. The division of video into layers is based on the macroblock address. Half of the macroblocks are sent to one multicast address and the other half to the other multicast address. The image was sent in the packet form by using the RTP (Real Time Transport Protocol), and was sent in a best effort way. The receiver can take one or both the

Figure 3: Receiver receiving to the base layer

Figure 4: Receiver receiving enhancement and base layer

layers. Figures 3 and 4 show the effect of receiving only one and of receiving both the layers. As can be seen there is marked difference between the quality of the two images.

The work is going on to divide the video in layers at the codec level. Video would be divided in layers at macroblock level in some context dependent way. For instance, in the videoconferencing scenario the portion of the image that contains the head and shoulder of a person could be sent to the base layer and the rest of information could be sent to the refinement layers. Here the concept of conditional replenishment may also be applied to monitor each macroblock so that the macroblocks, which have changed most, can be assigned to the base layer and others to the refinement layers.

Forward Error Correcting Code to Compensate for Packet Loss

Reed Solomon code [10] is a standard code for reconstruction of lost packets. Its algorithm is widely available in the standard textbooks. This code has been encoded in Java language. Currently it stands alone but in later stage it shall be sent as the third layer.

Reservation of bandwidth

In the real network there are constant variations in the availability of the bandwidth. When the video is divided into two layers according to the current system the available bandwidth gets divided into two equal parts (Figure 5).

At the next stage, a part of the total available bandwidth shall be reserved for the base layer. The base layer shall contain the essential information and the enhancement layer shall get the rest (Figure 6).

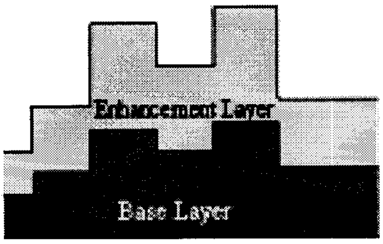

Figure 5: Base layer and enhancement layer getting equal share of the available bandwidth.

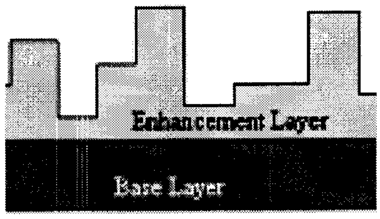

Figure 6: Constant reserved base layer variable enhancement layer

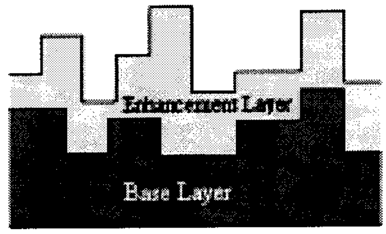

Figure 7: Variable reserved base layer with variable enhancement layer.

Since the demand for the essential part itself may keep on varying, some method shall be found to decide on essential part of the video. The reserved base layer shall get varying bandwidth based on requirement and the rest of the available bandwidth shall be given to the enhancement layer (Figure 7).

Simulation studies may also be carried out to see the performance of the three methods.

Conclusion

Video has been split into two - base and enhancement - layers. The two layers have been sent to different multicast addresses. The receiver has the option of taking one or both the layers. The work is going on the division of layers at the codec level. FEC based on the Reed Solomon code has been written.

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