MOTION VIDEO CODING FOR PACKET-SWITCHING NETWORKS — AN INTEGRATED APPROACH

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ABSTRACT

The advantages of packet video, constant image quality, service integration and statistical multiplexing, are overshadowed by packet loss, delay and jitter. By integrating network-control into the image data compression algorithm, the strong interactions between the coder and the network can be exploited and the available network bandwidth can be used best. In order to enable video transmission over today's networks without reservation or priorities and in the presence of high packet loss rates, congestion avoidance techniques need to be employed. This is achieved through rate and flow control, where feedback from the network is used to adapt coding parameters and vary the output rate. From the coding point of view the network is seen as data buffer. Analogously to constant bit rate applications, where a controller measures buffer fullness, we attempt to avoid network congestion (eq. buffer overflow) by monitoring the network and adapting the coding parameters in real-time.

1. INTRODUCTION

The transmission of video over packet-switching networks enables novel applications based on the integration of several services into a common format. It is believed that the integration of audio, video, and discrete data into multimedia-workstations connected by packet-switching networks will have as much impact as the introduction of the personal computer itself.

Conventionally, only circuit-switching, application-specific networks have been used for video, e.g. CATV. Usage of these constant-rate channels for variable-rate transmission results in wasted capacity since the channel has to accommodate worst-case rates. Using packet-switching network, video delivery with variable rate and constant quality can be achieved efficiently by virtue of statistical multiplexing[2,20]. A thorough analysis and modelling of statistical multiplexing is given in a paper by Maglaris et al.[23].

In order to exploit the advantages of packet video, service integration, constant quality, and statistical multiplexing, the following problems associated with the transmission of video over packet-switching networks have to be overcome: First is the discrepancy between the data rates that can be accommodated by today's computer systems and networks on the one hand, and the values associated with real-time video that have to be bridged through data compression techniques on the other hand. Fortunately, images contain a lot of redundant and irrelevant information that can be successfully exploited by several compression algorithms.

Secondly, the service of continuous media is associated with certain end-to-end requirements in terms of bandwidth, delay and jitter that are not met by today's packet-switching networks or computer systems. The general directions of research that are currently pursued can be categorized into two classes: The first approach aims at extending current operating systems and network protocols to provide performance guarantees[6,24]. These schemes necessitate admission control and will not be in service in the near future on a broad scale. The other approach adapts the continuous-media stream to the given systems, in which service is not guaranteed but is delivered on a best-effort basis. In these systems, packet loss, for example, has to be compensated for by the receiver. This leads to a complex problem, because not only are both channel capacity and data rate variable, but they also interact.

Several researchers have begun investigating video coding algorithms for transmission of video over packet-switching networks. Their work points out the advantages of constant image quality and bandwidth sharing by statistical

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multiplexing; they propose that redundancy and error correction protocols be employed to deal with packet loss; but, in general, they fail to provide adequate solutions to the network congestion problem. In the literature we observed that either an image data compression viewpoint or a networks viewpoint is taken: image coding algorithms are mostly simulated using greatly simplified network models, e.g. fixed packet loss rate, and/or they assume the existence of network services not yet widespread such as networks, which respect different packet priorities or offer classes of service[7,17], and/or networks that guarantee the delivery of a certain percentage of the packets[10]. Network issues, on the other hand, are mostly based on models of the video source, e.g. Markov models[23].

Among the techniques proposed for packet video coding are vector quantization[12], subband coding[3,17], motion-compensated interframe coding[26], two-layer coding[10], pyramidal encoding[28], discrete cosine transform[20], and DPCM[30]. A good overview of new image coding results for ATM networks can be found in the recent Image Communication journals[13]. Network issues, which are also partly discussed in the literature cited above, are error recovery, congestion control[15,8], statistical analysis and modelling of packet loss[22], and characterization of video sources[25].

Although the mutual dependency of video coding and packet networks has already been pointed out by several researcher[1,18,7], no satisfactory solutions have been proposed so far. Since channel bandwidth varies with network load conditions, minimum packet loss will be achieved by senders that can adjust dynamically their maximum transmission rates to match the channel bandwidths. Motion video compression algorithms are suited to produce such changing maximum transmission rates as several compression parameters can be varied[4] in response to congestion signals from the network. Thus, we propose an integrated approach to motion video coding for packet-switching networks: the coding algorithm and the network (through a network estimator) are part of a feedback loop that seeks to avoid network congestion. Unlike other approaches that try to maintain constant picture quality, our scheme produces lower received signal quality when the network load increases. However, in general, we use bandwidth more efficiently since a coding agent ensures best image quality for a given bandwidth.

Section 2 of this paper introduces the general notion of feedback communication system and their applicability to packet video. This leads to a network-integrated solution where the networks serves as data buffer. The main components of this system – video coding agent and network estimator – are discussed in section 3. The results of experiments with a real-time software coder that uses local-area networks interconnected by routers to transmit video frames are reported in section 4.

2. FEEDBACK COMMUNICATION SYSTEM

One of the great promises of variable bit rate coding is constant image quality. Unlike in constant bit rate environments, where the quality of demanding scenes may be severely degraded in order to stay below line bit rate, packet networks offer capacity on demand. This advantage is overshadowed by the possibility of packet loss, which may affect image quality at the receiver even more severely than normal coding artifacts, and by the variability of packet delays, which may make it difficult to use packet-switching networks for interactive applications. Data transmission protocols overcome packet loss by retransmission at the cost of added delay and network load. Retransmission of lost packets