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Department of Computer Science & Engineering - Washington University in St. Louis
Campus Box 1045 - St. Louis, MO - 63130 - ph: (314) 935-6160.

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Department of Computer Science
Washington University
Campus Box 1045
One Brookings Drive
St. Louis MO 63130

# An Efficient Quality Scalable Motion-JPEG2000 Transmission Scheme

Ruibiao Qiu[1] and Wei Yu[2]

[1] Applied Research Laboratory, Department of Computer Science, Washington University, St. Louis, MO 63130
ruibiao@arl.wustl.edu
[2] Computer Communication Research Center Washington University, St. Louis, MO 63130
weiyu@ccrc.wustl.edu

**Abstract.** Video application over the Internet are getting increasingly popular because of the explosive growth of the Internet. However, video packets loss due to network congestions can degrade the video quality substantially. In this paper, we propose a transmission scheme for Motion-JPEG2000 video sequences over IP networks. Our scheme utilizes the progression modes in Motion-JPEG2000. It can be implemented in an active network environment efficiently. Our simulation shows that our scheme gracefully adapts to network congestion and improves the quality of video transmission in congested IP networks.

## 1 Introduction

Video communication over the Internet is becoming increasingly popular today. However, network congestion in the Internet can degrade the video quality substantially. When congestions happens, packets accumulate in the queues on the routers with overloaded downstream links. When sustained congestion occurs, the incoming packets fill up the queues on the router, making subsequent packets be dropped by the router. Such packet drops can happen to a random packet in a video stream depending on the traffic dynamics on the route between the video sources and sinks. Such random packet loss makes the decoders at the video sinks unable to reconstruct the original video because of the information loss in the dropped packets.

Because of the heterogeneity in the Internet and the majority TCP traffic behaviors, the congestion in the Internet is likely to exist. In order to reconstruct video quality for Internet video, various solutions have been proposed. Some solutions employ quality adaptation mechanisms in the video transmission. The idea is to make the video streams adapt to the available resources in transit from the source to the sink. When the resource is limited, the video data with the most contribution to the reconstructed video is transmitted, and the less important portion of the data is dropped to preserve resource consumption. In order for such an adaptive scheme to work properly, a video code stream that contains proper priority information of the segments in the code stream is preferred. In particular, if packets in a video streams can be prioritized according to their contribution to the image quality, we can restore the video sequences with reasonable quality despite the packet drops in the code stream.

As our previous work showed, Motion-JPEG2000, the moving picture coding standard of ISO's future still image coding standard, JPEG2000 [7], has advantages for video processing [19]. This is not only because of its high compression performance, strong error resilience, and good perceptual image quality, but also because it utilizes the wavelet transform [2] and embedded block coding with optimized truncation (EBCOT) [16] algorithm, which provide excellent signal-to-noise ratio (SNR) and spatial-frequency resolution scalability. This feature makes Motion-JPEG2000 a perfect coding scheme candidate for adaptive video transmission. A code stream of Motion-JPEG2000 is composed of image data packets, which are the units of collected image data arranged by certain progression orders. When the SNR scalability progression order is used, each data packet has different contribution to the whole reconstructed images. Data packets with different priorities can be extracted from the code stream.

With a quality scalable Motion-JPEG2000 code stream, we can develop adaptive video transmission systems by exploiting the SNR and resolution scalability in the code streams. We can implement the adaptation on the route between the sources and the sinks by selectively dropping the packets with lower priorities when network congestion happens. However, implementing the adaptation on the routers is more effective than on the video sources and sinks because packet drop caused by network congestion happens on the routers, and adaptation at the source or the sink is inadequate. It is impractical to make updates to all routers to just incorporate the video transmission scheme. However, based on active network, we can deploy the quality adaptive video transmission schemes naturally. Active networks enable data processing inside the networks. Active nodes, usually active routers and switches, can process packets of specific types. If we put some load of video processing on the active nodes, we can selectively drop packets to achieve quality adaptation when there is shortage of bandwidth resources.

In this paper, we describe an efficient quality adaptive transmission scheme for Motion-JPEG2000 video sequences over congested networks using active network approaches. We evaluate our scheme with simulation, and our results show that our scheme can provide video better quality scalability when network congestion happens.

Yeadon et al. [18] proposed to filter hierarchically encoded streams to ease network fluctuations. The difference between our work and Yeadon's is that Yeadon focused on the MPEG code streams, and the minimum dropping unit is a frame. In addition, because MPEG1 or MPEG2 code streams are not hierarchal, the flexibility is very limited, and it is difficult to decide the priorities of packets. In contrast, all wavelet-based code streams, such as JPEG2000 and MPEG4, have hierarchical structures. Because of this inherent characteristic, the mapping between the image data packets and the IP packets is natural. Our scheme also makes the active processing efficient because it takes constant time to process each packet without packet buffering and decoding.

WaveVideo [8, 5] system is an integrated, adaptive video coding architecture that can adapt video quality to network bandwidth resources. However, WaveVideo uses their own wavelet-based coding scheme instead of a standard coding scheme. Therefore, there is still improvement space in compression efficiency, and it can not be easily used with standard video. In addition, because the non-standard wavelet coding is used, its mapping of image quality to packet priority using a recursive luminance(Y) and color difference($C_bC_r$) channel tree is not a precise mapping scheme. The assumption of low subbands having high priority needs more strict mathematical proof. In addition, the WaveVideo codec has a fixed number of up to 50 priority levels equal to the number of the decomposition tree nodes. However, our scheme can generate variable number of priority levels making the fine tuning of the video quality with small granularity possible. Besides of their use of the 3-tuple of ( color channel, recursion depth, subband ) to determine the priority of an image data packet, the video transmission system must be tightly coupled with the codec, whereas in our scheme the video transmission system is loosely coupled with the Motion-JPEG2000 codec. All the information we need can be extracted from the generated code streams from the standard codec. Furthermore, because the packets have different sizes in the WaveVideo code stream, the rate adaptation mechanism is based on a *trial-and-fail* scheme. This can not guarantee to converge to the target rate. In contrast, in our scheme, because Motion-JPEG2000 can generate equal size packets for all quality layers, we can confidently adjust the target rate by dropping proportional number of packets. Additionally, the WaveVideo plugin requires a table lookup to make decision for each packet, whereas a simple value comparison is needed for our scheme. This difference of a memory access and a comparison instruction makes an active plugin with our scheme more efficient than WaveVideo.

Feedback-based Internet video transmission schemes have also been proposed. Good examples of such schemes are presented by Rejaie, Handley, and Estrin [12], Puri et. al. [11], Kim et. al. [9]. However, such schemes rely on information sent back from the receivers to adjust the bandwidth. This increases the response time to network congestion, and require more resources.

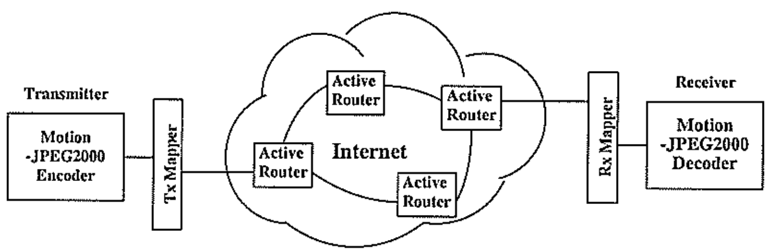
In addition to the EBCOT coding used in JPEG2000, other wavelet coding schemes, such as SPIHT [13] and EZW [14], have also been used in video transmission, such as [15] and [17]. Video processing is a good application for active networks. Hammi and Chen [6] advocate the use of active agents on boundary routers to perform user-specific services such as error resilient retransmission and filtering of video. Balakrishnan and Ramakrishnan [3] presented their work to adapt MPEG video streams to available on their active router prototype by dropping P and B frames. This scheme can not achieve accurate rate adaptation and is not efficient on the active router because of the decoding tasks.

Fixed network based schemes for Internet video transmission are also proposed. Systems with such schemes try to provide absolute guarantees on resource availability Therefore, packets drops can be effectively eliminated. Good example of such schemes are presented by Alwan et. al. [1], and Krishnamurthy and Little [10]. However, this scheme is hard to deploy because it requires a lot of router changes in the Internet. In addition, such schemes have no conflicts with our proposed scheme, as we can simply

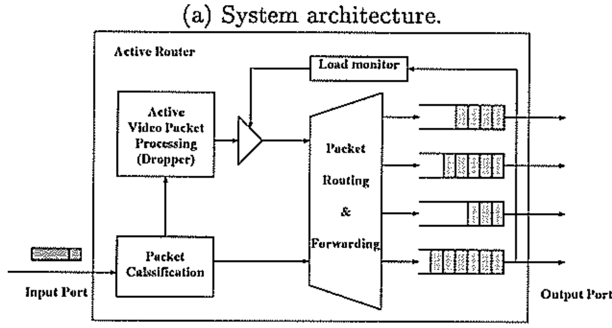
The rest of the paper is organized as follows: the proposed scheme is presented in Sect. 2. We evaluate our scheme through simulations in Sect. 3. Finally, Sect. 4 concludes the paper and outlines our future work.

## 2 Proposed Scheme

We propose a new scheme to deal with the network congestion in transmitting wavelet-based code streams. In particular, we have implemented the scheme with Motion-JPEG2000 video streams. In our proposed scheme, we introduce a mapping between the quality layer of the image data packets and the priorities in real network packets to preserve the importance of image data packets in the image code streams in network packets. In a system implementing our scheme(Fig. 1), there is a *Tx Mapper* between the video source encoder and the network, and a *Rx Mapper* between the network and the video sink decoder. At the *Tx Mapper*, image data packets are extracted from the code stream, and mapped to network packet with a small header. Correspondingly, at the *Rx Mapper*, headers are removed from the video packets, and restored into the original data stream with preserved priorities to the decoder. On the route between the video source and sink, quality adaptation is achieved by selective packet dropping with an active application on the routers in case of network congestion. Our proposed scheme is depicted in Fig. 1a, and Fig. 1b shows the operation of the quality adaptation active application on an active router.




(a) System architecture.




(b) Active router plugin operation.

**Fig. 1:** Architecture of our proposed scheme.

### 2.1 Mapping Model

Our mapping model maps the data packets in a Motion-JPEG2000 frame into network packets. This mapping scheme prioritizes the image data packets according to the progression order in use. We primarily focus on the progression by fidelity order. In this progression order, the image packets are first ordered by their quality layers, and then by resolutions, components, and precincts. Thereby, each image data packet corresponds to a 4-tuple identifier, namely, $(l, r, c, p)$, where $l$, $r$, $c$ and $p$ specify the quality layer, resolution level, component, and precinct, respectively. The quality level indicates the contribution of a data packet to the whole image, where layer 0 has the highest contribution, and the contribution decreases as the layer number increases. The number of quality layers can specified at the encoder according to the requirements for quality adaptation granularity. The Motion-JPEG2000 encoder can generate image data packets of the same size for all quality layers. Therefore, for a data packet of quality layer $l$, we map this packet to a network packet with a priority label of $\mathbb{P}$, where $\mathbb{P} = l$. When generating the network packet, the $\mathbb{P}$ value along with a sequence number $\mathbb{S}$ (the number of all packets transmitted before this packet since the first packet of the first video frame) and the range of priorities $\mathbb{R}$ ($\mathbb{R} = L$), where $L$ is the total number of quality layers in an encoded frame. Table 1 depicts the layout of the network packets.

**Table 1:** Network packet header layout

| Packet sequence number ($\mathbb{S}$) | Packet priority ($\mathbb{P}$) | Priority range ($\mathbb{R}$) | Payload (Original image data packet) |
|---|---|---|---|

When a packet is received, either by the *Rx Mapper* host or by an active router, the embedded priority can be used to process the packet accordingly. For a *Rx Mapper*, the 2-tuple $(\mathbb{S}, \mathbb{R})$ can be used to restore the order of image data packets in a frame. For an active router, a relative priority value $\eta$ can be derived from $\mathbb{P}$ and $\mathbb{R}$ by $\eta = \mathbb{P}/\mathbb{R}$. According to $\eta$ and currently load $\lambda$ and a threshold value $\beta$, the active routers can make a decision on whether to drop the packet or to forward it to the next hop.

### 2.2 Rate Adaptation

To implement the network resource adaptation using the mapping scheme as we discussed previously, we will need a *dropper* active application on active routers. A *dropper* can be anywhere on the route between a sender and a receiver. A *dropper* adjusts the video flows to adapt to the available bandwidth and buffer resources. The rate adaptation on the active routers can be quite efficient in our scheme. When the *dropper* receives a video packet in a packet flow under monitor, it can extracts the packet priority ($\mathbb{P}$) and the packet priority range ($\mathbb{R}$), and calculates an importance parameter $\eta = \mathbb{P}/\mathbb{R}$. Meanwhile, the active router keeps track of the current load on the destination outgoing port, say $\lambda$. If $\lambda$ is greater or approaching a threshold parameter $\beta$ at the time when the video packet comes in, and $\eta > \beta$, the *dropper* drops the video packet to preserve outgoing bandwidth. The pseudo code of the rate adaptation algorithm is shown in Algorithm 1.

**Algorithm 1** Rate adaptation algorithm on the active router

```
loop
  if a video packet v comes in from the monitored packet flow then
    if λ < β then
      forward v to outgoing port
    else
      extract IP and IR from v
      η = IP/IR
      if η > β then
        drop v
      else
        forward v to outgoing port
      end if
    end if
  end if
end loop
```

## 3 Simulations and Experimental Results

### 3.1 Simulations

In order to evaluate the performance of our proposed scheme, we simulate a video transmission scenario over the Internet with three PCs. Each PC acts as a sender, a receiver, and an active router, respectively. Different IP packet dropping schemes are employed on the active router node, including random packet dropping and selective packet dropping. When the random packets dropping scheme is used, two schemes are used to handle received network video packets at the receiver. Namely, we either use all the received packets to reconstruct the coding stream, or use only network packets that have higher priorities than the dropped packets to reconstruct the coding stream. Table 2 lists the simulations we ran.

Table 2: Simulation schemes.

| Name | Packet dropping scheme at the active router | Packet reassemble scheme at the receiver |
|---|---|---|
| *noloss* | No packet drop (Baseline comparison) | Use all received packets to reconstruct the image |
| *maxlyrs* | Random packet drop | Use packets before the dropped one to reconstruct the image |
| *discard* | Random packet drop | Use all received packets to reconstruct the image |
| *lastlayr* | Packet drop by priority scale | Use all received packets to reconstruct the image |

We measure the peak signal-to-noise ratio (PSNR) at the sender sider before the packet is transmitted and at the receiver after the final images are reconstructed. The PSNR is generally used as a good indicator of the image quality. By comparing the PSNR changes with different packet dropping schemes, we can assess the impacts of the packet dropping to the image quality. Different packet dropping probabilities are used reflecting the degree of network congestion. For a packet drop probability of $p$, we can infer the downstream link is $(1/(1-p))$ overloaded. We used the $p$ values of 5%, 10%, and 20% in our simulations.

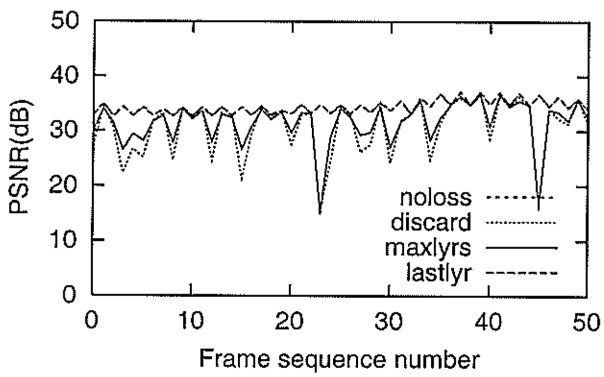
The video sequence we used (referred as *susi* video sequence) is a widely-used video sequence. It is a natural video sequence with image size of $714 \times 480$ pixels, three components (R, G, B), and 8 bits per pixel per component. The JPEG2000 codec used is *JasPer*, which is under consideration by ISO as an official reference implementation.

### 3.2 Experimental Results

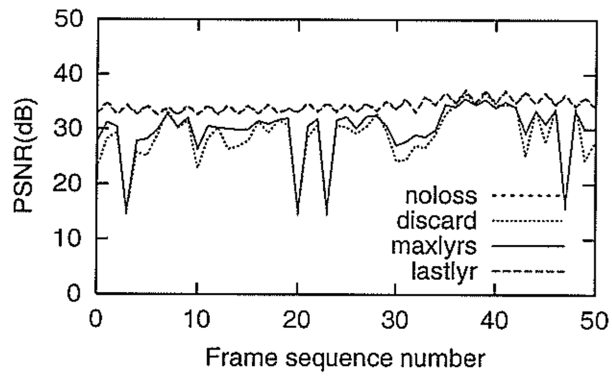
Our experimental results of the PSNR tests are shown in Fig. 2. The curves are labeled with the simulation names in Tab. 2.

As shown in Fig. 2, the PSNR of video frames with dropped packets decrease dramatically if no selective packet drop scheme is used. As the packet drop probability increases, the PSNR curves of *discard* and *maxlyrs* decrease substantially. However, the PSNR curves with our proposed selective packet dropping scheme (*lastlyr*) stays very close to the reference curve (*noloss*). This is because in our scheme the active router drops the packets with lowest priorities with inverse proportional to the traffic load and proportional to the packet drop probability. These result show that the packets dropped using our scheme have little to the final video quality, whereas random packet drops cause substantial degradation to video quality.

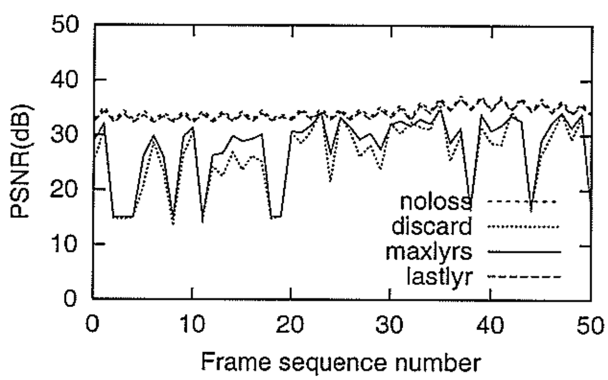
In addition, these results also show that the image reconstruction methods at the receiver end do not affect the quality of the reconstructed video. In the *discard* simulation, the receiver uses all received packets to reconstruct the video frame disregarding the dropped packets; in the *maxlyrs* simulation, if the receiver detects that there is a dropped packet in the received packets, it discards the rest of packets in the frame, and uses only the reliablely received packets to reconstruct the frame. The results show that the *maxlyrs* curve has better PSNR tha *discard*. This is because of the inter-layer dependency in the encoded images. Even if the *discard* scheme takes more data, it can not reconstruct images with higher quality because there is information missing from the previous layer. This is the standard behavior of the decoder. Changes can be made to the decoding algorithm by reduce inter-layer dependency to cope with this issue. However, this adds penalty to normal video transmission without loss because it reduce the compression efficiency.



(a) Packet drop probability: 5%



(b) Packet drop probability: 10%



(c) Packet drop probability: 20%

**Fig. 2:** PSNR results of different packet dropping schemes when the packet dropping probability are 5%, 10%, and 20%.

## 4 Conclusions and Future Work

In this paper, we describe an efficient quality adaptive transmission scheme for Motion-JPEG2000 video sequences over congested networks using an active network approach. We evaluate our scheme with simulation, and our results show that our scheme can provide video better quality scalability when network congestion happens. Because of our use of quality layers as a guide for packet discards, we can maintain high video quality even when congestion happens.

Currently, we are working on implementing our video processing as an active router plugin in a real active router environment [4]. The preliminary results show that it is very efficient. Meanwhile, we are looking into the performance issues of the wavelet-based codec because it is necessary for real time video applications, and the reference codec can achieve efficiency. We will analysis the computational complexity of the JPEG-2000 codec, and exploit the possibilities of improve its performance by parallel processing in its bit-plane coding.

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