Journal of High Speed Networks 15 (2006) 329–340
IOS Press

Integrated traffic modeling and frame skipping for pre-stored streaming videos over cellular networks

Chu-Chuan Lee [a,b,*] and Pao-Chi Chang [a]
[a] *Department of Electrical Engineering, National Central University, Taiwan 320, Taiwan, R.O.C.*
[b] *Chunghwa Telecommunication Laboratories, Taiwan 326, Taiwan, R.O.C.*
Tel.: +886 3 4245661; Fax: +886 3 4244888;
E-mail: chuchu@cht.com.tw, secondly active address: cclee@roger.ee.ncu.edu.tw

Abstract. With the advance of computing technology, video compression technology, and cellular network infrastructure, rich multimedia applications are dramatically boosted to users. Currently, a deterministic service model has been developed in which a QoS aware network assisted by the deterministic traffic modeling can provide bounded delay and loss free guarantees to video packets. However, the limited memory capacity and access bandwidth of mobile terminals still affect the delivery of pre-stored video contents. Besides, from the cellular network viewpoint, any halfway request for increasing bandwidth may complicate the resource management. Therefore, this paper proposes an Elastically Deterministic Video Traffic Regulator (ED-VTR) to regulate the video traffic, and then generate a series of monotonically decreasing bandwidth demands which effectively simplify the resource management and satisfy the decoder buffer limitation. Moreover, for preventing possible non-sustained playback due to insufficient decoder buffer space, this study proposes an intelligent video frame skip algorithm to determine the most suitable temporal range for skipping frames. Simulation results reveal that ED-VTR effectively reduces the initial bandwidth demand and the requirement of decoder buffer space, while still maintaining advantages of deterministic services. More importantly, ED-VTR maintains the sustained playback with good picture quality and outperforms traditional schemes in the case of insufficient decoder buffer space.

Keywords: Frame skipping, cellular network, traffic modeling

1. Introduction

With the advance of cellular network infrastructure, computing technology, and video compression technology, rich multimedia services have dramatically boosted to mobile clients. Regarding video applications, they can be classified into three categories: pre-stored, live broadcast and real-time interactive videos. The pre-stored video content, such as the streaming movie, is encoded in advance and part of the compressed video data is pre-loaded to the decoder buffer before playback. Consequently, a sufficient decoder buffer space is required to maintain the stable playback [1–3]. However, regardless of desktop or wireless clients, insufficient storage space is always a problem in providing more advanced applications if the filing management of stored personal data and software programs is improper. In particular, the embedded memory of cellular clients is usually limited and needed to be shared by numerous software programs and masses of personal data, such as Java games, video data, multimedia messages, photographs, images and ringing tones in gallery, calendar, to-do notes, phone book, and so on. For example, an embedded or downloaded Java game may occupy 500 kbytes to 2 Mbytes memory space and the mass of personal photographs may occupy significant memory capacity [4,5]. This is especially true with heavy use of any of the features, even when a top mobile terminal is used. Moreover, a top mobile terminal is usually associated with advanced applications that require more memory space than regular services. Therefore, the problem of insufficient memory space exists at mobile terminals.

*Corresponding author. No. 12, Lane 551, Min-Tsu Road Sec. 5, Yang-Mei, Taiwan 32617, R.O.C.

0926-6801/06/$17.00

Regarding the bandwidth utilization of cellular networks, the overall network utilization drops if a source requests more bandwidth than it actually uses. Conversely, packet loss and delay will become significant if the source exceeds its bandwidth request. In contrast to wired line environments, the resource management for a halfway increasing bandwidth request in cellular networks does not only consider the bandwidth re-allocation issues, but also consider at least the power control, signal-to-noise ratio, and possible handoff events [6–8]. Obviously, pre-stored video applications over cellular networks face many technical challenges which are significantly different from the problems typically encountered in the wired line desktop environment [9]. In light of the inherent burstiness of Variable Bit Rate (VBR) video applications, techniques for reducing its rate variability are of significant interest.

Generally, there are two transmission models for video applications, namely the stochastic [10,11] and deterministic service classes [12,13]. A deterministic service model is one in which a QoS aware network assisted by the deterministic traffic modeling can provide the bounded delay and loss free guarantees to packets [13,14]. The major difference between these two service classes is that the former utilizes statistical properties to achieve high bandwidth utilization, while the latter utilizes deterministic bandwidth reservation to obtain QoS guarantees. Deterministic approaches have the advantage of being parameterized easily, making their implementation more practical than that of stochastic approaches. Currently, there are many related researches in this area. Grossglauser et al. [15] proposed a Renegotiated Constant Bit Rate (RCBR) mechanism, where the video traffic was described by a *transmission schedule* that comprised a series of bandwidth requests. Soleimanipour et al. [16] explored the resource management of WCDMA for wireless multimedia. The capability of bandwidth provision was limited obviously by the cellular environment. Wrege et al. [13] proposed an Empirical Envelope Modeling (EEM) scheme to characterize the video traffic. This scheme used the worst-case description to get an empirical-envelope function E^* from the cumulative transmitted data, and then generated a transmission schedule from E^* using the piece-wise linear concave upper approximation. Chang [12,17] developed a filtering theory for deterministic traffic regulation and service guarantee. A traffic regulator was implemented by a linear time-invariant filter. The stability and delay of deterministic and stochastic networks were also analyzed. Salehi et al. [18] proposed a Work-Ahead Smoothing (WAS) strategy to solve problems of transmitting pre-stored videos from a server to a client.

In this paper, an Elastically Deterministic Video Traffic Regulator (ED-VTR) that belongs to deterministic service models is developed for delivering the video data with deterministic QoS guarantees, reducing the load of resource management, and satisfying the memory capacity limitation of mobile clients. ED-VTR executes traffic regulating operations based on the current decoder buffer space and generates a series of monotonically decreasing bandwidth requirements, which can effectively simplify the resource management. Additionally, a non-sustained video playback sometimes may occur due to insufficient decoder buffer space. Rather than abruptly increasing the transmission rate during the playback or executing arbitrary video frame discarding at the decoder, ED-VTR includes an intelligent video frame skip algorithm that is executed in advance at the encoder. The algorithm can identify the most suitable temporal range for skipping frames and prevent arbitrary discarding from inappropriate video frames such as I-frames. These proposed operations are named *traffic regulation* because they not only execute the integrated traffic modeling and smoothing functions, but also execute intelligent active frame skipping operations to regulate the video traffic.

The rest of this paper is structured as follows. In Section 2, the operations of ED-VTR are presented in detail. In Section 3, the simulation environment and test functions are described. In Section 4 performances of the proposed mechanism are evaluated through simulations. Finally, Section 5 concludes this paper.

2. Elastically deterministic video traffic regulator

The proposed ED-VTR consists of two modules, one is the Deterministic Traffic Regulation (DTR) module and the other is the Intelligent video Frame Skip (IFS) module. When a mobile client requests a video streaming service, the selected video bitstream is sent to the DTR module and IFS module simultaneously. The video bitstream is first regulated by DTR based on the current decoder buffer space that is obtained in the connection negotiation stage. During the regulating period, some video frames with low significance may be skipped by IFS if the regulating result reveals that the decoder buffer overflow may happen. Finally, the regulated traffic of the selected video is modeled by a transmission schedule and sent to the network. In this section, a video streaming sequence with N frames is considered and a discrete time model with the unit of frame time is used, where the encoded bits of frame i is denoted as f_i, for $i = 1, 2, \ldots, N$. The detail operations of DTR and IFS are described as follows.

2.1. Deterministic traffic regulation

The DTR module of ED-VTR first constructs an estimative function $L(n)$ for the cumulative consumed data at the decoder, which can be expressed as

$$L(n) = \begin{cases} 0, & 0 \leqslant n \leqslant D \\ \sum_{i=1}^{n-D} \{(1+\alpha) \cdot f_i\}, & D+1 \leqslant n \leqslant D+N, \alpha \geqslant 0, \end{cases} \tag{1}$$

where D is the preloading time limited by the decoder buffer space and $n = 0$ is the time instant that the first video packet arrives to decoder. Here α is a safety factor for reducing the possibility of underflow at decoder due to the unexpected transmission delay. Since the transmission delay in deterministic service models can be bounded to a guaranteed value [13], the value α is set to zero in this paper for getting the upper bound of link utilization and coinciding with the comparison conditions to other conventional methods such as [13] and [18]. From the client viewpoint, any possible transmission schedule for the client must always be larger than or equal to $L(n)$ to avoid buffer underflow. That is, $L(n)$ is the lower bound for any possible transmission schedule. Moreover, as mentioned earlier, the decoder buffer space of mobile terminals available for video contents may be insufficient. To prevent the possible buffer overflow, this work further adds an upper bound into the process of traffic regulating. The upper bound $U(n)$ is then defined as

$$U(n) = \min\left\{\frac{L(n)}{(1+\alpha)} + B, \frac{L(D+N)}{(1+\alpha)}\right\}, \tag{2}$$

where B represents the current decoder buffer space of mobile terminals. To satisfy both bounds $L(n)$ and $U(n)$, maximize the link utilization, and include the advantages of deterministic services, DTR utilizes the piece-wise linear concave upper approximation to bound $L(n)$ directly. A raw transmission schedule $S^R(n)$ that consists of multiple piece-wise linear segments is then generated. These linear segments can be expressed as a set of traffic parameter pairs $\{(\sigma_i^R, \rho_i^R) \mid i = 1, 2, \ldots, M^R\}$, where M^R represents the maximum number of parameter pairs, ρ_i^R is the minimum requested token rate which does not cause the buffer underflow, and σ_i^R is the i-th corresponding token depth. The relationship between $S^R(n)$ and its corresponding traffic parameter pairs can be further described by

$$S^R(n) = \min_i \left(\rho_i^R \cdot n \cdot T_f + \sigma_i^R\right) \tag{3}$$

with

$$\rho_i^R = \max_{a_i < \beta \leqslant b_i} \left\{\frac{L(\beta) - L(a_i)}{\beta - a_i}\right\}, \tag{4}$$

$$\sigma_i^R = L(a_i) - \rho_i^R \times a_i, \tag{5}$$

where n is the frame number and the frame time T_f is the reciprocal of frame rate. Besides, the a_i and b_i denote the beginning and end time of the i-th linear concave upper approximation, respectively. The relationship among $L(n), U(n)$, and (σ_i^R, ρ_i^R) is shown in Fig. 1(a).

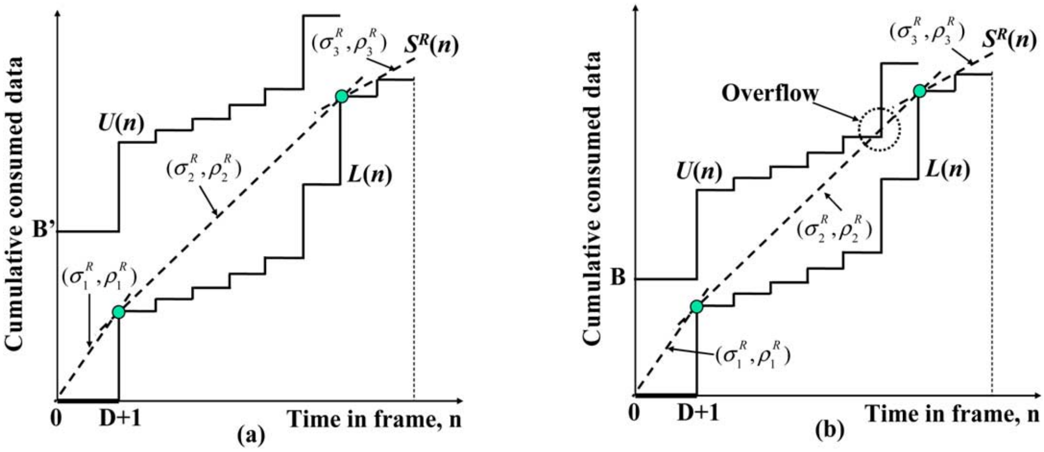

Fig. 1. Illustrations of deterministic video traffic regulation: (a) with sufficiently large buffer space B′; (b) with insufficient buffer space B.

2.2. *Intelligent video frame skip*

As mentioned in Section 1, a client with insufficient decoder buffer space may cause possible non-sustained playback. In such a situation, most traditional traffic smoothing schemes usually increase and decrease alternatively the transmission rate by a wide margin to cope with the problem. However, any halfway request for increasing the used bandwidth may complicate the resource management and cause unpredicted packet loss or latency, particularly when the available bandwidth of network is nearly exhausted. As shown in Fig. 1(b), the raw transmission schedule generated by DTR also may cause the decoder buffer overflow if the decoder buffer space is insufficient. Since the dynamic source rate control scheme is not suitable for the pre-stored video that is encoded in advance, ED-VTR adds the IFS algorithm to solve the problem while still maintaining the advantages of deterministic services.

For explanation, both $L(n)$ and $U(n)$ are redrawn by the simplified piece-wise linear segments, as shown in Fig. 2(a), where a token rate ρ_k^R is obtained after executing the k-th linear concave upper approximation for the interval $[J_k, P_k]$ of $L(n)$. In Fig. 2(a), the *concave depth* F_1 is defined as the minimum distance between the point $L(T_r)$ and the k-th linear concave upper bound. A large concave curvature is associated with a large concave depth which may cause high possibility of buffer overflow in the case of insufficient decoder buffer space. Therefore, whenever a possible decoder buffer overflow is detected by DTR, the target of IFS is to reduce the concave depth efficiently.

From Fig. 2(b), if a client has a sufficiently large decoder buffer space B' that exceeds B, no buffer overflow occurs in $[J_k, P_k]$ and the finally required token rate in $[J_k, P_k]$ is determined to be ρ_k^R directly. However, when the decoder buffer space is limited to B bits that is insufficient in this case, the possible buffer overflow in $[T_{o1}, T_{o2}]$ is detected by DTR if ρ_k^R is used. Meanwhile, IFS is activated and a new required token rate without causing the decoder buffer overflow in $[J_k, P_k]$ must be recalculated by

$$R_{\max} = \min_{J_k < \alpha \leqslant P_k} \left\{ \frac{U(\alpha) - L(J_k)}{\alpha - J_k} \right\}, \tag{6}$$

where the time instant T_r that determines the value of $R_{\max}$ can be further expressed by

$$T_r = \big\{ \tau | U(\tau) = R_{\max} \cdot (\tau - J_k) + L(J_k) \big\}, \quad \text{for } J_k < \tau \leqslant P_k. \tag{7}$$

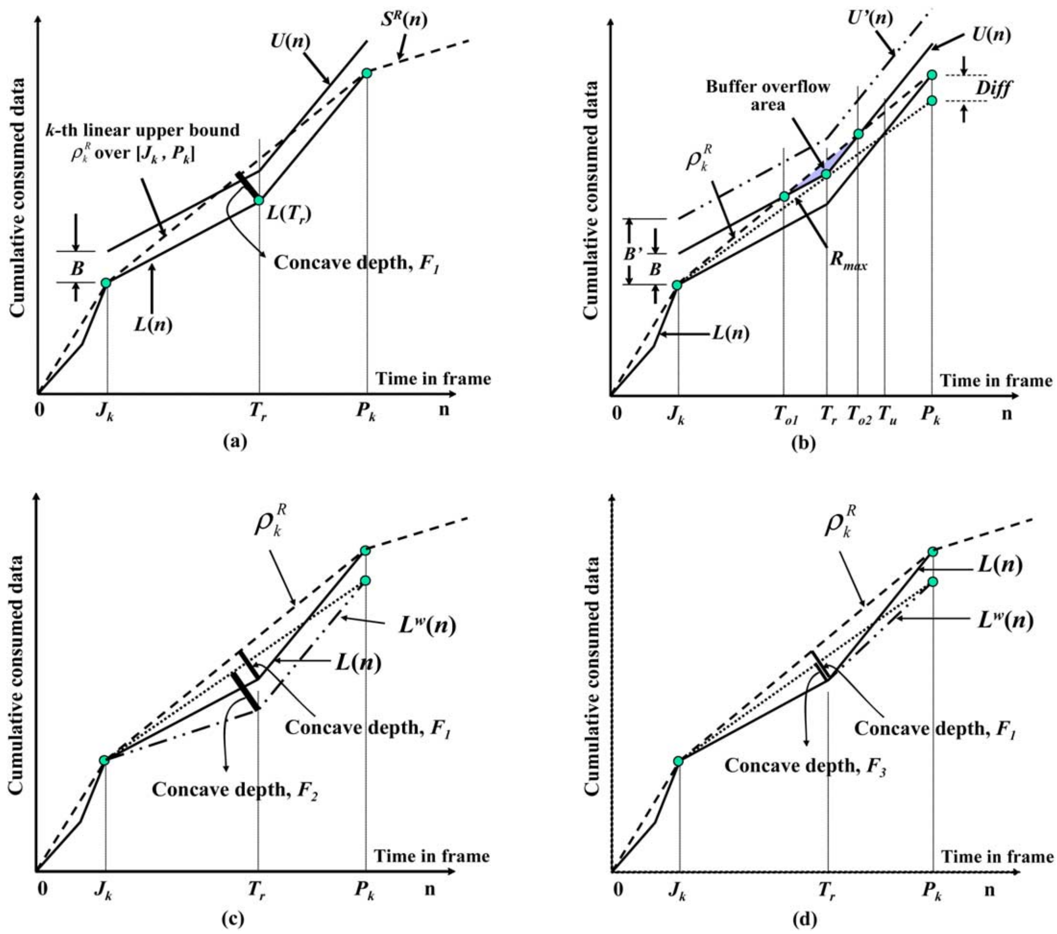

Fig. 2. Illustrations of intelligent video frame skip: (a) simplified $L(n)$ & $U(n)$ with related parameters; (b) the region of buffer overflow and underflow; (c) the case of skipping frame in $(J_k, T_r]$; (d) the case of skipping frame in $(T_r, P_k]$.

Although using $R_{\max}$ can prevent the buffer overflow, there still exists a buffer underflow in $(T_r, P_k]$. The reason is that $R_{\max}$ is less than ρ_k^R which is the minimum rate without causing buffer underflow in $[J_k, P_k]$. The first time instant T_u that causes the buffer underflow can be formulated by

$$T_u = \min_{J_k < \tau \leqslant P_k} \left\{ \tau \,\middle|\, \frac{L(\tau) - L(J_k)}{\tau - J_k} \geqslant R_{\max} \right\}. \tag{8}$$

To solve the buffer overflow problem while avoiding unnecessary underflow, skipped frame selection is very important. An inappropriate selection not only degrades the received picture quality but also worsens the above problem. Meanwhile, some video frames are needed to skip in $(J_k, P_k]$ for eliminating the difference *Diff* between the cumulative arrived data using $R_{\max}$ and the cumulative consumed data $L(P_k)$ at time P_k. The relationship among $R_{\max}$, ρ_k^R, T_r, T_u and *Diff* is presented in Fig. 2(b).

To identify a suitable temporal range for skipping video frames in $(J_k, P_k]$, IFS first considers one of the possible temporal ranges, $(J_k, T_r]$. After skipping video frames in $(J_k, T_r]$, a new cumulative consumed function $L^w(n)$ that has the concave depth F_2 in $[J_k, P_k]$ is regenerated, as shown in Fig. 2(c). In $(J_k, T_r]$, the slope of $L^w(n)$ is smaller than that of $L(n)$ because of frame skipping, but the slope of $L^w(n)$ in $(T_r, P_k]$ is the same as that of $L(n)$. From Fig. 2(c), the concave depth F_2 is larger than F_1. That is, the operation of skipping frames in $(J_k, T_r]$ cannot

PROCEDURE intelligent_frame_skip ( ) {
$R_{\max} = \min_{J_i<\alpha\leqslant P_i}\left\{\frac{U(\alpha)-L(J_i)}{\alpha-J_i}\right\};$
$\sigma_{\min} = L(J_i) - R_{\max} \times J_i;$
$T_r = \left\{\tau | U(\tau) = R_{\max}\cdot(\tau - J_i) + L(J_i)\right\};$
$Diff = \left(\rho_i^R - R_{\max}\right)\cdot P_k \cdot T_f + \left(\sigma_i^R - \sigma_{\min}\right);$
WHILE ($Diff > 0$) {
  SKIP a B-frame with size D_B in$(T_r, P_k]$;
  $Diff = Diff - D_B$;
  UPDATE $L(n)$ and $U(n)$; }
OUTPUT $L(n)$ and $U(n)$; }

Fig. 3. Intelligent frame skip algorithm.

achieve the target of reducing the concave depth but even worsen the problem. The above statements can be easily proved by simple triangle geometry. IFS then considers another possible temporal range $(T_r, P_k]$ for effectively skipping frames. Using similar derivations, it can be easily proved that the concave depth in $[J_k, P_k]$ is effectively reduced if skipped video frames are located at interval $(T_r, P_k]$, as shown in Fig. 2(d). Summarizing above results, the suitable temporal range for skipping frames in $(J_k, P_k]$ is limited to the interval $(T_r, P_k]$.

From the viewpoint of the temporal scalability in scalable video coding, skipping B-frames can minimize the influence on received picture quality. Therefore, some B-frames are actively skipped in $(T_r, P_k]$ for removing the difference *Diff*. The frame skip operation is repeated until the difference *Diff* at time P_k disappears. After accomplishing the frame skip operation in $(T_r, P_k]$, a finally required traffic parameter pair $(\sigma_k^{*R}, \rho_k^{*R})$ for $[J_k, P_k]$ is then determined and added to the mature transmission schedule $S^{*R}(n)$, where $\rho_k^{*R} = R_{\max}$ and $\sigma_k^{*R} = L(J_k) - R_{\max}\cdot J_k$. Figure 3 details the proposed frame skip algorithm.

In summary, the $S^{*R}(n)$ generated by ED-VTR has the following important properties:

(1) $S^{*R}(n)$ always stays within two bounds $L(n)$ and $U(n)$, i.e.,

$$L(n) \leqslant S^{*R}(n) \leqslant U(n). \tag{9}$$

(2) $S^{*R}(n)$ has the QoS guarantees provided by deterministic service models.
(3) The requested bandwidths generated by $S^{*R}(n)$ are always monotonically decreasing, that is

$$\rho_i^R > \rho_{i+1}^R, \quad \text{for } i = 1, 2, \ldots, M^R - 1. \tag{10}$$

As mentioned in Section 1, a deterministic service model can provide the bounded delay and loss free guarantees to packets. It is noted that most deterministic traffic modeling methods such as [13] do not consider the first property and most traffic smoothing schemes such as [18] do not consider the second and third properties of $S^{*R}(n)$ in cases of insufficient decoder buffer space. In contrast, ED-VTR not only considers the decoder buffer limitation, but also has the deterministic QoS guarantees to the regulated video traffic. More importantly, no halfway request for increasing bandwidth occurs in $S^{*R}(n)$, which can effectively simplify the resource management of networks. To prove above properties of $S^{*R}(n)$, this work executes the process of worst-case description that is used in [13] and obtains the empirical-envelope E^* for $S^{*R}(n)$. Expectably, the obtained E^* is identical to $S^{*R}(n)$, which concludes that $S^{*R}(n)$ is always monotonically decreasing and has the advantages of deterministic services. From the cellular network perspective, the load of the resource management can be significantly reduced if a transmission schedule for a video bitstream without halfway requests for increasing bandwidth. No additional work for power control and time-slot re-arrangement is required for a halfway increasing bandwidth request.

3. Simulation environment and test functions

In following simulation scenarios, this work assumes that ED-VTR is implemented at the video streaming server where the output link bandwidth is set to 45 Mbps. This study uses MoMuSys MPEG-4 codec and FGS (Fine Granularity Scalability) codec to generate five compressed video sequences, the sequence "*Foreman*", "*Akiyo*" and "*Stefan*", a part of movie "*Jurassic Park I*", and a part of movie "*The Firm*". All video sequences have following properties: (1) All video sequences are QCIF format with the frame rate of 30 fps. (2) A GOP consists of 15 frames and its pattern is set to I-B-B-P format. (3) Considering the property of mobile video applications, all sequences are 3 minutes in length. Since "*Foreman*", "*Akiyo*" and "*Stefan*" have only 300 frames, it is repeated cyclically to form the test video sequence. (4) Within these three sequences, the "*Akiyo*" and the part of movie "*Jurassic Park I*" have the lowest and highest picture complexity, respectively.

This work assumes that all admissible video connections are homogeneous with identical priority. To evaluate the traffic regulating performance in terms of link utilization, two call admission test functions are used. The first test function that can provide the bounded delay guarantee is expressed by

$$C(d) = \max\left\{ c \,\middle|\, \frac{1}{L} \max_{n \geqslant 0} \left\{ \left\{ \sum_{j=1}^{c} S_j(n) - n \cdot T_f \cdot L \right\} + S_m \right\} \leqslant d \right\}, \tag{11}$$

where d is the maximum delay tolerance of a connection (sec); $C(d)$ is the maximum number of admissible connections with d; c is the current number of admissible connections with d; L is the link bandwidth (bps); S_m is the maximum packet size (bits); $S_j(n)$ is the transmission schedule for connection j (bits).

The second test function that can provide high link utilization but without bounded delay guarantee is shown in (12). A new call is accepted if

$$a \cdot T_f \cdot L - \left\{ \sum_{j=1}^{c^*} S_j(a) + S_{c^*+1}(1) \right\} \geqslant 0, \tag{12}$$

where c^* represents the current number of admissible connections without bounded delay guarantee at $n = a$.

4. Simulation results

This paper first evaluates the performance between two rate reduction methods for pre-stored videos, i.e., frame skipping and FGS bit plane truncating. Figure 4 shows the results in which two QCIF video sequences, *Akiyo* and *Stefan*, are used and both MoMuSys MPEG-4 codec and FGS codec are applied. The base layer of FGS is set to 64 kbps and the maximum bit plane number of FGS enhancement layer is not limited. With the encoded video bit stream generated from MoMuSys MPEG-4 codec, this work skips B-frames based on a skipping probability and obtains a modified bitrate, R_{FS}. Then, this work skips the bits of FGS enhancement layer according to R_{FS}. From Fig. 4, it is obvious that FGS has 2.5–3.5 dB reduction in video quality in the loss free case. This is mainly due to the low video coding efficiency of FGS [19]. Moreover, considering the case that B-frame skipping probability is 0.5, the received PSNR using frame skipping still outperforms that using FGS solution up to 2.6 dB for Akiyo and 0.9 dB for Stefan. Although the effect of packet-loss flexibility of FGS is gradually obvious when the B-frame skipping probability increases, the inefficient video coding still dominates the PSNR degradation.

This study then compares the performance of ED-VTR with the traditional EEM method [13]. On the assumption that the decoder buffer space is infinite, two test video sequences, "*Jurassic Park I*" and "*The Firm*" are modeled by EEM and ED-VTR, respectively. The curves in Fig. 5 present that the initial bandwidth requirement generated from EEM exceeds the 384 kbps that is the outdoor upper bound of 3G cellular networks. Conversely, ED-VTR can significantly reduce the requirement of the initial bandwidth, particularly for complex video sequences such

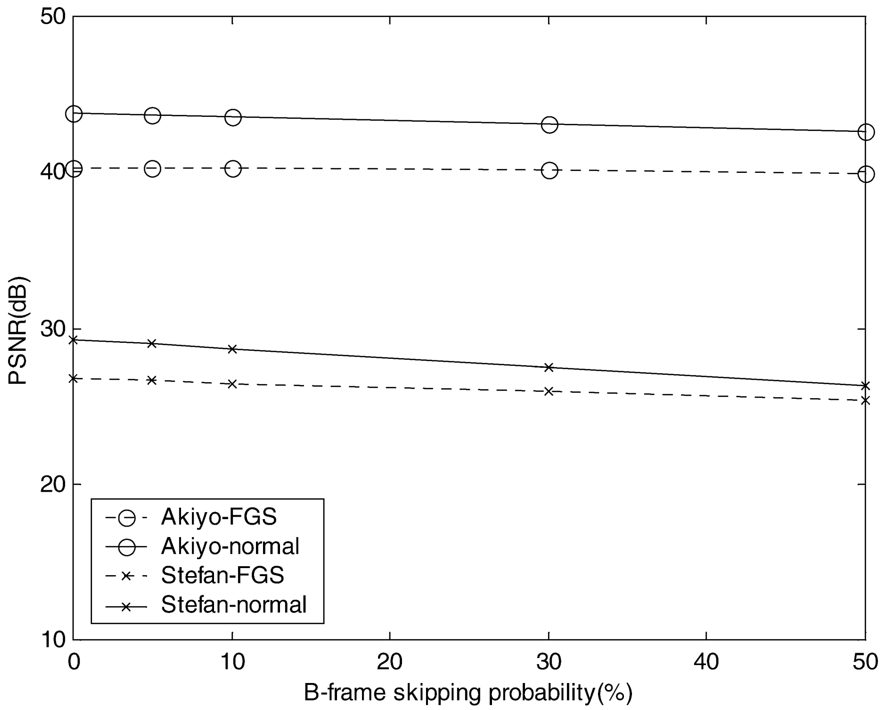

Fig. 4. Performance evaluation of FGS coding efficiency.

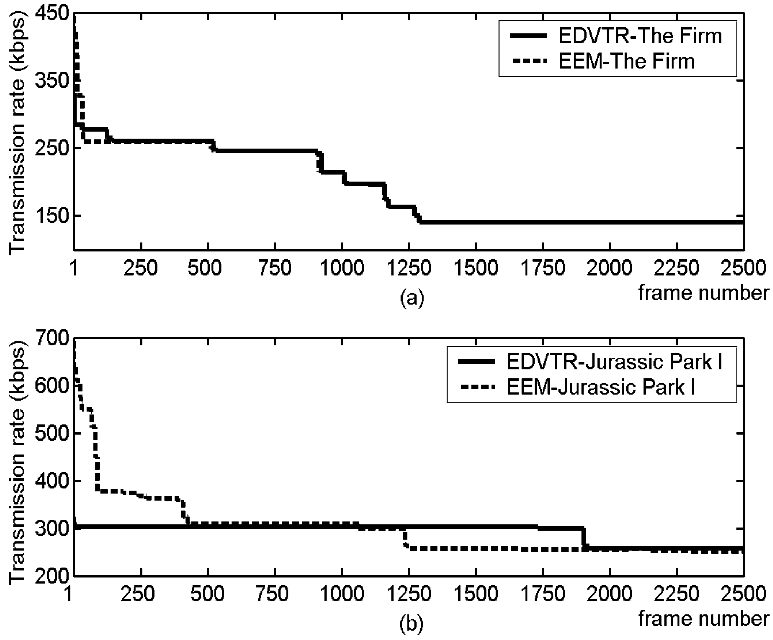

Fig. 5. Advantages of ED-VTR over EEM in the case of: (a) "*The Firm*" sequence; (b) "*Jurassic Park I*" sequence.

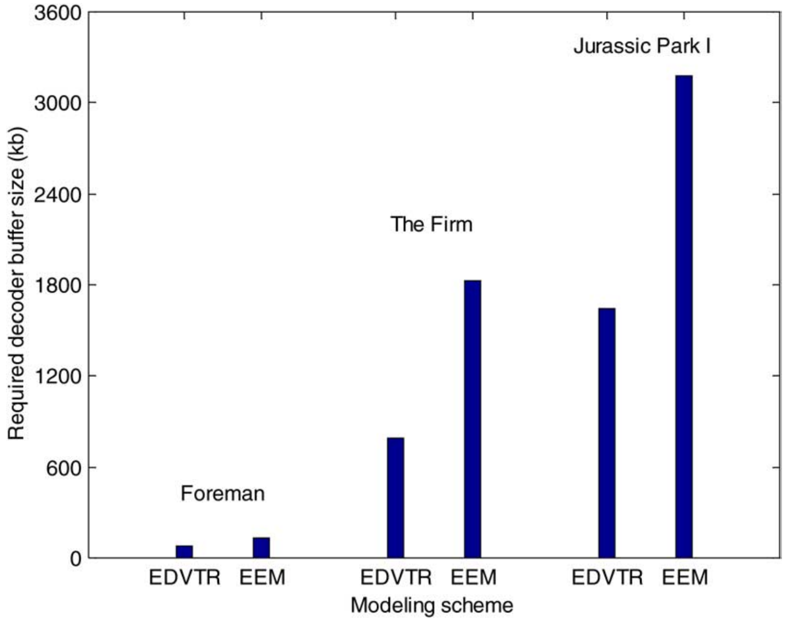

Fig. 6. Required decoder buffer space for achieving bounded delay guarantee.

as "*Jurassic Park I*". It is noted that both ED-VTR and EEM schemes generate a series of bandwidth requests whose values are monotonic decreasing and thus reduce the load of resource management. In summary, ED-VTR maintains the advantages of EEM, and has improved capability in cases of limited network bandwidth. Moreover, from Fig. 6, the minimum required decoder buffer space for achieving bounded delay guarantee is presented. Based on three different video sequences, we observe that ED-VTR requires less decoder buffer space than the EEM method. To summarize, ED-VTR can effectively enhance the capability of EEM.

Figure 7 presents the results of smoothing the test sequence by ED-VTR and WAS [18], respectively. By observing the curves in Fig. 7(a), both ED-VTR and WAS provide the identical smoothing performance in the case of sufficiently large decoder buffer space, where the preloading time D of (1) is set to two frame-times and the available decoder buffer space is set to 2000 kbits. However, if the decoder buffer space is insufficient such as the case of Fig. 7(b), we observe that the transmission rates resulted from WAS may increase and decrease alternatively the transmission rate by a wide margin during the playback. This phenomenon may complicate the resource management and cause unpredictable packet loss or latency if the available network bandwidth is nearly exhausted. In contrast, the transmission rates resulted from ED-VTR always keep the monotonically decreasing property. Briefly, ED-VTR keeps the advantages of WAS in cases of sufficiently large decoder buffer space, while improving the capability of WAS in cases of insufficient network bandwidth and decoder buffer space.

Figure 8 plots the maximum number of admissible connections that can be supported simultaneously by the network as a function of delay bound d. The simulation scenario utilizes (11) to evaluate the link utilization of EEM, WAS, and ED-VTR, respectively. To examine the impact of preloading operations on the link utilization, simulations are executed with two different preloading times, $D = 2$ and $D = 0$, respectively. From Fig. 8, ED-VTR can significantly increase the link utilization compared with other two methods, regardless of the settings of the delay bound and preloading time. Moreover, from the results of EEM in Fig. 8, the two curves with $D = 2$ and $D = 0$ are coincident with each other. This means that the link utilization of EEM is independent of the preloading operation. The main reason is that EEM always uses the worst-case description to generate a transmission schedule, and the preloading operation for reducing the huge initial bandwidth demand of EEM is invalid if a complex video frame section appears in the middle of a video sequence. In contrast, regarding the WAS or ED-VTR scheme, an appropriate preloading operation can effectively improve the link utilization when the delay bound is small; however, the effectiveness of the preloading operation obviously decreases when the permitted delay bound increases.

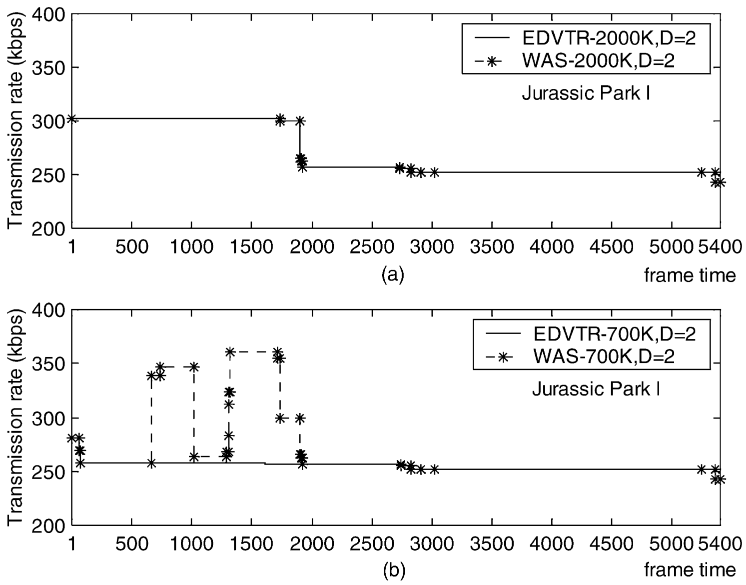

Fig. 7. Advantages of ED-VTR over WAS. (a) Buffer size = 2000 kbits, $D = 2$. (b) Buffer size = 700 kbits, $D = 2$.

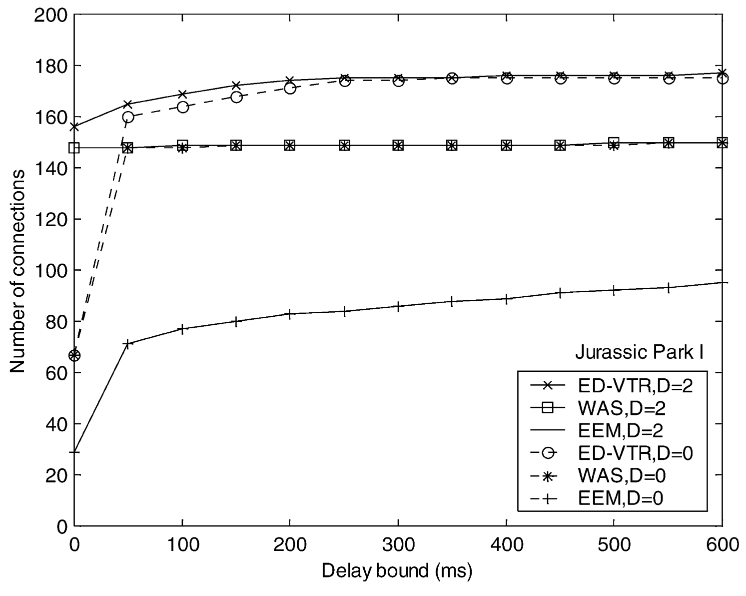

Fig. 8. Maximum number of admissible connections as a function of delay bound d.

From the viewpoint of video coding characteristics, all frames belonging to the same GOP are hurt due to error propagation in the decoding process if an I-frame is lost. This phenomenon causes significant degradation of received picture quality. Moreover, all succeeding frames belonging to the same GOP are also hurt if a P-frame is lost. This work thus defines the *hurt frame* as a video frame that is lost or hurt in quality because of error propagation.

Based on the results of Fig. 7, this study further compares the received picture quality and the amount of hurt video frames obtained using ED-VTR with that using WAS in the case of insufficient decoder buffer space. The

Table 1
PSNR improvement of ED-VTR over WAS

	Buffer space	Original PSNR	Received PSNR	Hurt I-frame	Hurt P-frame	Hurt B-frame
WAS	700 Kb	41.38 dB	34.52 dB	25	122	294
	1200 Kb	41.38 dB	35.09 dB	14	80	188
ED-VTR	700 Kb	41.38 dB	35.35 dB	0	0	277
	1200 Kb	41.38 dB	35.53 dB	0	0	138

Table 2
PSNR evaluation of ED-VTR under different wireless conditions of cellular networks

	Error free	BER $5.0\times e{-5}$	BER $1.4\times e{-4}$	BER $5.0\times e{-4}$
Frame error rate	0	$1.52\times e{-3}$	$3.91\times e{-3}$	$1.12\times e{-2}$
PSNR (dB)	35.53	35.15	34.26	32.87

scenario utilizes the test function defined by (12) and assumes that all admissible connections are simultaneously served. In addition, the error resilience functions of MPEG-4 are activated and various decoder buffer limitations, including the 700 Kb and 1200 Kb, are applied here. A halfway request for increasing bandwidth during the playback is rejected if the available bandwidth of network is nearly exhausted. From Table 1, the amount of hurt frames using ED-VTR is observed to be less than that using WAS regardless of 700 Kb or 1200 Kb. Note that no I-frame or P-frame is dropped when ED-VTR is used. Moreover, the received PSNR value obtained using ED-VTR exceeded that using WAS at both 700 Kb and 1200 Kb. The main reason is that ED-VTR can actively select the suitable frames to be skipped if needed, instead of passively and arbitrarily skipping the congested or delayed frames by the network or decoder. Finally, this study extends the simulation scenario of Table 1 to evaluate the received PSNR of ED-VTR under different wireless conditions of cellular networks, as presented in Table 2. Herein the decoder buffer limitation is set to 1200 Kb and some error patterns that are commonly applied to related researches of 3G networks are used [20,21]. The link rate, terminal speed and interleaving depth of these error patterns are set to 384 kbps, 50 km/h and 40 ms, respectively. From Table 2, it is obvious that the received PSNR degrades if the bit error rate of cellular network increases. Note that the PSNR degradation is not significant in the worst case of this simulation scenario since the error resilience functions of MPEG-4 are activated here.

5. Conclusions and future works

Regarding the delivery of pre-stored video data, the proposed ED-VTR has the same advantages as traditional smoothing and deterministic modeling schemes when the decoder buffer space of mobile clients is large enough. More importantly, ED-VTR retains the sustained playback and outperforms traditional schemes in cases of insufficient decoder buffer space. In the future, we shall extend the contributions of this paper to real-time/live video applications and best effort network environment.

References

[1] A. Dutta, J.C. Chen, S. Das, M. Elaoud, D. Famolari, S. Madhani, A. Mcauley, M. Tauil, S. Maeda, N. Nakajima, Y. Ohba and H. Schulzrinne, Implementing a testbed for mobile multimedia, in: *Proc. IEEE GLOBECOM'01*, 2001, pp. 1944–1949.

[2] I. Elsen, F. Hartung, U. Horn, M. Kampmann and L. Peters, Streaming technology in 3G mobile communication systems, *Computer* **34** (Sep.) (2001), 46–52.

[3] I.M. Pao and M.T. Sun, Encoding stored video for streaming applications, *IEEE Trans. Circuit Syst. for Video Technol.* **11** (Feb.) (2001), 199–209.

[4] H. Tomimori and Y. Nakamoto, An efficient and flexible access control framework for Java program in mobile terminals, in: *Proc. Distributed Computing Systems Workshops*, 2002, pp. 777–782.

[5] [Online] Available: http://threegmobile.ringtone-logo-game.com.

[6] J. Zhang, J. Huai, R. Xiao and B. Li, Resource management in the next-generation DS-CDMA cellular networks, *IEEE Wireless Communications* **11**(4) (2004), 52–58.

[7] D. Calin and M. Areny, Impact of radio resource allocation policies on the TD-CDMA system performance: evaluation of major critical parameters, *IEEE J. Select. Areas Commun*. **19**(10) (2001), 1847–1859.

[8] D. Kivanc, G. Li and H. Liu, Computationally efficient bandwidth allocation and power control for OFDMA, *IEEE Trans. Commun*. **2**(6) (2003), 1150–1158.

[9] S. Hartwig, M. Luck, J. Aaltonen, R. Serafat and W. Theimer, Mobile multimedia-challenges and opportunities, *IEEE Trans. Consumer Electron*. **46**(Nov.) (2000), 1167–1178.

[10] J.S. Wu and F.J. Kuo, bound on waiting time for multiplexing leaky bucket enforced sources in ATM networks, *Proc. IEE Commun*. **144**(Feb.) (1997), 17–23.

[11] V. Frost and B. Melamed, Traffic modeling for telecommunications networks, *IEEE Commun. Mag.* **32**(3) (1994), 70–81.

[12] C.S. Chang and Y.H. Lin, A general framework for deterministic service guarantees in telecommunication networks with variable length packets, in: *Proc. Int. Workshop on Quality of Service*, 1998, pp. 49–58.

[13] D.E. Wrege, E.W. Knightly, H. Zhang and J. Liebeherr, Deterministic delay bounds for VBR video in packet-switching networks: fundamental limits and practical trade-offs, *IEEE/ACM Trans. Networking* **4**(3) (1996).

[14] C.C. Lee and P.C. Chang, Deterministic traffic regulation with decoder buffer constraints for streaming videos, in: *Proc. IEEE ICME 2004*, session PD2, Taipei, Taiwan, 2004.

[15] M. Grossglauser, S. Keshav and D.N.C. Tse, RCBR: a simple and efficient service for multiple time-scale traffic, *IEEE Trans. Networking* **5**(6) (1997), 741–755.

[16] M. Soleimanipour, W. Zhuang and G.H. Freeman, Optimal resource management in wireless multimedia wideband CDMA systems, *IEEE Trans. Mobile Computing* **1**(April–June) (2002), 143–160.

[17] C.S. Chang, Stability, queue length, and delay of deterministic and stochastic queueing networks, *IEEE Trans. Automat. Contr.* **39**(5) (1994), 913–931.

[18] J.D. Salehi, Z.L. Zhang, J. Kurose and D. Towsley, Supporting stored video: reducing rate variability and end-to-end resource requirements through optimal smoothing, *IEEE Trans. Networking* **6**(Aug.) (1998), 397–410.

[19] M. van der Schaar and H. Radha, Adaptive motion-compensation fine-granular-scalability (AMC-FGS) for wireless video, *IEEE Trans. Circuit Syst. for Video Technol.* **12**(6) (2002), 360–371.

[20] ITU-T Study Group 16, Multimedia Terminals and Systems Expert. Group, Doc. Q11-I-11, Monterey, CA, Available: ftp://standards.pictel.com/q11-site/LBCmobile/error_pattern/wcdma_turbo.zip, Feb. 1999.

[21] G. Cote, S. Shirani and F. Kossentini, Optimal mode selection and synchronization for robust video communications over error-prone networks, *IEEE J. Select. Areas Commun.* **18**(6) (2000), 952–965.