Decoder Buffer Modeling and Simulation for End-to-End Transport of MPEG2 Video with ATM Network Jitter *

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Abstract

In this paper, the operation of MPEG-2 decoder buffer is modeled and simulated when a VBR MPEG2 stream is delivered through an ATM network with jitter. End-to-end packet based analysis is performed for delivery of MPEG-2 transport streams over ATM networks. A novel approach to analyzing the decoder buffer in the presence of network jitter is given in this paper. The probability density function of the interarrival time of the ATM adaptation layer 5 (AAL5) Protocol data unit (PDU) is derived from a bit-rate model of the video source as well as a ATM network jitter model. Based on the decoding timing requirement of the MPEG-2 system target decoder, a simulation of the decoder buffer is implemented. In the simulation, the transport stream packets arrivals follow the derived probability density function of the AAL5 PDU interarrival time. The modeling and simulation results show that packet loss occurs for a given buffer size, which happens when the TS packets arrive in burst because of network jitter.

Key Words: MPEG2 Video, ATM Network Jitter

I. Introduction

There has been tremendous interest in video transport over ATM networks recently. Due to the statistical multiplexing capability of ATM and the abundant transmission bandwidth capacity, ATM can support multimedia application, i.e., video, audio and data simultaneously. Video can be transported over ATM network either at a constant bit-rate (CBR) or variable bit-rate (VBR). Recent research interest has been focusing on VBR video since it has several advantages over CBR video, such as constant quality picture and low delay. However, the statistical multiplexing characteristics of ATM results in delay jitter and cell/packet loss, which then affect the quality of reconstructed video. In this paper, we analyze the influence of the network delay jitter on the MPEG2 decoder.

The MPEG2 standard (ISO/IEC 13818) specifies the operation and interaction of video and audio coding, as well as related system functions[1]. It supports full motion video and transmits audio-video information at about 4, 10 and 20 Mbits/s with an image quality similar to respectively the present standard TV systems, the one specified by ITU-R recommendation 601 and HDTV. Its video coding standard is also adopted by the US Grand Alliance HDTV system [2]. MPEG2 system assumes that the delay from encoder to decoder (end to end delay) is constant [1]. This ensures that the encoder and decoder clock operate at the same frequency such that the decoder buffer will not overflow or underflow. On the other hand, ATM networks vary in delay in delivering the data stream from encoder to decoder. This type of variation in the network delay is known as network introduced jitter and is called cell delay variation (CDV). Due to this jitter, the decoder buffer behaves differently from the encoder buffer. In general we assume the decoder buffer operates in a clock frequency that oscillates slightly about that of the encoder buffer, which may cause the decoder buffer either underflow or overflow.

Until now, most strategies of preventing decoder buffer from underflowing or overflowing are considered from the source side by using rate control scheme without considering the ATM network jitter [9] [6] [5]. In this paper, a novel approach to analyze the decoder buffer behavior in the presence of network jitter is proposed. First, we derive a statistical model of MPEG2 transport stream (TS) packet interarrival time. Specifically, the probability density function (PDF) of the AAL5 PDU interarrival time in destination is derived from an assumed model of the bit rate in the source and an assumed model of ATM network jitter. Based on this model and the MPEG2 System Target Decoder, we simulated the behavior of the decoder buffer. Simulation results are shown in terms of the relation between average buffer size vs. utilization and that between packet loss ratio (PLR) vs. buffer size. These relations are important for designing the decoder buffer.

The rest of this paper is arranged as follows: In Section II, an overview of the MPEG-2 system and timing model is presented. Section III describes the MPEG2 transport scheme over ATM networks. Section IV derives the PDFs of AAL5 PDU interarrival times, which is equivalent to the PDF of the TS packet interarrival times by a scaling factor. Section V presents results

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of a simulation study of the decoder buffer based on the PDF derived from Section IV. In Section VI, a conclusion is given, along with a discussion on further work.

II. Overview of MPEG2 System and Timing Model

As stated above, MPEG2 supports a large number of applications, including terrestrial digital TV broadcasting, 2-way communication, video on demand, video on LANs, and HDTV over cable, satellite, terrestrial and broadband networks. It also supports interactive video, such as video on multimedia workstation.

The MPEG2 system first packetizes elementary streams (compressed video sequences) to produce PES (packetized elementary streams) packets. PES packets are further combined with system information to form Transport Streams (TSs) or Program Streams (PSs) by multiplexing. The PS results from combining one or more PES packets into a single stream all of which have a common time base. The PS is designed for use in error-free environments and is suitable for applications which involve software processing of system information such as interactive multimedia applications on CD-ROM. PS packets may be of variable and relatively great length. The TS combines one or more programs with one or more independent time bases into a single stream. A program here is a collection of elementary streams with a common time base. The TS is designed for use in environments where the errors are likely, such as lossy or noisy storage or transmission media, e.g., video distribution over long distance networks and in broadcasting system, in which packet losses may occur. The TS packets are of fixed length of 188 bytes. In this paper, we only consider TS packets.

Next we will mainly address the timing issue in MPEG2 system. MEPG2 system assumes that the delay from the encoder to the decoder (end to end delay) is constant[1]. There is a single common system clock in the encoder, and this clock is used to create time stamps that indicate the correct presentation time (Presentation Time Stamp-PTS) and decoding time (Decoding Time Stamp-DTS), as well as to create time stamps that indicate the instantaneous value of the system clock itself (System Clock Reference-SCR in Program Stream; Program Clock Reference-PCR in transport Stream). The PCR was sent at least at every $100ms$ or at a equivalent frequency of 10 Hz. The recreation of the system clock in the decoder and the correct use of the time stamps (DTSs and PTSs) provide a synchronization between the encoder and the decoder. Since the decoding timing affects the behavior of the decoder buffer, correct recovery of the PTS and DTS in decoder guarantees that the decoder buffers will not overflow nor underflow. The correct PCR value can be used to set the instantaneous value of the decoder STC. In practice, in order to match the decoder STC with encoder's STC, the decoder STC must slave its timing to encoder using the received PCR. The usual method of slaving the decoder's clock to the received data stream is via a phase-locked-loop(PLL) [1].

If a network varies in delay in delivering the data stream from encoder to decoder, such variations tend to cause a difference between the received PCR and the actual PCR. Since the received PCR value is used to set the instantaneous value of the decoder's STC, this may cause the decoder STC to fluctuate when the PLL is used to recover the source clock from the received PCR. When recovered STC is not at the same frequency as that in the encoder due to jitter, buffer fullness of the decoder can't be maintained to a level compatible with that of the encoder. This causes the decoder to be either overflew or underflew. In applications where a significant amount of PCR jitter is present at the decoder, additional buffer space at the decoder is needed to absorb the jitter.

III. MPEG2 Transport over ATM Networks

The issue of how to transport MPEG2 over ATM network has been debated for a while. The topics of discussions include the class of service (CBR-constant bit rate vs. VBR-variable bit rate) for ATM connection, the ATM adaptation layer to be chosen and the kinds of additional functionalities above ATM network to be provided (including clock recovery in the presence of jitter and error concealment).

MPEG2 TS packets will be adapted before entering ATM networks. For this adaptation, several layers are defined based on the class of service. ATM adaptation layer 1 (AAL1) has been defined to support CBR traffic, and an adaptive clock method to smooth the network jitter has been proposed. The downside of AAL1 is that too much overhead is needed. AAL2 was proposed for VBR traffic. However, it was not well-defined and has not been considered so far. ATM adaptation layer 5 (AAL5) originated for available bit rate (ABR) data transportation can support both CBR and VBR video. Recently, the ATM Forum has reached an agreement on transporting MPEG2 over ATM networks using AAL5[11].

When transporting MPEG2 over ATM networks using AAL5, MPEG-2 TS packets are mapped into AAL5 packets with a null service specific convergence sublayer (SSCS). The mapping of MPEG-2 TS packets into the AAL5 service data unit (SDU) will be referred to as 1/N mapping. The default AAL5 CPCS-SDU (Common Part Convergence Sublayer) size is 2 TS packets, i.e., N=2. An AAL5 PDU containing 2 TS packets can be converted to 8 AAL5 cells. The first 7 AAL5 cells have 48 bytes payload each, and the last AAL5 cell has 40 bytes payload plus an 8-byte trailer. Finally, the 5-byte ATM cell header is added to all 8 AAL5 cells to form 8 ATM cells with 53 bytes each.

To transport MPEG2 TS streams over ATM networks, from ATM networks side, it is standardized by the ATM Forum that first, connection admission control (CAC) is needed between user and network for a call setup (policing) to indicate call rejection or acceptance based on required QoS (Quality of Service). Once a connection is established, then secondly, usage parameter control (UPC) is used which employs a leaky bucket (for CBR) or multiple leaky buckets (for VBR) to monitor the traffic (traffic shaping).

At the destination, an AAL5 PDU is re-assembled by accumulating eight ATM cells. Then $N(=2)$ TS packets are recovered from an AAL5 PDU. The details will be discussed in the next section. Due to the use of statistical multiplexing in ATM switches, the jitters in cell delay are accumulated to generate the AAL5 PDU jitter, and consequently TS packet jitter. Here the delay jitter means a time deviation from the expected constant delay. Jitter is dimensioned in units of second. In this paper we only consider the jitter resulting from the ATM switches, but not the jitter incurred in the assembling and reassembling AAL5 PDUs.

IV. Modeling of AAL5 PDU Interarrival Time in the Presence of Network Jitter

To model the bit rate of a video source, two classes of traffic models have been investigated, i.e., the single source model and the multiplexed source model. The multiplexed source model is usually used in traffic management because of the capability of capturing the effects of statistically multiplexing bursty sources. Single source modeling is usually used for constructing a traffic descriptor, or used for end-to-end rate-control. In this paper, we use the single source model for obtaining the PDF of AAL5 PDU interarrival time in the source side. Based on this model, we simulated the interarrival times of the TS packets.

Single source modeling has been considered for one-layer coding and two-layer coding. For one layer coding, usually two models are used: discrete autoregressive (DAR) [4], [12] and discrete state continuous time Markov model [10], [8]. A model for two-layer coding is presented in [3]. In this paper, we only consider a video source generated by one-layer coding. For analysis convenience, we use the DAR model, which was first proposed by Heyman et al., [4] for video conference traffic. This model was later extended by Yugenoglu et al. [12] and applied to full motion video. In [12], a three-class AR model was used to model I, B and P frames and the PDF of the total bitrate is approximated by a composite Gaussian function. Let p_1, p_2 and p_3 ($\sum_{i=1}^{3} p_i = 1$) denote the steady state probability of the state 1, 2 and 3, then from the 3-class AR model, the PDF of bitrate ($\mathbf{R}$) is described by[12]:

$$f_{\mathbf{R}}(R) = \sum_{i=1}^{3} p_i G(\eta(i), v^2(i); R), \tag{1}$$

where $G(\eta(i), v^2(i); R)$ is the Gaussian PDF with mean $\eta(i)$ and variance $v^2(i)$. This bit rate model is used in this study.

Next we will show how to obtain the PDF of AAL5 PDU interarrival time in the source from the above PDF of the bitrate. To show the relation between the interarrival time and the bitrate, Fig. 1 illustrates AAL5 PDU packetization process. Fig. 1(a) is the relation between bitrate and time, Fig. 1(b) is the resulting AAL5 PDU after packetization, and T_i is the interarrival time between AAL5 PDU i and AAL5 PDU $i+1$ at the output of the encoder. Note that each AAL5 PDU contains two consecutive TS packets, which has a total length of $C = 376$ bytes. From Fig. 1(a), we obtain

$$\int_{t_i}^{t_i+T_i} R(t)dt = C. \tag{2}$$

Usually, the magnitude of R is very large and that of T_i is relatively very small, so we can approximate the above equation to

$$R_i \cdot T_i = C. \tag{3}$$

Thus we have

$$T_i = \frac{C}{R_i}. \tag{4}$$

From the above equation, we obtain th PDF $\mathbf{T}_{PDU,S}$ of the AAL5 PDU interarrival time at the source side:

$$f_{\mathbf{T}_{PDU,S}}(T) = \frac{C}{T^2} f_{\mathbf{R}}(\frac{C}{T}), \tag{5}$$

where $f_{\mathbf{R}}(\cdot)$ is the PDF of the bitrate of the video source.

Recall that each AAL5 PDU is split into eight ATM cells after adding proper trailer and ATM header information. In the destination side, after transporting through ATM networks, the AAL5 PDU interarrival time is equal to the AAL5 PDU interarrival time in the source ($\mathbf{T}_{PDU,S}$) plus the network introduced jitter. Using $\mathbf{T}_{PDU,N}$ to represent the AAL5 PDU jitter, i.e., the PDU delay variation (PDV), then the interarrival time of AAL5 PDU in the decoder is:

$$\mathbf{T}_{PDU,D} = \mathbf{T}_{PDU,S} + \mathbf{T}_{PDU,N}. \tag{6}$$

Next we will determine the relationship between the PDV and the CDV. After AAL5 re-assembly in the destination, 8 ATM cells are formed into an AAL5-PDU. This process is illustrated in Fig. 5. Let the PDF of the CDV, $\mathbf{T}_{CDV}$, be $f_{\mathbf{T}_{CDV}}(T)$, and assuming that the jitters of the 8 ATM cells are independent, then the PDF of an AAL5-PDU jitter, $\mathbf{T}_{PDU,N}$, is equal to the convolution of the PDF of the $\mathbf{T}_{CDV}$ with itself eight times, i.e,

$$f_{\mathbf{T}_{PDU,N}}(T) = \overbrace{f_{\mathbf{T}_{CDV}}(T) * f_{\mathbf{T}_{CDV}}(T)}^{convolve\ eight\ times}, \tag{7}$$

where * denotes convolution. In general the statistics of the network introduced jitter are unknown except that its average is zero [7]. Until now, most studies assume that the PDF of the CDV follows the Laplacian or Gaussian distribution. For simplicity, in this paper, we assume the CDV is a zero mean Gaussian random variable with variance σ^2, i.e.,$f_{\mathbf{T}_{CDV}}(T) = G(0, \sigma^2; T)$. Then, $f_{\mathbf{T}_{PDU,N}}(T)$ is also Gaussian, i.e., $f_{\mathbf{T}_{PDU,N}}(T) = G(0, 8\sigma^2; T)$. For representation simplicity, from now on we use $f_{PDU,D}(T)$, $f_{PDU,S}(T)$ and $f_{PDU,N}(T)$ represent $f_{\mathbf{T}_{PDU,D}}(T)$, $f_{\mathbf{T}_{PDU,S}}(T)$ and $f_{\mathbf{T}_{PDU,N}}(T)$, respectively. From Eq.(6), the PDF of the AAL5 PDU interarrival time in the destination is given by:

$$f_{PDU,D}(T) = \int_0^T f_{\mathbf{T}_{PDU,S},\mathbf{T}_{PDU,N}}(x; T-x)dx, \quad (8)$$

where $f_{\mathbf{T}_{PDU,S},\mathbf{T}_{PDU,N}}(x;y)$ is the joint PDF of $\mathbf{T}_{PDU,S}$ and $\mathbf{T}_{PDU,N}$. If we assume $\mathbf{T}_{PDU,S}$ and $\mathbf{T}_{PDU,N}$ are independent, then

$$\begin{aligned} f_{PDU,D}(T) &= f_{PDU,S}(T) * f_{PDU,N}(T) \\ &= \int_0^T f_{PDU,N}(T-x) f_{PDU,S}(x)dx. \end{aligned} \quad (9)$$

After AAL5-PDU de-accumulating, 2 consecutive TS packets are obtained from an AAL5-PDU. Assuming the de-accumulating process has a constant delay, the interarrival time between pairs of TS packets is equal to the interarrival time of an AAL5-PDU. On the other hand, the interarrival time between the two adjacent TS packets in one AAL5 PDU is zero. Fig. 2 illustrates an example of the PDF of $\mathbf{T}_{PDU,D}$, which is obtained by using simulation parameters given in the next section.

From the PDF in Eq.(9), we can derive the PDU interarrival rate by:

$$\lambda_{PDU} = \frac{1}{E\{\mathbf{T}_{PDU,D}\}} . \quad (10)$$

Recall that each AAL5 PDU contains two TS packets, therefore, the arrival rate of the TS packets is:

$$\lambda_{TSP} = 2\lambda_{PDU}. \quad (11)$$

In the next section, we will analyze the decoder buffer behavior.

V. Simulations of Decoder Buffer Based on the PDF of the AAL5 PDU Interarrival Time

Decoder buffer usually operates at the same clock frequency as the encoder so that decoder buffer can avoid underflowing and overflowing. The system clock in the decoder is recovered from PCRs embedded in TS packets. Jittered PCRs can cause fluctuation of the recovered STC, thus result in inaccurate DTSs which may cause decoder to underflow or overflow. In order to absorb the network introduced jitter, usually additional jitter buffer is needed. However, little has been done in the analysis of the decoder buffer behavior when the jitter is present. This problem is investigated in this section.

In our simulation, we assume that the rate of MPEG-2 video source has composite Gaussian distribution given in Eq.(1). The steady state probabilities of the Markov chain for the three states are chosen to be $p_1 = 0.344$, $p_2 = 0.194$ and $p_3 = 0.462$, respectively [12]. The mean and standard derivation are set to $\eta_1 = 37,482$ bits per frame (106.48 ATM cells or 26.62 TS packets) and $\sigma_1^2 = 2401$ bits (6.82 cells or 1.71 TS packets) per frame for state 1, $\eta_2 = 49,203$ bits (139.78 ATM cells or 34.95 TS packets) per frame and $\sigma_2^2 = 2461$ (6.99 cells or 1.75 TS packets) per frame for state 2, and $\eta_3 = 71,108$ bits (202.01 cells or 50.50 TS packets) per frame and $\sigma_3^2 = 13238$ bits (37.61 cells or 9.4 TS packets) per frame for state 3 [12]. Starting from this source model, following Eq. (9) in Section IV, and assuming the standard derivation of jitter is 0.1 ms, we can obtain the PDF of the AAL5 PDU interarrival time in the decoder side. From this, we can generate the AAL5 PDU interarrival process, a random sequence representing the time of arrival. From the bit-rate generated based on the composite Gaussian model, we also generate a random sequence that represents the number of packets in successive frames, which can provide us the buffer service time.

We simulated the decoder buffer based on the MPEG-2 Transport Stream system target decoder (T-STD), which is a hypothetical decoder (reference model). T-STD provides a formalism for timing and buffering relationship. The entire decoder buffer consists of two buffers, one is called the transport buffer (TB) and another one is called the main buffer (B) or elementary steam buffer. The decoding process is illustrated in Fig. 3. Data from an MPEG2 TS enter the T-STD at a piece-wise constant rate. The i^{th} byte of the TS, $M(i)$, enters the TB at time $t(i)$ which can be recovered from the input stream by decoding the input PCR field [1]. Each complete transport packet which has entered TB is removed instantaneously and immediately placed in buffer B at a time specified as latency following the time when one-half of the transport packet has entered B. The symbol $tb(p)$ indicates the time when the p^{th} transport packet of the TS enters B. The main buffer consists of a multiplexing buffer and a Video Buffer Verifier (VBV). For the main buffer, all the data for the j^{th} access unit, $A(j)$, is removed instantaneously at its decoding time $td(j)$. Here an access unit means the coded representation of a picture frame. The T-STD decoder shall remove the access unit data from the main buffer at the earliest time consistent with the defined decoding time and DTS or PTS value encoded in the bitstream. In the non-progressive and low-delay mode, when the buffer does not contain the complete data for an access unit at its decoding time, the buffer is re-examined at a regular interval until the complete data is present in the buffer. Packet loss occurs when the buffer is full. The so-called "picture skipping" is permitted to occur continuously without limit. Not that the decoder may be unable to reestablish correct decoding and display times until the "skipped pictures" ceases.

In our simulation, we have simulated the low-delay mode for field structure frames. In this simulation, the decoding process is as follows: we first generate packet arrivals based on the PDF of packet interarrival time, we then check whether an access unit is complete at an interval of one field period. Specifically, the time interval between two successive examinations follows

$$td(j+1) - td(j) \; = \; 1/(2R), \quad (12)$$

where R is chosen to be 30 frames/second. If all the data in an access unit are complete, then this access unit is removed immediately. Otherwise we check at an interval equal to the field time (1/2R) until all the data in this access unit are complete. The time required to remove a TS packet is derived from an assumed server bandwidth (BW), which is the speed at which the bits are removed. The service rate is determined as $\mu_{TSP} = BW/(188 \times 8)$ packets/second. The packet interarrival rate is calculated according to Eq. (11) in which the mean the mean interarrival time $E\{\mathbf{T}_{PDU,D}\}$ is estimated from the generated interarrival times. Two figures are drawn from the simulation results. Fig. 4 is the relation between the average queue size vs. server utilization. Here the server utilization is defined as $\rho = \lambda_{TSP}/\mu_{TSP}$. Different server utilizations are simulated by varying the server BW. Fig. 6 is the simulation result of the PLR vs buffer size for a given server utilization $\rho = 0.15$.

From Fig. 4, we can see that the average queue size (buffer occupancy) is varied with the server bandwidth. From Fig. 6, it can be seen that to prevent decoder buffer overflow, we have to choose the buffer size based on the server bandwidth and arrival rate. We also see that if we want the packet loss ratio in the decoder buffer to be low, we need large buffer size to accommodate packets arriving in bursts.

VI. Conclusion and Discussion

In this work, modeling and simulation are conducted for VBR traffic in the presence of ATM network jitter. To the best of our knowledge, it is the first time that joint analysis and simulation of VBR traffic and network jitter are considered. The simulation is implemented based on the decoding timing requirement of MPEG2 T-STD and the derived PDF of transport streams (TS) packet interarrival time from a video source bitrate model and an network jitter model. The analysis and simulation results show the relationship between the decoder buffer and cell loss. The tradeoff and interaction between the decoder buffer size and cell loss ratio are addressed for a given network jitter. When packet losses occurs, a feedback information may be sent to the encoder to inform the encoder to change the bitrate. Studying the decoder buffer behavior in the presence of network jitter (usually people assume that it is the same as the encoder buffer) , possibly with decoder feedback information to encoder, remains an open topic of our further research.

In this paper, we did not consider traffic shaping. When traffic shaping is considered, the CDV for a single stream can be reduced to some degree. However, the CDV due to multiplexing of multiple streams cannot be eliminated. For the traffic shaping, the key issue is to determine the Peak Cell Rate at which to perform the traffic shaping. Further research will study the impact of traffic shaping on the decoder buffer for VBR traffic in the presence of network jitter.

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Appendix A Glossary of Acronyms

AAL:	ATM adaptation layer
ABR:	Available Bit Rate
ATM:	Asynchronous Transfer Mode
BISDN:	Broadband Integrated Service Digital Network
CAC:	Connection Admission Control
CBR:	Constant Bite Rate
CDV:	Cell Delay Variation
CPCS:	Common Part Convergence Sublayer
DTS:	Decoding Time Stamps
GCRA:	Generic Cell Rate Algorithm
LAN:	Local Area Network
HDTV:	High Definition Television
LPF:	Low Pass Filter
MPEG:	Moving pictures experts Group
NPC:	Network Parameter Control
PCR:	Program Clock Reference
PDF:	Probability Density Function
PDU:	Protocol Data Unit
PDV:	AAL5 PDU Delay Variation
PES:	Packetized Elementary Stream
PLL:	Phase Locked Loop
PLR:	Packet Loss Ratio
PS:	Program Streams
PTS:	Presentation Time Stamps
QoS:	Quality of Service
SSCS:	Service Specific Convergence Sublayer
SAR:	Segmentation and Reassembly
SCR:	System Clock Reference
SDU:	Service Data Unit
STC:	System Time Clock
STD:	System Target Decoder
TS:	Transport Streams
T-STD:	Transport Streams System Target Decoder
UPC:	User Parameter Control
VBR:	Variable Bite Rate
VBV:	Video Buffer Verifier

References

[1] ISO/IEC JTC 1/SC 29/WG 11, *Generic coding of moving pictures and associate audio information, Part 1:systems, Part 2:Video, Part 3:Audio*, CD 13818, May 1995.

[2] The Grand Alliance, "The U.S. HDTV Standard", *IEEE Spectrum*, pp. 36-45, April, 1995.

[3] K. Chandra and A.R. Reibman, "Modeling two-layer MPEG2 video traffic" *Multimedia Communications and Video Coding*, Y. Wang, *et al.* eds., Plenum Press, to appear.

[4] D. Heyman, A. Tabatabai, and T.V. Lakshman, "Statistical analysis and simulation study of video teleconference traffic in ATM networks", *IEEE Trans on Circuits and Systems for Video Technology*, vol.2, no.1, March 1993.

[5] C. Hsu and A. Ortega, "Joint encoder and VBR channel optimization with buffer and leaky bucket constraints," *Multimedia Communications and Video Coding*, Y. Wang, *et al.* eds., Plenum Press, to appear.

[6] K. Joseph and D. Reininger, "Source traffic smoothing for VBR video encoders", *Proc. 6th Intl. workshop on Packet Video,* Sept.26-27, 1994, Portland, OR.

[7] R. C. Lau and P. E. Fleischer, "Synchronous techniques for timing recovery in BISDN", *IEEE Trans. on Communications*, vol.43, no.2/3/4, pp.1810-1818, Feb./March/April, 1995.

[8] D.A. Lucantoni, M. F. Neuts and A. R. Reibman, "Methods for performance evaluation of VBR video traffic models", *IEEE/ACM Trans. on Networking*, vol.2, no.2, pp.176-180, April, 1994.

[9] A. R. Reibman and B.G. Haskell "Constraints on variable bit-rate video for ATM networks," *IEEE Trans. on CAS for Video Technology*, vol.2, no.4, Dec., 1992.

[10] P. Sen, B. Maglaris, N. E. Rikli, and D. Anastassiou, "Models for packet switching of variable-bit-rate video sources," *IEEE J. Selected Areas Commun.,* vol. 7, no.5, pp.865-869, June 1989.

[11] S. A. Wright, "Service Aspects and Applications (SAA) Audio-visual Multimedia Service (AMS) implementation Agreement," *ATM Forum 95-0012(R6).*

[12] F. Yegenoglu, B. Jabbari and Y.-Q. Zhang, "Motion-classified autoregressive modeling of variable bit rate video", *IEEE Trans on Circuits and Systems for Video Technology*, pp. 42-53, vol.3, no. 1, Feb., 1993.

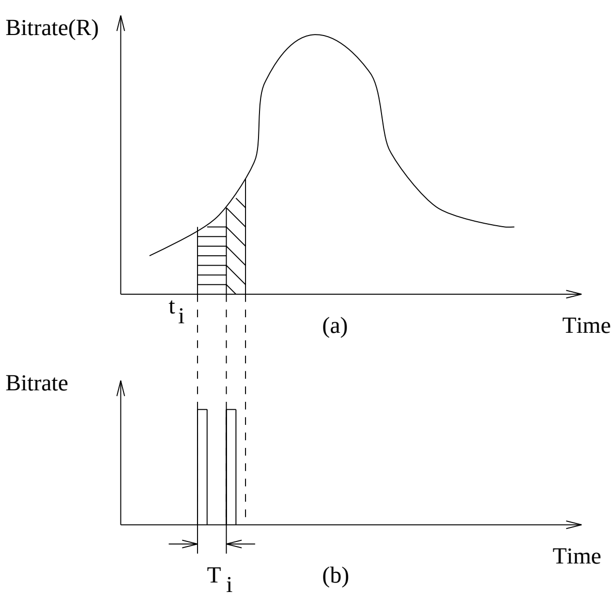

Figure 1: The process of packetization into AAL5 PDUs.

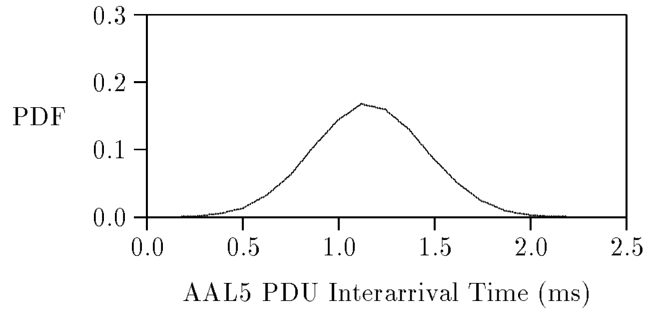

Figure 2: An example of the PDF of the AAl5 PDU interarrival time (with jitter).

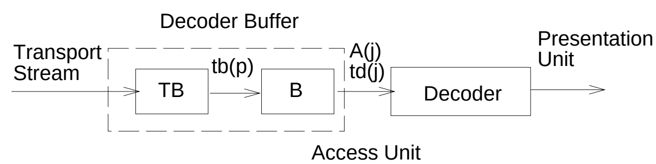

Figure 3: MPEG-2 Transport Stream system target decoder.

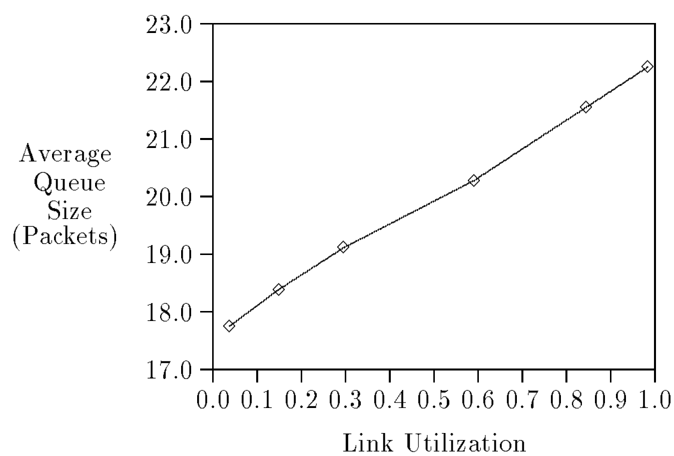

Figure 4: Simulation of average queue size vs sever utilization.

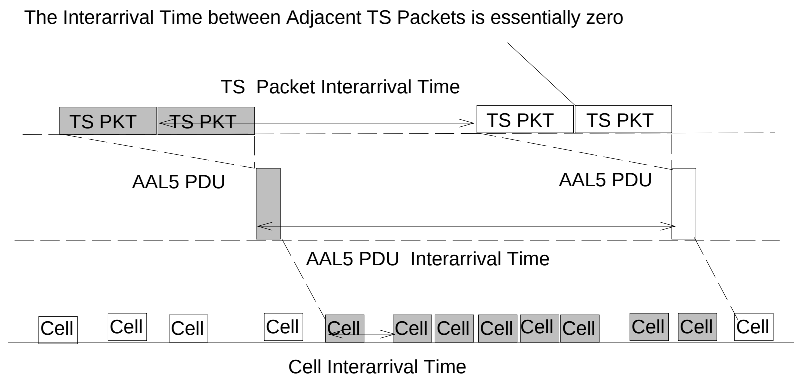

Figure 5: The process of demultiplexing into TS packets.

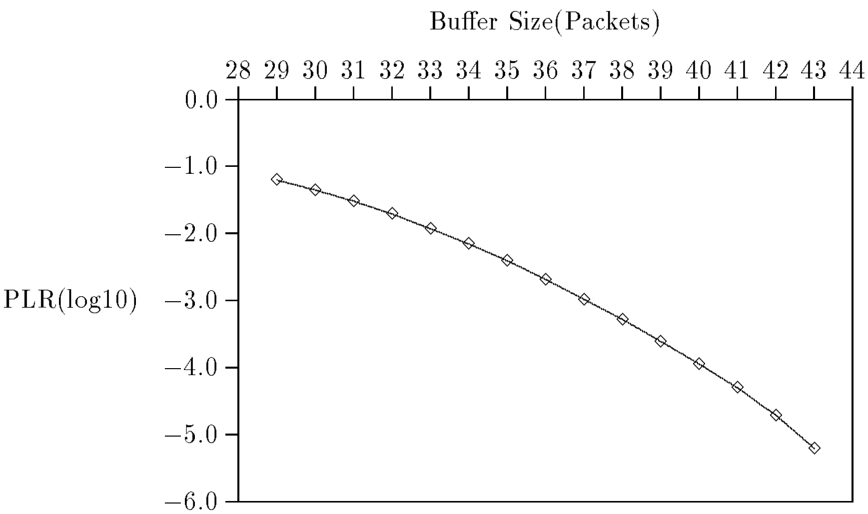

Figure 6: Simulation result for packet loss ratio vs queue size with $\rho = 0.15$.