MODELING AND SIMULATION OF MPEG-2 VIDEO TRANSPORT OVER ATM NETWORKS CONSIDERING THE JITTER EFFECT

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Abstract - In this paper, the operation of MPEG-2 systems is modeled and simulated when an MPEG-2 transport stream is delivered through a ATM network with jitter. A novel approach to analyzing the decoder buffer behavior in the presence of network jitter is presented. The probability density function of the interarrival time of the ATM adaptation layer 5 (AAL5) Protocol Data Unit (PDU) is derived from a MPEG-2 video source model and an ATM network jitter model. Based on a real-time decoding requirement of the MPEG-2 transport stream (TS) system target decoder (T-STD), the decoder buffer behavior is simulated. The modeling and simulation results show that jitter affects decoder buffer size and packet loss ratio in a significant way.

INTRODUCTION

There has been great interest in video transport over ATM networks in recent years. Due to the statistical multiplexing capability of ATM and increased transmission bandwidth capacity, ATM can support multimedia applications, i.e., audio, video, and data, simultaneously. Video can be transported over ATM networks either at a constant bit-rate (CBR) or variable bit-rate (VBR). Recent research interest has been focusing on VBR video since it utilizes network resources much more efficiently than CBR video. However, the use of statistical multiplexing in ATM networks introduces cell delay variation (CDV or jitter) and cell/packet loss, which may affect the video quality at the receiving end. This paper models the end-to-end trans-

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port of MPEG-2 video and investigates the impact of ATM network jitter on the operation of the MPEG-2 decoder.

The MPEG-2 standard (ISO/IEC 13818) specifies the operation and interaction of video and audio coding, as well as related system functions [1]. The standard assumes that the end-to-end delay from MPEG-2 system's encoder to decoder is constant. This is necessary to ensure that the encoder and decoder clocks operate at the same frequency and the decoder buffer does not overflow nor underflow. On the other hand, an ATM network introduces jitter when transporting MPEG-2 packets from a source to a destination. Due to this jitter, the behavior of the decoder buffer differs from that of the encoder buffer, which may cause problems in decoder operations.

To the best of our knowledge, most strategies for preventing decoder buffer from underflowing or overflowing use rate control schemes without considering the ATM network jitter [2] [3][4], except in [5]. In this paper, we employ a novel approach to investigate the decoder buffer behavior in the presence of ATM network jitter. Specifically, we first generate MPEG-2 TS packets based on a statistical model at the source. We then segment these TS packets into ATM cells and simulate their transport over ATM networks by introducing jitter into these cells. Finally these cells are reassembled into TS packets, which are delivered into the upper layer TS packet decoder. We simulate the behavior of the decoder buffer under a real-time decoding requirement and show the interactions among packet loss ratio, decoder buffer size, and network jitter level.

MPEG-2 TS PACKETS GENERATION PROCESS

MPEG-2 source models have been developed for one-layer [6] and two-layer coding modes [7]. In this paper, we only consider a video source generated by a one-layer coder. Specifically, we use the MPEG-2 video source model proposed by Heyman et al. [6]. In this model, the video sequence is progressive and is coded using the Main Profile at Main Level (MP@ML) with a fixed quantizer step size. The coding parameters used are M=3 and N=15, which yields the following fixed pattern within each group of pictures:

$$I\ B\ B\ P\ B\ B\ P\ B\ B\ P\ B\ B\ P\ B\ B.$$

The I, B and P frames are modeled with a log-normal distribution with the following mean (η) and standard deviation (σ) (in unit of cell[1]): $\eta = 256.6, \sigma = 75.4$ for I-frame; $\eta = 93.4, \sigma = 41.6$ for P-frame, and $\eta = 86.0, \sigma = 20.5$ for B-frame.

With the above statistics, we generate a MPGE2 bitstream at the frame rate of $R = 30$ frames/sec. Within a frame interval (1/R=1/30 second), successive TS packets (188 Bytes each) are generated with equal time interval.

[1]Each ATM cell contains 48 bytes of payload or 384 bits.

That is, TS packets are transmitted at the same rate within a frame, although the interarrival time of TS packets in separate frames are different.

TRANSPORTING MPEG-2 TS PACKETS OVER ATM NETWORKS

Here, we consider the transport of MPEG-2 transport streams over ATM networks using AAL5. In this scheme, MPEG-2 TS packets are mapped into AAL5 protocol data units (PDUs), each containing 2 TS packets. Each PDU is then converted to 8 AAL5 cells with additional header information attached. At the destination, 8 ATM cells are combined into an AAL5-PDU, which is then decomposed into 2 TS packets. Let $\mathbf{T}_{PDU,S}$ denote the PDU interarrival time at the source side, $\mathbf{T}_{PDU,D}$ the PDU interarrival time at the destination, and $\mathbf{T}_{PDU,N}$ the network introduced jitter for a PDU. Then

$$\mathbf{T}_{PDU,D} = \mathbf{T}_{PDU,S} + \mathbf{T}_{PDU,N}. \tag{1}$$

Let $\mathbf{T}_{CDV}$ denote the CDV and $f_{CDV}(T)$ its probability density function (PDF). Since the jitter of a PDU is determined by the jitter of the last cell in this PDU, the PDF of $\mathbf{T}_{PDU,N}$, $f_{PDU,N}(T)$, is equivalent to $f_{CDV}(T)$ if we assume that jitters of all ATM cells are independent and identically distributed. In this paper, we assume $f_{CDV}(T)$ is a zero mean truncated Gaussian function with variance σ^2. Assuming that $\mathbf{T}_{PDU,S}$ and $\mathbf{T}_{PDU,N}$ are independent, then

$$\begin{aligned} f_{PDU,D}(T) &= f_{PDU,S}(T) * f_{PDU,N}(T) \\ &= \int_0^\infty f_{PDU,N}(T-x) f_{PDU,S}(x)dx, \end{aligned} \tag{2}$$

where $f_{PDU,S}(T)$ and $f_{PDU,D}(T)$ are the PDFs of PDU interarrivals at the source and decoder, respectively.

DECODER BUFFER SYSTEM DEFINITION

We model the decoder buffer operation based on the MPEG-2 TS system target decoder (T-STD) [1]. MPEG-2 video packets enter the T-STD decoder buffer after they are reassembled from ATM cells. Here we consider a simplified version of the T-STD in which decoding is performed at a fixed time interval (1/R second). We contend that even with this simplified decoder, the salient features of the MPEG-2 decoding mechanism still remain and the overall performance would not change in any significantly way. Also, simulation results based on this simplified model would give us the worst case scenario and provide an upper bound on the decoder buffer requirements.

In our simulations, at every frame interval of $1/R$ second, we check whether the TS packets for one video frame (knowns as an access unit) have arrived completely. If all the TS packets have arrived, they are removed from the decoder buffer instantly. Otherwise we wait and check for one more frame time interval (equal to $1/R$) to see whether the data for this frame are complete. After this extra frame time interval, the arrived data for this frame are removed from the decoder buffer even though they may be incomplete. The late PDUs are counted as lost PDUs in our packet loss ratio calculation (in addition to those PDUs lost due to decoder buffer overflow). Note that for the purpose of extracting information that may be essential for the decoding of the next frame, all the TS packets that arrive at the decoder buffer will be decoded, even though some of them do not meet timing requirements.

SIMULATION RESULTS OF DECODER BUFFER BEHAVIOR

We have performed an end-to-end system simulation based on the source model and system definition described above. The variances of jitter considered are 10^{-6}, 2×10^{-6}, 5×10^{-6}, and 10×10^{-6} second^2. Figure 1 shows the interactions among decoder buffer size, packet loss ratio and jitter levels. It can be seen that the jitter level significantly affects the packet loss ratio. Specifically, at a relatively small jitter variance, the packet loss ratio reduces when the buffer size increases. When the jitter is sufficiently large ($\sigma^2 > 5 \times 10^{-6}$), increasing buffer size does not affect the packet loss ratio. This is because, when the jitter is small, packet loss is mainly caused by buffer overflow. On the other hand, when the jitter is very large, late packets contribute to the majority of lost packets. In other words, the jitter level is the predominant factor that determines the packet loss ratio. It is useless to increase the physical buffer size since the late packets cannot meet the time constraint.

CONCLUSIONS AND DISCUSSIONS

We have conducted an end-to-end modeling and simulation of MPEG-2 video transport over ATM networks with the consideration of the effect of jitter. The PDF of the PDU interarrival time at the destination is derived based on a video source model and a network jitter model. Our simulation is implemented based on the real-time decoding requirement of MPEG-2 T-STD. The modeling and simulation results show the tradeoff and interaction among the decoder buffer size, packet loss ratio and network jitter level.

When packet loss occurs, a feedback information may be sent to the encoder to inform the encoder to change the bitrate by adjusting the quantization step. A study of the decoder buffer behavior when the encoder excises such kind of rate control may yield more interesting and useful insights.

Recently, Internet is growing explosively. Study of MPEG-2 video transport over Internet Protocol (IP) based networks is becoming imperative. To

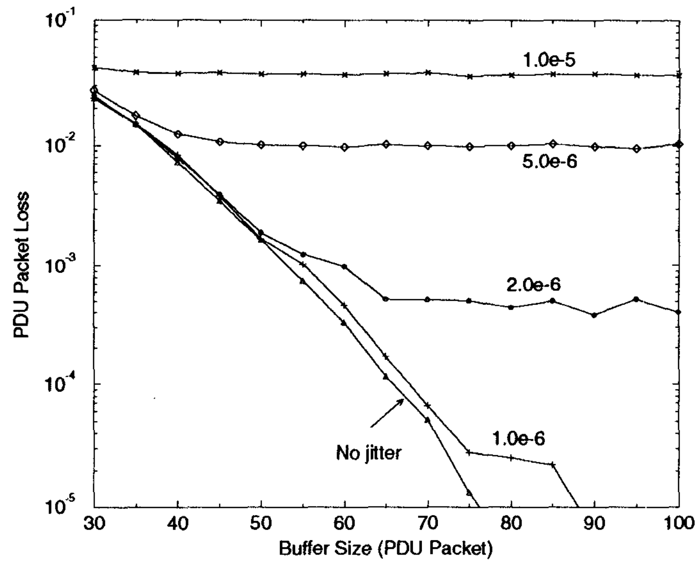

Figure 1: Relations among packet loss ratio, buffer size and jitter level. The number above each curve specifies the variance of the jitter simulated.

provide real-time transport of MPEG-2 video over IP networks, the real-time transport protocol (RTP) is expected to be used above the TCP or UDP protocol. To transport MPEG2 TSs with RTP, multiple TS packets can be encapsulated into RTP packets. At the same time, the resource reservation protocol (RSVP) can be used for dynamic reservation of network resources (such as bandwidth and QoS). Compared to the ATM network, the IP network will introduce significantly larger delay jitter. Therefore, study of the jitter impact on decoder buffer in IP networks are even much more challenging. This is one of our future work.

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