Journal of Mobile Multimedia, Vol. 3, No. 4 (2007) 298–313

TSFD: TWO STAGE FRAME DROPPING FOR SCALABLE VIDEO TRANSMISSION OVER DATA NETWORKS

BING ZHENG
Bookham Technology
San Jose, CA 94124, USA

MOHAMMED ATIQUZZAMAN
Telecommunication and Computer Networks Lab
School of Computer Science, University of Oklahoma
Norman, OK 73019, USA
atiq@ou.edu

Received September 19, 2007
Revised October 16, 2007

Scalable video transmission is requried to transmit video over bandwidth limited channel like the Internet. However, previous scalable video transmission schemes which are based on static priority layer/slice encapsulation did not provide an algorithm to optimally choose parameters and/or required changes in the standard network protocol. The objective of this paper is to develop a new scalable video transmission scheme which can transmit stored video with low bandwidth requirement and eliminate the underflow/overflow at client to ensure QoS. The main contributions in our scheme are: parameters are chosen based on the bit rate and burstiness of video, video frames are priority encapsulated and dropped dynamically by the server and/or network depending on the network congestion, the need for decoder/encoder combination at the server is eliminated, and no major changes are required in the standard network protocol. An analytical model is developed to determine the performance and quality of service offered by our proposed scheme as the function of the network size, network congestion level, and video burstiness. Results show that our scheme requires low bandwidth as a function of network size, network congestion level and video burstiness.

Keywords: Scalable video, ATM, computer networks, multimedia

Communicated by: D. Taniar

1 Introduction

Video transmission over high speed networks is the basis of realization of client-server type interactive Multimedia Information Service (MIS) which includes video on demand, video shopping, distance education using video, home video game, etc. Different networks can be used to run MIS involving a client and a server [1, 2]. Advantages of the Asynchronous Transfer Mode (ATM) network, such as lower bit error, high speed, large trunk capacity, high statistical multiplexing gain make it very suitable for video transmission in a MIS. Of the four ATM Forum defined and standardized service types, CBR and VBR services can offer quality of service (QoS) guarantees and have simple bandwidth management while ABR service costs much less [3], since the ABR service utilizes the *variable* available bandwidth.

To transport video over networks with elastic bandwidth allocation, such as Internet and ATM-based network using the ABR service, the source must be able to change its rate and profile in response to network congestion. By scaling video bit rate, part or all the video information will be transmitted depending on the level of network congestion.

Compared with other traffic types, video is characterized by high bit rate and burstiness. Therefore, video bit rate reduction and a reasonable client buffer size are required. There are two methods to reduce video bit rate with acceptable quality. The static method is to reduce the bit rate regardless of network congestion. This results in low network bandwidth utilization and high cost. One of the dynamic methods, known as the scalable method, reduces the video bit rate depending on the congestion of the network. This method has two advantages: reducing the video bit rate dynamically and utilizing network bandwidth efficiently [1, 4, 5]. Therefore, the scalable method is much more suitable for video transmission.

A buffer is an integral part of a video client because of the bursty nature of video traffic. Too small a buffer results in buffer overflow; a large buffer results in large queuing delay and jitter. Both will degrade picture quality. Moreover, MPEG video has several picture structure with varying level of burstness in traffic. Therefore, it is critical to study the buffer dimensioning characteristic for different MPEG picture structures with different scaling before networked multimedia information system can be widely deployed.

Video transmission requires a large bandwidth from the network. During periods of network congestion, the bit rate of video needs to be scaled dynamically depending on the level of network congestion [6, 5, 7, 8, 9, 10, 11]. There has been interest in scalable video transmission over wireless networks [12, 13] and peer-to-peer networks[14]. Efforts in exploiting scalability features of coding schemes have been reported in [15]. Network parameters required to run interactive video over ATM Available Bit Rate (ABR) service has been investigated by Zheng et al. [6]. Authors in [1] investigated layer based scalable MPEG video transmission scheme over ATM networks. However, the previous schemes have to be implemented at the *video source coder*. The authors in [5, 16] studied static layer based and slice based scalable MPEG transmission over the ATM VBR/ABR hybrid service. Unfortunately, their scheme needs modification of the the standard AAL5 protocol, and hence is not suitable for practical deployment. Moreover, they did not provide any algorithm to choose an optimal value of Minimum Cell Rate (MCR) to ensure QoS. The *objective of this paper* is to develop a scalable scheme for transmitting stored MPEG video over a bandwidth-limited channel without requiring major changes of the standard network protocols.

From the above study, it is clear that several problems need to be solved before the widespread commercial deployment of MIS. First, realizing scalable transmission without the requirement of expensive hardware or major changes in the standard network protocols. Second, selecting the MCR value when an ABR connection is set up. Third, setting the client buffer to ensure QoS. Fourth, determining the effect of MPEG video structure on the client buffer requirement and QoS.

We propose a novel Two Stage Frame Dropping (TSFD) Scheme for scalable MPEG transmission over an ATM ABR service. In addition to frames being dropped by the server in the case of network congestion, the server also marks low priority frames to be dropped by the network in the case of severe congestion. An important *contribution* of this paper is the choice of MCR *as a combination of the bit rate and burstiness of video*. Our proposed TFSD

f	MPEG video frame rate in frames/second;
n	Distance between I frames for MmNn GoP;
m	Distance between P frames for MmNn GoP;
X_I, X_P, X_B	Average size of I, P and B frames in bits;
x_I	I frame size in bits;
β_I	Average bit rate of I frames defined as $X_I f$;
β_P	Average bit rate of P frames defined as $X_P f$;
β_B	Average bit rate of B frames defined as $X_B f$;
$E_0[\beta]$	Average bit rate of video stream with I, P and B frames;
$E_1[\beta]$	Average bit rate for video stream with I and P frames;
$E_2[\beta]$	Average bit rate for video stream with only I frames;
k	Speed factor for fastforward / fastbackward (FFW/FBW) operation in a video on demand system;
ACR, MCR	The available and minimum cell rates respectively during playback;
C_c	Critical value of client buffer size;
ρ	The ratio of MCR to $E_2[\beta]$;
T_d	Fixed Round Trip Time (FRTT) from server to client;
τ_f	Duration of FFW/FBW operation;
τ_c	Duration of network congestion;
$b = \frac{\beta_I}{\beta_B}$	Burst coefficient for MPEG;
n_1, n_2	Expected number of requests before the server gets the FFW/FBW and playback bandwidth respectively.

Fig. 1. Modeling Parameters.

scheme is based on encapsulating video frames with priority information which is used to drop frames by the network during congestion. The scheme requires *no major change of network protocols*. The effect of MPEG video GoP on the client buffer size has also been analyzed. A general framework has been developed to determine the client buffer size for no overflow at the client.

The rest of this paper is organized as follows. The principle of TSFD is presented in Section 2, followed by Section 3 where we develop an analytical framework to determine the optimal client buffer size. The analytical modeling framework has been used to obtain the numerical results presented in Section 5. Concluding remarks are given in Section 6.

2 Two Stage Frame Dropping (TSFD)

The *Two Stage Frame Dropping* (TSFD) scheme proposed in this paper consists of two parts. The first is the dynamic priority encapsulation and frame discarding procedure under different network congestion. The second is the adaptive choice of MCR by taking into consideration the burstiness and bit rate of video. Notations used throughout the paper are given in Fig. 1.

2.1 Principle of TSFD

Loss of I, P, or B frames have different effects on video quality. I frame is the reference frame which is most important, P and B frames are less important because they contribute mainly to the improvement of space and time resolution. Therefore, in case of network congestion, B and P frames can be discarded to reduce the bandwidth requirement of video. As shown in

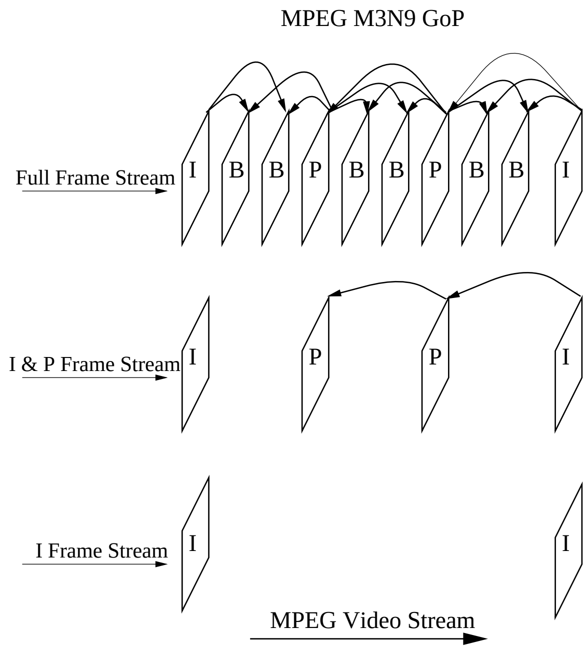

Fig. 2. Three video streams having different combinations of frame types.

Figure 2, different combination of frame types require different amounts of bandwidth. Our proposed TSFD scheme can be described as follows:

- During normal playback, frames are dropped either by the server and/or by the network depending on the level of congestion.
- The assignment of the Cell Loss Priority (CLP) bit (in the ATM cell header) by the server is done as follows. If the bandwidth offered by the network meets the requirements for transporting full frame video stream, the server does not drop frames, and sets CLP=1 for B frames (Figure 3). However, if the bandwidth offered by the network meets the requirements for transporting only I and P frames, the server discards the B frames, and assigns CLP=1 to P frames. If the bandwidth offered by the network just

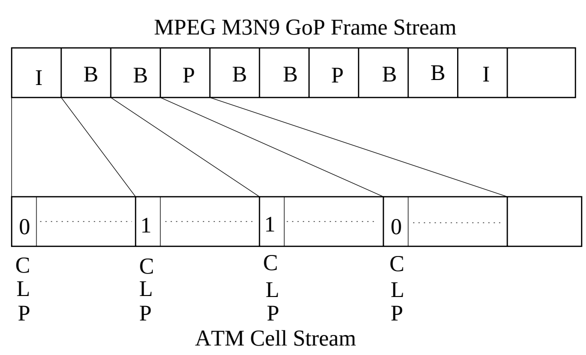

Fig. 3. Priority encapsulation in the TSFD scheme.

meets the requirements for transporting I frames, the server discards B and P frames; it randomly sets some of the I frames with CLP=1 allowing the network to drop those I frames in the case of congestion.

- In the FFW or FBW mode, the server discards all B and P frames, and sends only I frames to reduce the bandwidth.
- During FFW/FBW operation, the server accepts whatever rate is available from the network, but keeps sending in rate RM cells to request a rate which is no less than the required rate for FFW/FBW operation.

2.2 Adaptive Choice of MCR

Once an ABR connection is set up, any bandwidth request which is higher than the MCR is approved by the network with some probability. We describe below an algorithm to adaptively choose a value of MCR. From Figure 2, the average bit rate corresponding to the three frame streams can be expressed as:

$$E_0[\beta] = \frac{\beta_I + \beta_P(n/m - 1) + \beta_B n/m(m-1)}{n} \tag{1}$$

$$E_1[\beta] = \frac{\beta_I + \beta_P(n/m - 1)}{n} \tag{2}$$

$$E_2[\beta] = \frac{\beta_I}{n} \tag{3}$$

To ensure adequate quality of video at the client, the value of MCR should be between $E_2[\beta]$ and $E_0[\beta]$, and is given by

$$MCR = \rho E_2[\beta] \tag{4}$$

where ρ has three different values corresponding to the three different streams.

$$\rho_{low} = 1 \tag{5}$$

$$\rho_{middle} = 1 + \frac{1}{b}\frac{\beta_P}{\beta_B}\left(\frac{n}{m} - 1\right) \tag{6}$$

$$\rho_{high} = 1 + \frac{1}{b}\frac{\beta_P}{\beta_B}\left(\frac{n}{m} - 1\right) + \frac{1}{b}\frac{n}{m}(m-1) \tag{7}$$

ρ reflects the contributions from the B and P frames, and hence is a measure of the burstiness of video. The algorithm to choose MCR is shown in Figure 4.

3 Client Buffer Size

The previous section has described the two components of our proposed *Two Stage Frame Dropping* (TSFD) scheme. The buffer size at the client plays a critical role in the overall quality of the video. Too small a buffer size results in dropping of packets and thus poor quality of received video. Too large a buffer results in unnecessary cost at the client. In this section, we describe the optimum buffer size required at the client for our proposed scheme.

We define the *critical client buffer size* C_c as the size below which the client buffer overflows with a high probability. To determine C_c, we first need to determine the *minimum client buffer fill level*, which is defined as the buffer level at which there is no starvation at the client.

BEGIN *Connection setup*
Set $MCR = E_0[\beta]/\rho_{low}$
Send ABR Connection Setup Signal
If *Returned MCR* $<$ *MCR*, then Set $MCR = E_0[\beta]/\rho_{middle}$
Send ABR Connection Setup Signal
If *Returned MCR* $<$ *MCR*, then Set $MCR = E_0[\beta]/\rho_{high}$
Send ABR Connection Setup Signal
If *Returned MCR* $<$ *MCR*, then Wait a random time period and try again
END

Fig. 4. The algorithm for choosing MCR.

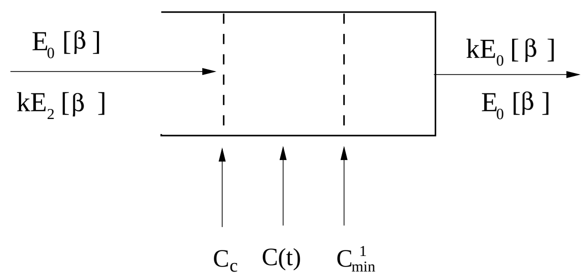

Fig. 5. Illustration of C^1_{min} and C_c at client buffer.

3.1 Minimum Client Buffer Fill Level

The value of the MCR that is accepted by the network may be lower than the average rate of the video which gives rise to the following two cases corresponding to normal operation and FFW/FBW operation, the latter case representing consumption of data from buffer at a high rate and thus a high possibility of buffer overflow.

- *Case 1:* In FFW/FBW operation, there should be no starvation at the client buffer;
- *Case 2:* In normal playback, there should be no starvation at the client buffer during network congestion.

3.1.1 Case 1: FFW/FBW Operation

As shown in Figure 5, we assume that at time t, the client sends a FFW/FBW request to the video server. The client starts its FFW/FBW operation, consuming video data at a rate $kE_0[\beta]$. We also assume that FFW/FBW lasts for a time duration τ_f. During this period, the client will consume $Q^1_{out} = \int_t^{t+\tau_f} kE_0[\beta]dt$ amount of data. On the other hand, because of network congestion and propagation delay, the client will not immediately receive data at the FFW/FBW rate after sending the FFW/FBW request. The delay consists of three parts: $T_d/2$ for the FFW/FBW request to arrive at the server, $n_1 T_d$ to obtain the requested bandwidth of $kE_2[\beta]$, and $T_d/2$ for the data to arrive at the client. During this time, the data input rate to client buffer is still at the rate of $E_0[\beta]$. The input data is denoted by

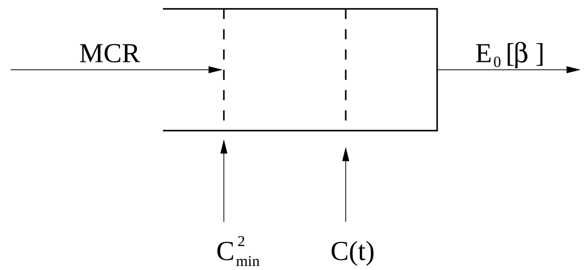

Fig. 6. Illustration of C^2_{min} at the client buffer

$Q^1_{in1} = \int_t^{t+(n_1+1)T_d} E_0[\beta]dt$. $Q^1_{in2} = \int_{t+(n_1+1)T_d}^{t+\tau_f} kE_2[\beta]dt$ is the amount of input data at FFW/FBW speed. Therefore, for no starvation at the client buffer, the amount of data consumed by the client must be equal to or less than the sum of the arriving data and the previously stored data in the buffer, i.e.,

$$C(t) + Q^1_{in1} + Q^1_{in2} - Q^1_{out} \geq 0 \tag{8}$$

where, $C(t)$ is client buffer fill level at time t. By writing Equation (8) in average value form:

$$\begin{aligned} C(t) + (n_1+1)T_dE_0[\beta] + (\tau_f - (n_1+1)T_d)kE_2[\beta] - \\ (n_1+1)T_dkE_0[\beta] - (\tau_f - (n_1+1)T_d)kE_0[\beta] \geq 0 \end{aligned} \tag{9}$$

Therefore, to prevent the client buffer from starvation during the FFW/FBW operation, the client buffer must have a minimum fill level of C^1_{min}. Note that C^1_{min} is the value of $C(t)$ for the minimum case in Equation (9).

$$\begin{aligned} C^1_{min} \geq (k-1)(n_1+1)T_dE_0[\beta] + \\ k(\tau_f - (n_1+1)T_d)(1 - \frac{1}{\rho_{high}})E_0[\beta] \end{aligned} \tag{10}$$

3.1.2 Case 2: Heavy Network Congestion

In case of heavy network congestion, $ACR = MCR$. Assume that the *congestion duration* lasts for a period of τ_c. To prevent the client buffer from underflow, a minimum fill level of C^2_{min} is required to compensate the difference between the low video data input rate and high data consumption rate from the client buffer as shown in Figure 6.

In the worst case, only after the end of heavy congestion, can the server obtain the required normal playback rate after an average of n_2 requests. During the time period $\tau_c + (n_2 + 0.5)T_d$, the amount of input data to the client buffer is

$$Q^2_{in} = \int_t^{t+\tau_c+(n_2+0.5)T_d} E_2[\beta]dt \tag{11}$$

and the amount of data consumed by the client from the buffer is

$$Q^2_{out} = \int_t^{t+\tau_c+(n_2+0.5)T_d} E_0[\beta]dt. \tag{12}$$

Therefore:

$$C(t) + Q_{in}^2 - Q_{out}^2 \geq 0 \tag{13}$$

C_{min}^2 and C_{min} are obtained from the following conditions:

$$C_{min}^2 \geq \left(1 - \frac{1}{\rho_{high}}\right)(\tau_c + (n_2 + 1)T_d)E_0[\beta] \tag{14}$$

$$C_{min} = \max(C_{min}^1, C_{min}^2) \tag{15}$$

It can be seen that the *minimum client buffer fill is directly related to the network congestion, video rate, and the FRTT of the channel*.

3.2 Critical Client Buffer Size

As illustrated in Figure 5, after the FFW/FBW operation is finished, the client will send a signal to the server to restore its outgoing data to normal playback rate. It takes $T_d/2$ for the server to receive the signal and another $T_d/2$ for the client to receive the data sent from the server at the normal playback rate. So, C_f is buffer space required by the client to accumulate this fluctuation.

$$C_f = T_d\left(\frac{k}{\rho_{high}} - 1\right)E_0[\beta] \tag{16}$$

From Equations (10), (14), (15) and (16), the critical client buffer size C_c at the client is:

$$C_c = C_f + C_{min} \tag{17}$$

The client buffer overflow probability can be estimated from the critical client buffer size requirement.

4 Quality of Picture

The quality of picture (QoP) at the client is directly related to the size of the client buffer. If the client buffer has video data up to the minimum fill level, the client can compensate slow data-in and fast data-out. From Equation (14), the maximum time τ_c that the client can tolerate in network congestion is given by:

$$\tau_c = \frac{C_{min}}{(\rho_{high} - 1)E_2[\beta]} - (n_2 + 1)T_d \tag{18}$$

On the other hand, with a given client buffer size, the burst nature of a dedicated video cause overflow in client buffer. The burstness mainly comes from the burstness of the I frame size x_I. Based on statistical characteristic of MPEG video, the distribution of I frame size can be modeled by Gamma pdf(probability density function) [17]:

$$p(x_I) = \frac{\lambda^\alpha x_I^{\alpha-1} e^{-\lambda x_I}}{\Gamma(\alpha)} \tag{19}$$

where $\Gamma(\alpha) = \int_0^\alpha x^{\alpha-1}e^{-x}dx$ is Gamma function.

By mapping the pdf of the I frame size distribution to the client buffer fill level distribution with the combination of Equations (19) and (17), the pdf of client buffer fill level can be expressed as:

$$p(C) = \frac{1}{\theta} \frac{\lambda^{\alpha}(\frac{C}{\theta})^{\alpha-1} e^{-\lambda \frac{C}{\theta}}}{\Gamma(\alpha)} \tag{20}$$

$$\theta = f \frac{((k - \rho_{high})(n_1 + 2)T_d + \tau_f k(\rho_{high} - 1))}{n} \tag{21}$$

The client buffer fill level also has a Gamma distribution. The overflow probability $P_{ov}(C)$ is the probability that the client buffer fill level exceeds a given client buffer size C:

$$P_{ov}(C) = 1 - F(C) \tag{22}$$

where $F(C) = \int_0^C p(c)dc$ is the probability distribution function for a fill level C.

5 Results and Discussion

In this section, we measure the performance of our proposed scheme in terms of buffer size requirement as a function of the round trip time, FFW/FBW duration, and duration of network congestion. The models developed in Sec. 3 have been used to derive the numerical results presented in this section.

The parameters chosen for the performance model used in the calculations are as follows. An MPEG video stream with typical value $X_I = 400$ kbits, $X_P = 200$ kbits and $X_B = 80$ kbits [18] is used to get the critical client buffer size C_c for different MPEG GoPs. A MPEG sequence *dino* [17] with following parameters is employed to study the QoS performance.

- *GoP Pattern:* M3N12 IBBPBBPBBPBB;
- *Frame Rate:* 24 frames/second;
- *Quantizer Scale:* 10 for I frame, 14 for P frame and 18 for B frame;
- *Resolution:*384*288 pels, 12 bit color information;
- *Mean Frame Size:* 13078 bits;
- *Burst Coefficient b:* ≥ 9.1;
- *Peak Bit Rate:* 1.01 Mbps;
- *Mean Bit Rat:* 0.33 Mbps.

Figure 7 shows the average bit rate for sequence *dino* corresponding to TSFD scheme with B frame dropping, layer based priority encapsulation with P enhancement, and layer based priority encapsulation with B enhancement. TSFD scheme requires the lowest bandwidth.

The critical client buffer size C_c versus FRTT is shown in Figure 8 for different GoP. As expressed in Equations (10), (14), (15), (16) and (17), the client buffer size depends on FFW/FBW time τ_f, video parameter ρ_{high} and $E_0[\beta]$, and FFW/FBW speed factor k. So, the T_d has different effect on the critical client buffer size for different GoP. High GoP pattern needs a larger buffer size than low GoP pattern since it gets more contribution from B and P frames.

The critical client buffer size C_c versus the FFW/FBW duration with a constant FRTT is shown in Figure 9. As the FFW/FBW time increases, the buffer size increases linearly for all

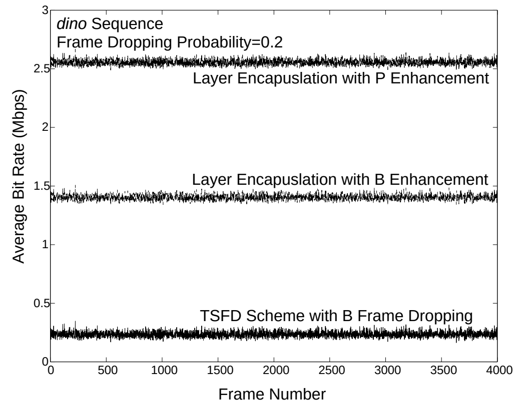

Fig. 7. Average bit rate corresponding to three encapsulating scheme for sequence *dino*.

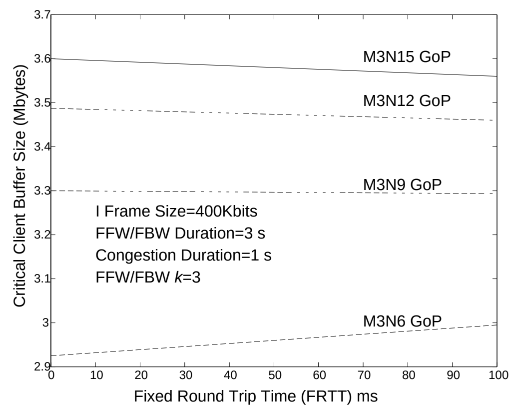

Fig. 8. Critical Client buffer size C_c versus fixed round trip time (FRTT).

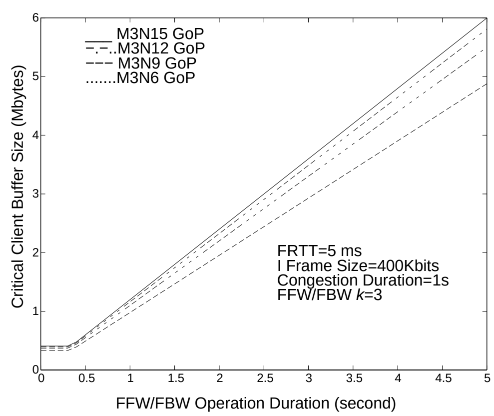

Fig. 9. Critical Client buffer size C_c versus FFW/FBW time.

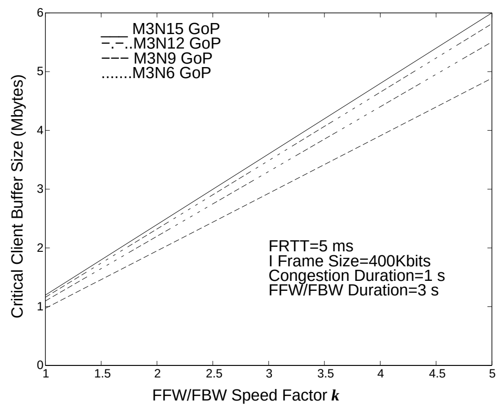

Fig. 10. Critical Client buffer size C_c versus FFW/FBW speed factor k.

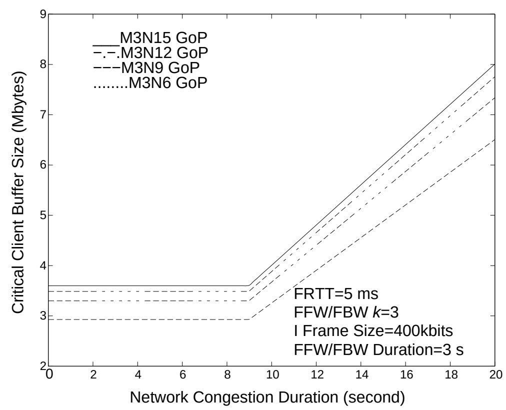

Fig. 11. Critical Client buffer size C_c versus network congestion duration.

MPEG GoP structures. This is because the longer the FFW/FBW operation, the higher the data consumed by the client. Higher GoP pattern has a larger ρ_{high} which implies a higher data rate. So, the critical client buffer size for a higher GoP pattern has a larger increasing slope. Note that the fixed part at the beginning is coming from the effect of Equation (16).

Figure 10 shows the critical client buffer size as a function of FFW/FBW speed factor k. As expected from Equation (17), the critical client buffer size increases linearly with the FFW/FBW speed factor. Similarly, because a higher GoP pattern has a larger ρ_{high} and data rate, it also has a larger slope.

Figure 11 shows the critical client buffer size as a function of the network congestion duration. When network congestion lasts for a period of time shorter than nine seconds, the critical client buffer size remains constant. Only when the congestion time become longer than nine seconds, the critical client buffer size increases linearly with the congestion time.

Figure 12 shows the relationship of critical client buffer with the burst coefficient of video. It is seen that TSFD has the advantage that the client buffer requirement is immune to the

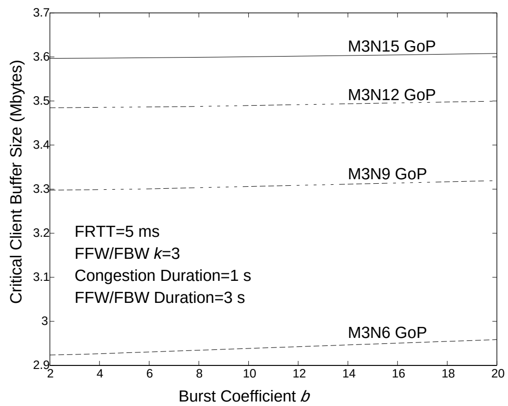

Fig. 12. Critical Client buffer size C_c versus Burst Coefficient.

video burst for different video types. The critical client buffer size almost stays constant when video burst increase. Therefore, our scheme is suitable for a wide variety of video types with different levels of burstiness.

Figures 13 and 14 show the critical client buffer versus the expected request times for the server to get the FFW/FBW bandwidth and the playback bandwidth. It is found that the critical client buffer is insensitive to the request time, that means the critical client buffer size decided by our TSFD scheme can tolerate a relative heavy network congestion while keeps the client no starvation happening. The TSFD scheme has a good congestion immunity characteristic.

Figure 15 shows the pdf of client buffer fill level for sequence *dino* with different FFW/FBW speed factor. When FFW/FBW speed factor increase, the client buffer fill level moves toward a large value, and the distribution of client buffer fill level becomes flat, resulting in a larger critical client buffer size.

The client buffer overflow probability versus the client buffer size for different FFW/FBW speed factor for sequence *dino* is shown in Figure 16. Since the critical client buffer requirement increases and the fill level becomes flatter, the risk of overflow for a given client buffer size increases. Therefore, the FFW/FBW is an important factor for potential overflow at client. However, it is found that the ratio of the client buffer size with no overflow to the critical buffer size is constant for different FFW/FBW. So, when the client buffer size is set to the value of greater or equal to three times of the critical buffer size, the overflow probability at client is almost zero for given FFW/FBW speed factor.

Figure 17 shows the pdf of client buffer fill level for sequence *dino* with different FRTT. It is found that the FRTT effect is not obvious; with very large FRTT change, the distribution of client buffer fill level only changes a little.

The client buffer overflow probability versus the client buffer size for different FRTT for sequence *dino* is shown in Figure 18. The overflow characteristic is insensitive to the FRTT or the network size. Therefore, TSFD scheme is suitable for networks with varying sizes.

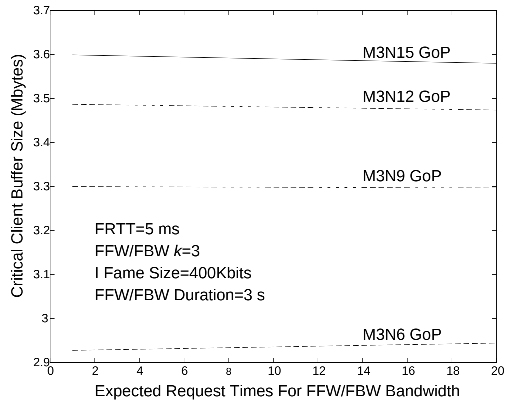

Fig. 13. Critical Client buffer size C_c versus expected request times for FFW/FBW bandwidth.

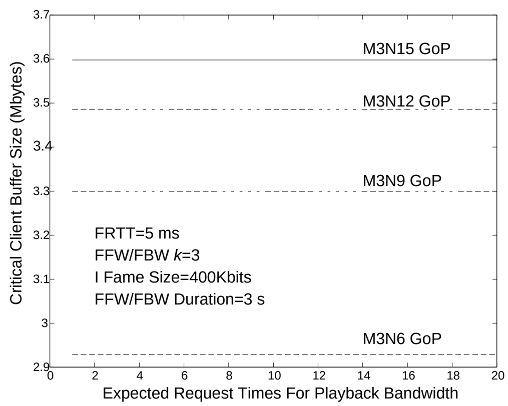

Fig. 14. Critical Client buffer size C_c versus expected request times for playback bandwidth.

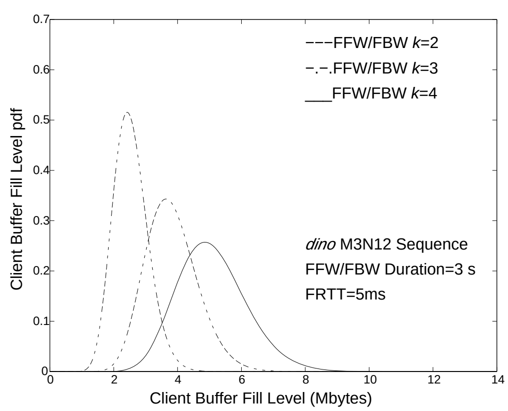

Fig. 15. pdf for client buffer fill level with different FFW/FBW speed factor.

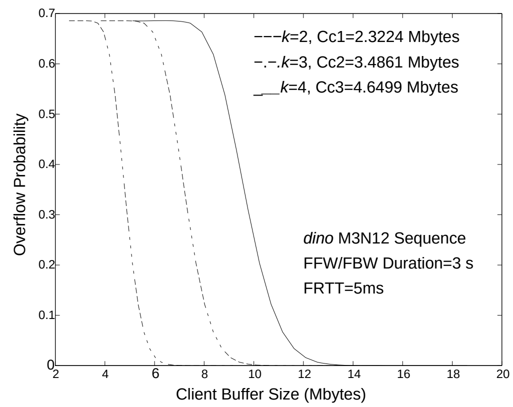

Fig. 16. Overflow probability versus client buffer size with different FFW/FBW speed factor.

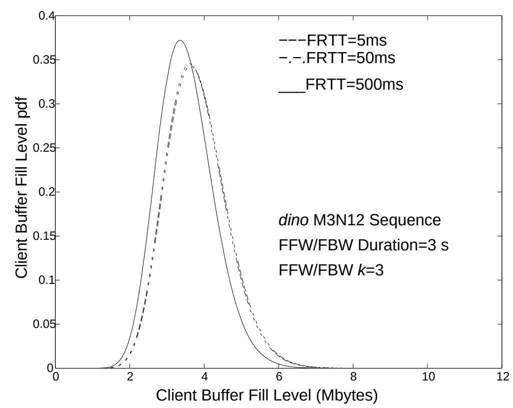

Fig. 17. pdf for client buffer fill level with different FRTT.

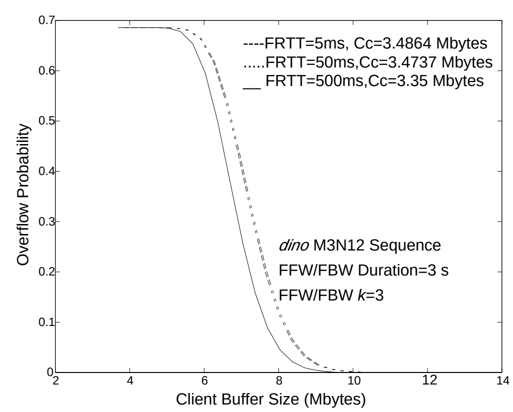

Fig. 18. Overflow probability versus client buffer size with different FRTT.

6 Conclusion

In this paper, we proposed a Two Stage Frame Dropping (TSFD) for scalable MPEG transmission over an ATM ABR service. Our scheme adaptively sets the value of Minimum Cell Rate, and also adjusts the video rate dynamically depending on network congestion. We have developed a statistical model to determine the client buffer size. We conclude that the client buffer size has a linear relationship with the FFW/FBW operation time and the FFW/FBW speed factor. The client buffer size increases linearly with long term network congestion, and is insensitive to network size, burstiness of the video, and short term network congestion. We have shown that the client buffer requirement depends on the MPEG GoP structure; usually, a large GoP pattern requires a large client buffer. The results in this paper can be used by system and network designers to determine the optimal buffer size and fine tune the network parameters to allow video on demand systems over the ATM ABR service.

References

1. Wenjun Luo and Magda El Zarki, "Quality control for VBR video over ATM networks," *IEEE Journal on Selected areas in Communications*, vol. 15, no. 6, pp. 1029–1039, August 1997.
2. Hayder Radha, Yingwei Chen, Kavitha Parthasarathy, and Robert Cohen, "Scalable internet video using MPEG-4," *Signal Processing: Image Communication*, vol. 15, no. 1, no. 1, pp. 95–126, 1999.
3. Lawrence G. Roberts, "Can ABR service replace VBR service in ATM network," *COMPCON'95 Conference*, Piscatway, New Jersey, pp. 346–348, 1995.
4. M. Andronico, A. Lombardo, S. Palazzo, and G. Schembra, "Performance analysis of priority encoding transmission of MPEG video streams," *IEEE GLOBECOM'96*, London, pp. 267–271, Nov 1996.
5. Ahmed Mehaoua and Raouf Boutaba, "Performance analysis of cell discarding techniques for best effort video communications over ATM networks," *Computer Networks and ISDN Systems*, vol. 29, no. 17-18, pp. 1999–2019, February 1998.
6. B. Zheng and M. Atiquzzaman, "System design and network requirements for interactive multimedia," *IEEE Transactions On Circuits And Systems For Video Technology*, vol. 15, no. 1, pp. 145–153, January 2005.
7. S. Xiao, C. Wu, and J. Jeong, "Robust video communication based on source modeling and network congestion control," *Advanced Information Networking and Applications*, pp. 488–491, Mar 28-30, 2005.
8. J. Yan, K. Katrinis, M. May, and B. Plattner, "Media- and TCP-friendly congestion control for scalable video streams," *IEEE Transactions on Multimedia*, vol. 8, no. 2, no. 2, pp. 196 – 206, 2006.
9. P.B. Ssesanga and M.K. Mandal, "Efficient congestion control for streaming scalable video over unreliable IP networks," *International Conference on Image Processing*, Singapore, pp. 2059 – 2062, Oct 18-21, 2004.
10. H.-F. Hsiao and J. Hwang, "A max-min fairness congestion control for layered streaming of scalable video," *IEEE Transactions on Circuits and Systems for Video Technology*, vol. 16, no. 9, no. 9, pp. 1074 – 1085, 2006.
11. Panagiotis Papadimitriou and Vassilis Tsaoussidis, "SSVP: A congestion control scheme for real-time video streaming," *Computer Networks*, vol. 51, no. 15, no. 15, pp. 4377 – 4395, 2007.
12. Hung-Hui Juan, Hsiang-Chun Huang, ChingYao Huang, and Tihao Chiang, "Scalable video streaming over mobile WiMAX," *Proceedings - IEEE International Symposium on Circuits and Systems*, New Orleans, LA, pp. 3463 – 3466, 2007.
13. Hsiao-Chiang Chuang, ChingYao Huang, and Tihao Chiang, "Content-aware adaptive media playout controls for wireless video streaming," *IEEE Transactions on Multimedia*, vol. 9, no. 6, no. 6, pp. 1273 – 1283, 2007.

14. Chen Zhijia, Yin Hao, Lin Chuang, and Ai Lu, "3d-wavelet based secure and scalable media streaming in a central-controlled P2P framework," *Proceedings - International Conference on Advanced Information Networking and Applications, AINA*, Niagara Falls, ON,, pp. 708 – 715, 2007.
15. Dieu Thanh Nguyen and Joern Ostermann, "Congestion control for scalable video streaming using the scalability extension of H.264/AVC," *IEEE Journal on Selected Topics in Signal Processing*, vol. 1, no. 2, no. 2, pp. 246 – 253, 2007.
16. A. Mehaoua, R. Boutaba, and G. Pujolle, "An extended priority data partition scheme for MPEG video connection over ATM," *IEEE Symposium on Computers and Communication'98*, Alexandria, Egypt, pp. 62–67, July 1998.
17. O. Rose, "Statistical properties of MPEG video traffic and their impact on traffic modeling in ATM system," research report, Institute of Computer Science, University of Wurzburg, Feb 1995.
18. O. Rose, "http://nero.informatik.uni-wuerzburg.de/mpeg/traces," March, 1995.