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Scaleable Delivery Mechanism for VBR MPEG-2 Video Streams over the Internet

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Technical Areas Suggested: Video-On-Demand, Rate Control, and Internet Traffic Management, etc.

Extended Abstract

Video applications range from low-bit-rate video conferencing to high-bit-rate full motion broadcast quality videos like Video-On-Demand. VBR transmission services are likely to play an important role in these applications because of the substantial benefits they provide, both in terms of network utilization and video quality.

With the emergence of Fast Ethernet and Gigabit switching capabilities, many people are now attempting to use IP-based networks to deliver pre-compressed video streams, like MPEG-2. However, the current IP-based networks cannot provide Quality of Service (QoS) guarantees. Therefore, when an IP-based network is getting congested, the MPEG-2 video quality may suffer extreme degradation as the packets are delayed excessively and lost in the network. However, how the real-time video can be delivered with some pre-defined level of QoS in such networks is a challenging problem. In this paper we propose a new scaleable delivery mechanism for pre-compressed VBR MPEG-2 video streams, where the traffic rate is adapted according to the network congestion status. Using our scheme, a reasonable level of QoS is maintained when the network is congested. Based on the proposed scheme a prototype system has been developed which demonstrates the usefulness of the scheme.

A substantial amount of work has been reported in the literature on the scaleable delivery of MPEG-2 video, for example, data partition, spatial, temporal and frequency resolution scaling [2]. Data partition assigns different frequency DCT coefficients with different priorities. Spatial resolution scaling involves switching among multiple video streams pre-compressed at different resolutions. Temporal resolution scaling involves selectively skipping frames to reduce the bit rate. Frequency resolution scaling involves requantizing the quantized information using a higher quantization step so that the desired bit rate can be achieved.

However, all these schemes have intrinsic disadvantages. Data partitioning has a lack of flexibility. Spatial resolution scaling requires more storage space to store multiple streams. Temporal resolution scaling introduces more serious video quality degradation compared to other schemes. Frequency resolution scaling increases the complexity of the server program and introduces a long processing delay.

Main features of our proposed scaleable delivery scheme are as follows. Firstly, the scheme adopts a protocol that can provide feedback of the network status in real time. For example, Real-Time Control Protocol (RTCP) [3] can be used. Secondly, our scheme is based on the feature that AC coefficients of

different frequencies have different contributions to the video quality, and that human beings are not very sensitive to the video quality degradation caused by the removal of high frequency AC coefficients. When applying our scheme to real applications, say a video-on-demand content server, an MPEG-2 bit stream is split and stored into a base file and few delta files. The base file contains headers, motion vectors and low frequency DC coefficients to maintain the basic video quality. The delta files contain the remaining high frequency AC coefficients, as shown in Figure 1.

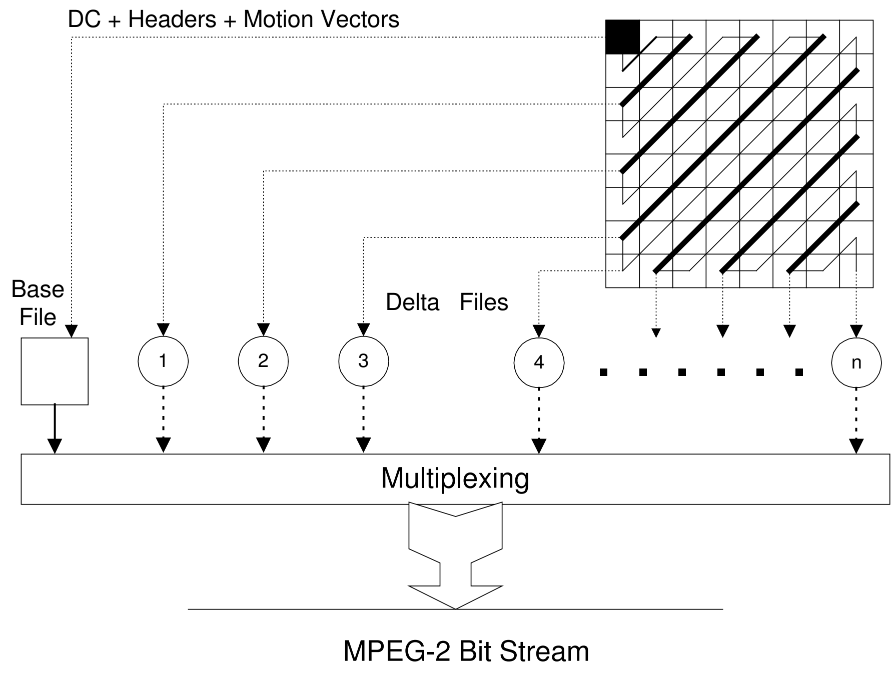

Figure 1: MPEG2 data organization for scaleable delivery

During transmission, the server dynamically multiplexes the base file and a certain number of delta files into a MPEG2 stream to be delivered. Under the normal network conditions, all files are combined to form the MPEG-2 video stream. However, when enough bandwidth is not available, only the base and some of the delta streams are multiplexed. The number of delta streams to be used in the multiplexing is determined at the video server according to the current network status obtained from the RTCP. The quality of the delivered video stream will vary depending upon the delta files used in the multiplexing. Figure 2 gives the indication of the video quality when the video stream is organized in one base file and five delta files. In this case, six different levels of video quality are possible depending upon the network conditions. The left side image shows the picture quality when all delta files are merged at the server before transmission. On the other hand, the right side image indicates the worst video quality when only the base file is transmitted.

In our implementation the multiplexed video stream is delivered using Real-Time Protocol (RTP). The algorithms have been devised and implemented to perform the multiplexing in an efficient manner, so that the scaleable delivery scheme can be implemented in real-time. The environment comprised a single video server and multiple client PCs/Workstations in a LAN environment or the Internet. The VBR MPEG-2 video stream stored in the server is split into the base and three delta files. During the transmission, the base stream and a proper number of delta streams are dynamically multiplexed and encapsulated into RTP packets following [4]. The number of delta files multiplexed is determined according to the packet loss ratio and jitter information provided by RTP/RTCP.

Among the design issues considered are the transport mechanism to be used (RTP/RTCP), CBR Vs VBR, number of delta files a video stream should be split into (2-5 ideally), number of delta files to be merged, network parameters to be taken into account (packet loss ratio and delay jitter), frequency of the network status to be obtained, content of the base file and the structure of the delta files.

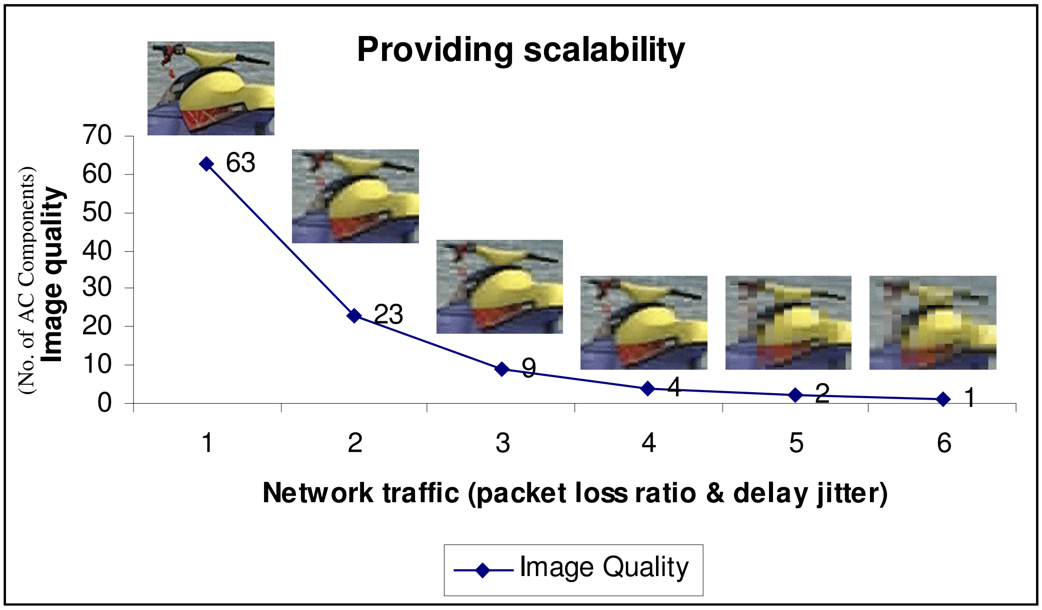

Figure 2 Number of delta files to multiplex is dependent on network traffic

Our design is more efficient than the existing approaches discussed. Compared to data partition, it is more flexible as we can produce as many delta-files as needed. Compared to spatial resolution scaling, much less extra space is required as delta files contain only AC coefficients and no motion vector information. Compared to temporal resolution scaling, our design results in a better video quality as no frames are skipped. Compared to frequency resolution scaling, the server program is simpler and introduces less processing delay as there is only a multiplex process and no re-quantization is involved. The scheme leads to an efficient utilisation of network resources and a smooth real-time video transfer with video quality adaptive to network status.

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