EE368C: project proposal
Transmission of layered video using priority dropping

Group #6
Athina Markopoulou, Sangeun Han
{amarko,sehan} @ stanford.edu

ABSTRACT
There is an increasing interest in the deployment of high quality video over the Internet, for applications such as Video on Demand or Video Conferencing. A significant amount of work has been done on scalable video coding to deal with the heterogeneous conditions in the Internet. However, scalable coding is only part of the problem and it should be appropriately used at transmission time. One of the router mechanisms that could work together with the scalable encoding, is priority dropping, which is provided by the AF class of the DiffServ standards. In this paper, we want to show the benefit of using priority dropping for the transmission of layered video and to provide some guidelines on how to configure the parameters, of both layering and dropping, in various network scenarios.

1. BACKGROUND
1.1 Scalable video encoding
The scalable video coder produces a bit stream, decodable in multiple layers, which provides different levels of quality. It allows computation-time and memory-limited decoding on less powerful hardware platforms, and it can substantially improve the quality of video transmitted over error-prone channels such as the Internet or wireless.

Temporal, SNR, and spatial scalability have been defined in the international video coding standards, e.g., the MPEG-2 [10] and the H.263 [9]. All of these types of scalable video consist of a base layer (BL) and one or more enhancement layers (ELs) as shown in Figure 1. The BL of the scalable video stream represents, in general, the minimum amount of data needed for decoding that stream. The EL represents additional information, and therefore it enhances the video signal representation when decoded by the receiver.

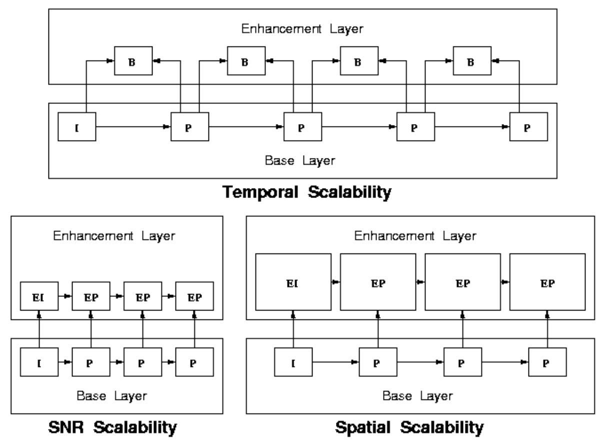

Figure 1. Basic Video Scalability Structures

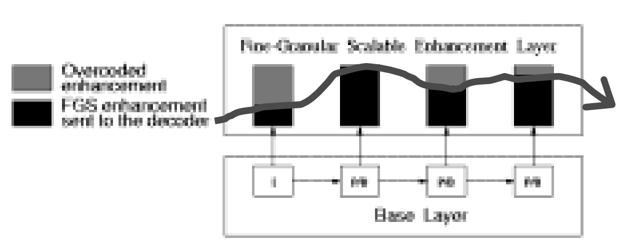

Figure 2. Video Scalability Structure with Fine-Granularity

It is virtually impossible to achieve a good coding efficiency/video-quality tradeoff over such a wide range of rates. To achieve fine granularity, van der Schaar et al. [2] proposed a scalable Internet video based on MPEG-4 scalable video coding method using both a prediction-based base layer and a fine-granular enhancement layer shown in Figure 2. One of the advantages of the fine-granular scalability approach is that the enhancement layer sub-streams can be combined at the receiver into a single stream and decoded using a single EL decoder.

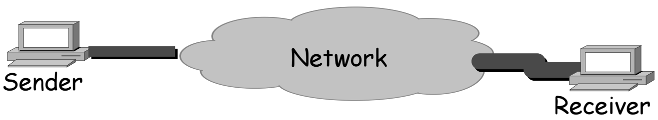

Figure 3. Transmission over the Internet

1.2 Transmission of video

At transmission time, the available bandwidth may be less than the total rate of the encoded stream. Therefore, intelligent decisions have to be made, by the server or the network, in order to use the available bandwidth to deliver the most important parts of the stream and maximize the quality, perceived by the receivers. Many approaches have been proposed so far:

- Feedback on the available bandwidth allows ***the server to adjust the transmission rate*** to adapt to the bandwidth variations. This approach is limited by the feedback delay; by the time feedback is reported back to the server, conditions may have already changed. It is also inappropriate for a single server multicasting to many receivers, because the poorest receiver limits the performance of the whole session.
- To remedy this last problem, McCanne et.al in [4] proposed Receiver Driven Layered Multicast (***RLM***). The server multicasts the layers on different multicast groups and each receiver subscribe to the appropriate layers, matching his capabilities.
- Another approach is to try to completely prevent congestion by using admission control, reservations and smoothing of video streams a la ***IntServ***. The well-known problems of this approach are high complexity and limited scalability.
- The opposite approach is to accept that loss is inevitable over the heterogeneous public Internet and try to limit its effect on the perceived quality, by dealing with it intelligently. In periods of congestion, the network should protect the most important data, by means of some preferential treatment, such as ***unequal error protection*** or ***priority dropping***.

1.3 Differentiated Services

The Differentiated treatment of packets according to their marking is exactly the key idea of DiffServ [6], the new Internet QoS architecture, currently under standardization by the IETF. An example of a Diffserv router is shown in Figure 4. Each packet bears a marking indicating the class it belongs to. The router looks at this marking and sends the packet to the corresponding queue.

The class of interest to us is called "Assured Forwarding" or AF, defines in RFC 2597 [7], and it provides minimum bandwidth guarantees as well as up to 3 dropping priorities as shown in Figure 5.

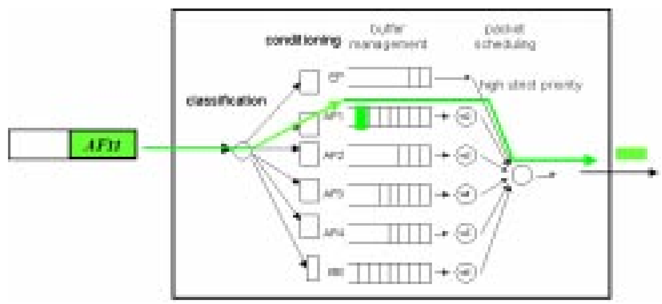

Figure 4. Example of a DiffServ node

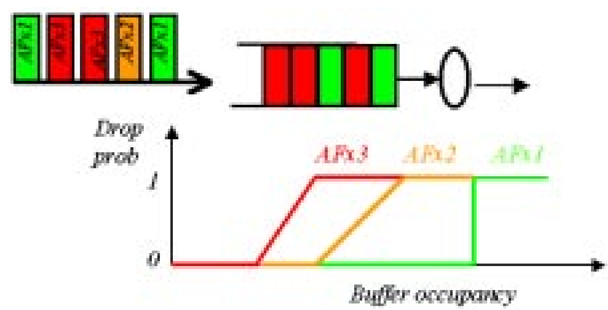

Figure 5. AF class

2. PROPOSAL

Our goal in this project is to quantify the benefit from combining layering capabilities and priority dropping. Intuitively, marking base layers with low drop precedence should lead to graceful quality degradation in times of congestion. Furthermore, we would like to come up with specific recommendations on how to do the layering and how to configure the routers implementing AF, in order to maximize the perceived quality over various realistic network scenarios.

We plan to use the Network Simulator (NS) as shown in Figure 6. We will use the layering capabilities of

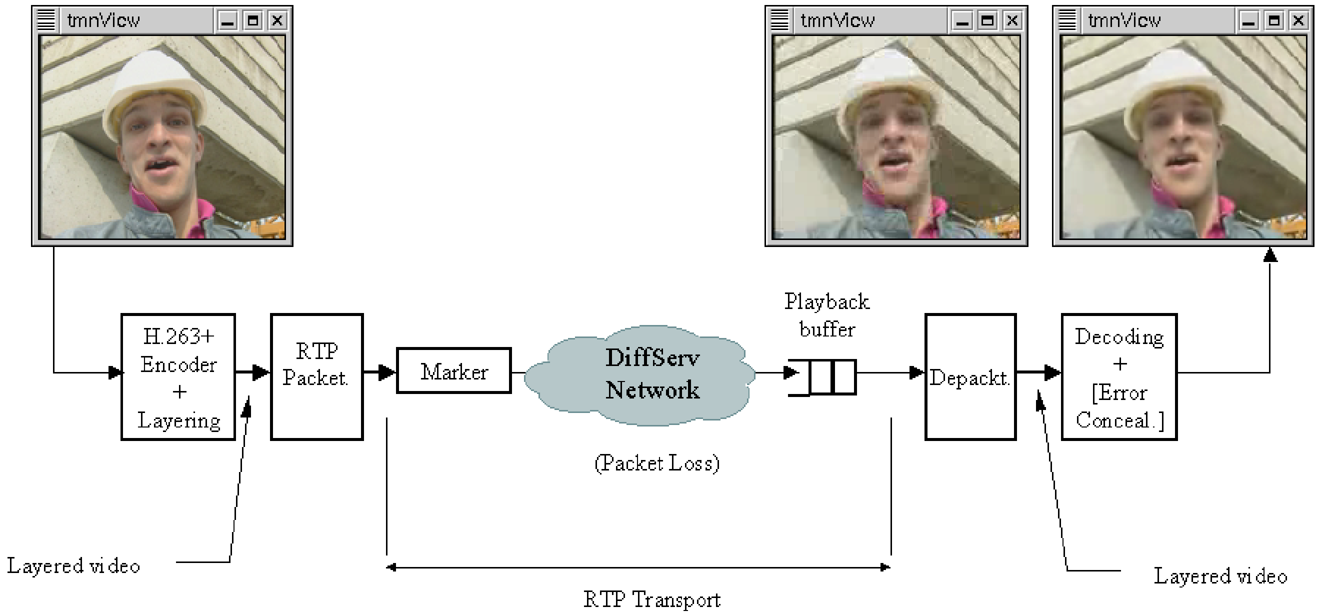

Figure 6. Simulation Block Diagram

the H.263 encoder. We will use RTP [11] to packetize. We will mark different layers to different drop priorities.

There are several issues to be explored in the above scenario, from both a layering and a network point of view. Here are some of the scenarios we would like to test and some of the results we expect to get:

- Given a rate-distortion curve, what is the quality degradation, for different loss patterns experienced by the enhancement layers. How does this improve by using priority dropping?
- How to optimize the H.263 layering given the loss patterns experienced by the enhancement(s) layers. Eg: (i) loss due to multiplexing of video streams over a single buffer or (ii) over multiple hops (iii) loss due to putting video and data in the same queue (iv) due over a wireless LAN
- Show the tradeoff between motion smoothness and quality of picture under bursty loss scenarios. PSNR vs intra rate.
- Show the benefit of using priority dropping in the scenario of multiplexed layered streams at a single hop. Quantify this benefit in terms of: (i) increased number of multiplexed streams at the same quality (ii) increased quality experienced by a single stream.
- How many layers and priorities are needed? Show that the benefit from having finer granularity decreases with the number of layers/priorities.
- How to optimally (in the rate-distortion curve) layer a given sequence to a fixed number of layers, having in mind that the bandwidth available at the last hop (dial-up modem, DSL/cable modem, corporate LAN etc). This is similar to what is as done by RealVideo, to encode their SureStream at different rates, while we would like to create different layers.
- Wireless LANs (802.11) have 4 discrete rates. Explore how layers would be used in this case.
- How to mark different types of streams (in terms of rates and layers). Eg. A high rate stream A has a BL-A and an EL-A. A low rate stream B has a BL-B and an EL-B. How should the BL-A, EL-A, BL-B, EL-B, should be marked and therefore assigned to the high and low AF drop priorities ?
- How to configure the AF buffer management parameters?
- Should Video and TCP data be put in the same or different queues?

Table 1. Schedule

Date	Person	Task
Week 02/05	Athina	Implement AF with 3 levels
02/12	Athina	Implement AF with n levels
02/12	Athina	Implement playback buffer
Week 02/05	Sangeun	RTP packetization
Week 02/05	Sangeun	H.263 encoding providing 2 layers
Week 02/12	Sangeun	H.263 encoding with n layers
Finalize week 2/12	both	Think of scenarios
Start week 2/12	both	Run simulation for the scenarios
Week 02/26	both	Prepare paper+presentation

3. REFERENCES

[1] J.Kimura, F.Tobagi et. al, "Perceived quality and bandwidth characterization of layered MPEG-2 video encoding", SPIE'99.

[2] M.Van Der Schaar & H.Radha, "A hybrid temporal-SNR for the Internet video", to appear in IEEE Trans. on Circuits and systems for Video Technology.

[3] U.Horn, B.Girod, "Scalable Video Transmission for the Internet", Computer Networks and ISDN Systems 1997.

[4] S.McCanne, N.Vetterli, V.Jacobson, "Low complexity video coding for receiver-driven layered multicast", JSAC 1997.

[5] IETF DIFFSERV Working Group, http://www.ietf.org/html-charters/diffserv-charter.html

[6] RFC 2475, "An Architecture for Differentiated Services", Dec. 98.

[7] RFC 2597, "Assured Forwarding PHB Group", June 1999.

[8] Network simulator, http://www.isi.edu/nsnam/ns

[9] ITU-T Recommendation H.263 (Annex O).

[10] MPEG-2, ISO/IEC recommendations, Oct.'00.

[11] RFC 1889, "RTP: a transport protocol for real time applications", Jan.1996.