Dynamic Source and Channel Rate Adaptation for Video Streaming over Wireless Fading Channels

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***Abstract*—Streaming video applications are gaining popularity over not only the fixed Internet but also mobile hand-held devices. Growth in wireless access technologies along with the advanced coding schemes show the promise of better streaming video service for the mobile users. However, wireless channels are characterized by fluctuating bandwidth, thereby making it challenging for streaming applications to ensure good quality of user experience (QoE). In this paper, we adopt an objective approach and propose a cross-layer adaptive video streaming technique, where we make use of the channel loss information to update the application layer video encoding at a slower rate and the link layer modulation and coding scheme at a faster rate. In our adaptive scheme, we focus on a pause-free playback by preventing buffer underflow at the receiver. Our simulation experiments show that, the proposed adaptive technique improves the QoE of the streaming video significantly by gracefully degrading video quality in the face of pathological cases such as wireless channel error, while ensuring an uninterrupted video playback.**

***Index Terms*—adaptive streaming video, fading channel, cross-layer design, encoding rate adaptation, transmission rate adaptation, quality of user experience**

I. Introduction

The immense popularity of online video entertainment sites such as Youtube, Metacafe, etc., shows the universal appeal of streaming video applications. A critical factor behind this popularity is the user experience with the streaming video quality, much of which is characterized by the fluent playback without freezes or pauses.

Concomitant technological advances in wireless networking, particularly in the area of broadband wireless access technologies such as WiMax, LTE, etc., show the promise of tetherless streaming of videos anywhere, anytime. These advances are expected to add to the improved user experience.

However, guaranteeing a rich user experience with streaming video over an wireless network is an imposing challenge because of the fluctuating resource availability in wireless networks, resulting in disturbances in video playbacks. Adaptive streaming [1] has been a common technique to alleviate the problem of fluctuating resource. But so far the major focus has been to mitigate the problem of congestion arising out of network resource variability [2]–[4]. Besides, although adaptation improves the overall data throughput, most of the existing works did not explore the benefit of adaptation objectively from streaming video perspective, i.e., whether adaptation can improve the overall user experience for playout of streaming video.

Another existing body of work on streaming video over wireless either considers some kind of error correction schemes [5]–[7] or uses a playout buffer at the receiver [8], [9]. Both strategies are considered unsuitable for streaming video with real-time quality of service requirements. Error correction mechanisms introduce additional coding redundancy or temporal diversity to reduce or eliminate errors due to packet loss, which aggravates network congestion besides adding an unacceptable error correction delay. Playout buffers also introduce similar undesirable delay at the receiver end.

Cross-layer adaptive techniques [10]–[12] have been found useful for bit streaming, where channel error information is leveraged to improve data throughput over wireless networks. The approach in [11] particularly relied on bit error rate (BER) feedback based transmission rate adaptation and showed it to be a more pertinent measure, compared to signal-to-noise (SNR) based or frame error rate based adaptation techniques, in aiding a higher access-level throughput. However, bit rate adaptation techniques are oblivious to video streaming application requirements and hence insensitive to quality of user experience (QoE). A recent work in [13] explored the use of application aware channel access in the context of energy savings for mobile devices. The key idea of predicting (voice or video) application behavior can be leveraged to power down the wireless device, thereby reducing channel contention. The approach in [12] considered link layer rate adaptation in an interference-aware multihop 802.11 mobile ad hoc networks. Another recent work [14] studied transmission rate adaptation in a mobile multimedia networks, which specifically addressed the quality medical image transmission in real-time. We use the application awareness and rate adaptation at the physical layer as well as at the source coding level, in order to increase video quality.

In this paper, we present a channel condition adaptive video streaming strategy for wireless networks that objectively addresses the problem by leveraging multi-level adaptations to improve the overall user experience (QoE) rather than just maximize the data throughput. In particular, we propose a scheme which adapts the video encoding rate along with the link layer transmission rate to deal with the variable channel conditions. This cross-layer adaptation helps maximize the user satisfaction by improving overall video quality and ensuring minimal freezing moments or pauses in video playback. Although the energy consumption will be affected and expected to improve in our proposed adaptive technique, our current studies do not focus on energy saving benefits. To the best of our knowledge, this objective approach to adaptive video streaming with cross layer information in wireless network resource has not been explored before.

In Section II we highlight the need for video streaming application specific multi-level rate adaptation. The possible techniques to improve user experience and a prelude to our proposed adaptation approach is outlined in Section III. The proposed two-level rate adaptation technique is presented in detail in Section IV. In Section V, simulation based experimental results are described. The paper is concluded in Section VI.

II. MOTIVATION

A. Wireless channel characteristics

The signal power received by a mobile station over a wireless channel is characterized by three different types of fluctuation or changes, *viz.* very slow changes, slow changes and fast changes [15]. Very slow changes are due to free space loss giving rise to a distance decay in the received power. It has been experimentally observed that, in typical mobile propagation paths the signal delay is inversely proportional to distance raised to the power n, where n varies between 2 and 6. Slow changes of signal strength are superposed on very slow changes and are due to the shadowing effects. Slow signal variations at the receiver are log-normally distributed about an average receive power (in dBm) and the variance of the distribution (in dB) depends on the operating environment. Fast changes are due to multi-path fading effects. The effects of multi-path include constructive and destructive interference, and phase shifting of the signal. The nature of fading depends on the strength of line-of-sight signal component at the receiver with respect to the non-line-of-sight multi-path components.

To illustrate the effects of such channel variations on a streaming video application we have emulated a wireless channel using Suzuki model [16] for a macro cell. The parameter taken for the channel emulation throughout the paper are the following. Mobile speed was taken $v = 10$ m/s; signal carrier frequency was $f_c = 2$ GHz; maximum signal-to-noise ratio (SNR) at the receiver was set at 30 dB; the number of FFT points considered was 256; sampling spacing in terms of fraction of wavelength ($\lambda_c = c/f_c$, where c is the speed of light in free space), $\delta = 4$, so the spacing between two samples was $t_s = \frac{\lambda_c}{\delta \cdot c}$ s; for shadow fading correlation distance was $d_c = 9$ m, so the number of samples within the correlation distance was $n_s = \frac{d_c \delta}{\lambda_c}$; shadow fading variance was taken 7 dB. The combined channel response variation in terms of received SNR using Suzuki model is shown in the Figure 1.

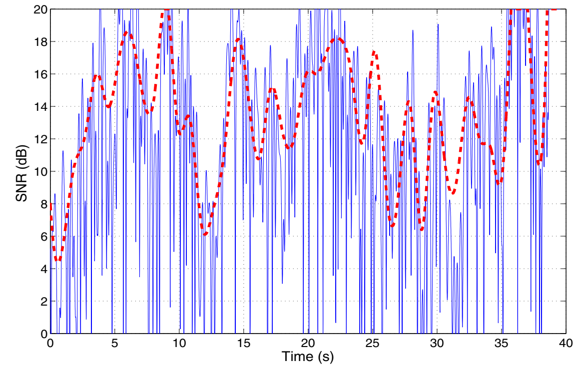


Fig. 1. Received SNR under the combined slow and fast channel variations, obtained using Suzuki model.

B. Reception quality without channel adaptation

To demonstrate the quality of video streaming performance in the face of time-varying wireless channel without any rate adaptation, we considered transmission of video frames that are encoded at a rate 620 kbps through a channel, the capacity of which varies between 800 kbps and 220 kbps. The playback rate was 15 fps (frames/s), and pre-buffering period was taken 2 s. Fig. 2 shows the variation of receiver buffer state and its effect on video playback without channel adaptation at the encoding. It can be observed that, during the period of reduced

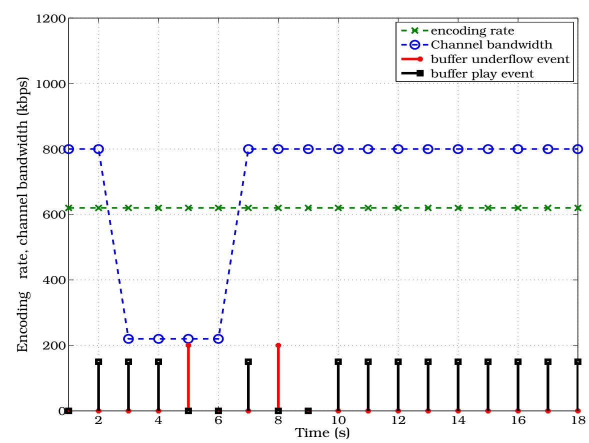


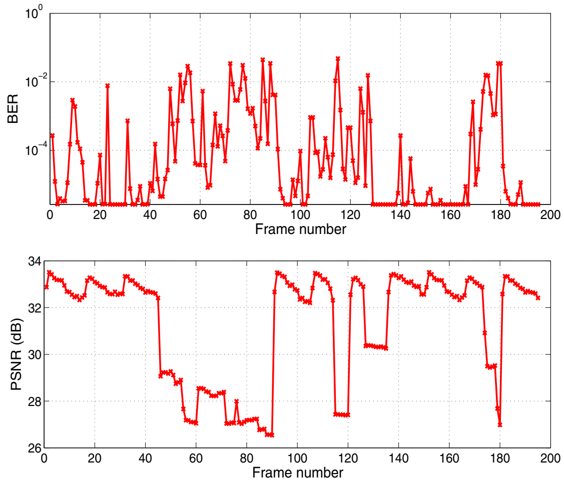
Fig. 2. Receiver buffer and playback status with channel bandwidth variation when the transmission rate is kept constant.

channel bandwidth, buffer underflow occurs, causing pauses in playback events.

Corresponding to the SNR variation in Fig. 1, bit error rate (BER) and peak signal-to-noise ratio (PSNR) variations for an MPEG-2 video streaming are shown in Fig. 3(a). PSNR is calculated frame by frame and is given by

$$PSNR = 20 \log_{10} \left(\frac{V_{peak}}{RMSE} \right),$$

where RMSE is the root mean square error of a received image frame with respect to the transmitted one and V_{peak} is the peak value of the image pixel.



(a) PSNR variation with BER

(b) Video frame during a bad channel state

(c) The original video frame at the transmitter

Fig. 3. MPEG-2 streaming video performance with channel-blind transmission

The video encoding rate was kept fixed at 600 kbps, the transmission rate was 900 kbps, and the playback rate was 15 fps. We observed that PSNR drops considerably during the period of channel error resulting in a poor quality of received stream. A snapshot of a video frame received in bad channel state is shown in Fig. 3(b) along with the originally transmitted frame shown in Fig. 3(c). Note that, since the transmission rate here is greater than the encoding rate, the video playback never freezes or pauses due to receiver end buffer underflow situation, but the quality of the video suffers considerably, as demonstrated by an almost indiscernible image quality.

In the following, we will first show how the link layer transmission rate adaptation can improve the video reception quality. We will then present our proposed multi-layer interaction strategy by jointly adjusting the encoding rate as well as transmission rate in response to the varying channel state behavior to further improve the QoE on video streaming.

III. Video Adaptation Techniques

Adaptation in video streaming can be introduced primarily in two dimensions, *viz.*, at the video encoding stage and at the wireless transmission stage by varying the modulation and coding schemes. Lowering the encoding rate reduces the transmission bandwidth requirement considerably and it can be used to mitigate temporary wireless errors. However, adjusting video encoding is a computationally expensive operation and hence it cannot be practiced to adapt to frequent and rather short-lived bursts of wireless channel errors. Assuming that the transmission rate is kept higher than the encoding rate to avoid buffer underflow situation, there is more chance of video frames getting corrupted during such pathological cases due to high transmission rate. This results again in poor video quality.

Another adaptive technique is to adjust the transmission rate as per the currently available wireless channel bandwidth [11]. In this case the transmission rate is decreased by reducing the modulation level and increasing the forward error correction overhead during the bad channel conditions [10]. The rationale behind this approach is to carry less information in an analog signal during channel errors and thereby reducing the chance of bit-level corruption. But this adaptation comes at a cost – when the frames are transmitted at a lower rate than the encoding rate, the receiver has to wait until all the frames belonging to a particular GOP (group of pictures) arrive. This results in buffer underflow situations or freezes and pauses in video playback.

In order to circumvent the overhead and uneven playback issues, we propose an adaptive scheme where the encoding rate is also varied along with the transmission rate to mitigate the buffer underflow while ensuring a higher quality of video delivery. The basic idea is to lower the encoding rate on a larger time-scale during high BER periods, so that the receiver requires lower data rate for decoding, hence solving the buffer underflow issue. At the same time we vary the transmission rate on a shorter time-scale to increase the chance of uncorrupted video data delivery during the period of wireless channel errors.

IV. The Proposed Adaptation Scheme

The proposed adaptive rate control algorithm takes informed decision on choosing an encoding rate and a transmission rate based on the channel condition and receiver buffer status. As the encoding rate change is a costly proposition, we propose to vary it slowly - with slow fading phenomenon of a wireless channel so that the encoding rate is commensurate to the average rate of transmission over a period of time. The transmission rate on the other hand is changed with the short-term changes in the channel condition, because such changes can be administered quickly by changing the modulation scheme on the fly. Besides short-term and long-term changes in the channel condition, we also consider the receiver buffer status feedback in deciding changes in encoding and transmission rates, so that the problems of receiver buffer underflow and overflow can be mitigated.

Algorithm 1 explains our joint rate adaptation approach.

Algorithm 1 Adaptation Algorithm

```
Initialize BER = 0, Encoding rate ER = ER0;
Transmission rate TR = TR0, NewSum = 0, PrevSum = 0,
BERold = 0;
while Video data is available do
   Encode the GOP with Encoding Rate = ER
   while GOP frame is available do
      Transmit frame data at transmission rate = TR;
      Get BER feedback as BERfeed;
      NewSum = NewSum + BERfeed;
      if BERfeed > BERold then
         Step down TR;
      else
         Step up TR;
      end if
      BERold = BERfeed;
   end while
   Get Buffer status and decide buffer underflow BU or
   buffer overflow BO;
   deltaBER = NewSum - PrevSum;
   PrevSum = NewSum;
   if deltaBER < 0 and BO then
      Step up ER;
   else
      Step down ER;
   end if
   Choose a new transmission rate TR in accordance with
   new ER and deltaBER;
end while
```

In this proposed adaptation algorithm we change the encoding rate at the GOP boundaries. We sum up the BER during GOP period to average out the fast variations, so that the encoding rate change will be with the slow variation of channel state. We also change the transmission rate within the GOP periods, i.e., in a shorter time scale, so that the good channel conditions – even for small time intervals – can be exploited by optimally increasing the transmission rate.

V. Experimental Evaluation and Discussion

In our simulations we have modeled a video streaming server and a playback client, as it could be in individual user-specific or or personalized video broadcast applications. The server encodes the video and sends it with a certain transmission rate. As discussed in Section II-A, we have emulated the wireless channel variations using Suzuki model. Unless otherwise mentioned, the channel parameters specified in Section II-A are kept unchanged throughout the simulations in the current work. The receiver sends feedback on its error rates at the frame level as well as the GOP level. Receiver's buffer status feedback is sent only at the GOP level.

Fig. 4 shows the simulation block diagram. The main functional entities in the system are as follows. The *Video Source* and *Display* are the transmitter and the receiver for the streaming video, respectively. The *Encoder* and *Decoder* respectively encodes and decodes the video in I, P frames using MPEG-2 standard. The *Transmit Buffer* and *Receive Buffer* are respectively transmitter and receiver side buffers used for caching the frames. The *Wireless Channel*, as described in Section II-A emulates an unpredictable transmission medium. The *Feedback Channel* is used for feedback on packet losses and receiver buffer status. The *Switching Decision Module* implements the rate adaptation algorithms as described in Algorithm 1. *PSNR Evaluation* module compares frame by frame the received video quality with the transmitted video for a statistical performance measure of the rate adaptation algorithms.

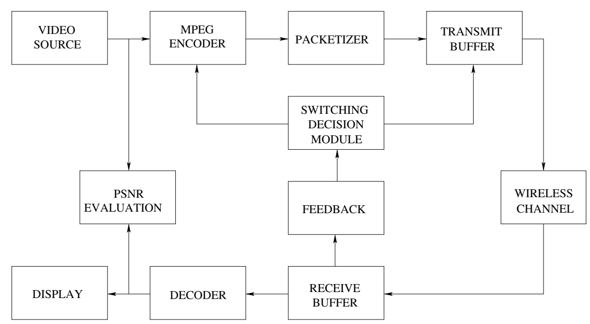


Fig. 4. Simulation block diagram

The instantaneous channel BER is derived using the model described in [11]. The BER depends on the modulation technique used. We considered a time span of 13 seconds to transmit 195 frames at a rate 15 frames/s for all the experiments. We then took summation of BERs for a GOP period to emulate the effect of slow fading. We show through the following results the effects of different schemes of adaptation.

A. With fixed encoding rate and variable transmission rate

In Section II, we have seen the effect of channel errors on streaming video without coding/transmission rate adaptation. With a constant transmission rate, there was no buffer underflow, but the quality of received video is significantly deteriorated. In the scheme with a variable transmission rate, encoding rate is kept fixed at 600 kbps and transmission rate is changed by varying the error coding scheme as well as modulation scheme. In Fig. 5, the effects of unequal error protection (UEP) on PSNR at different received SNR are shown, where standard Reed-Solomon (RS) coding is used with two different (n, k) combinations. Clearly, UEP has a better error performance over the equal error protection (EEP), and also protecting I-frames over the P-frames shows a higher benefit. In our subsequent transmission rate control studies, we have considered UEP with a higher protection for the I-frames.

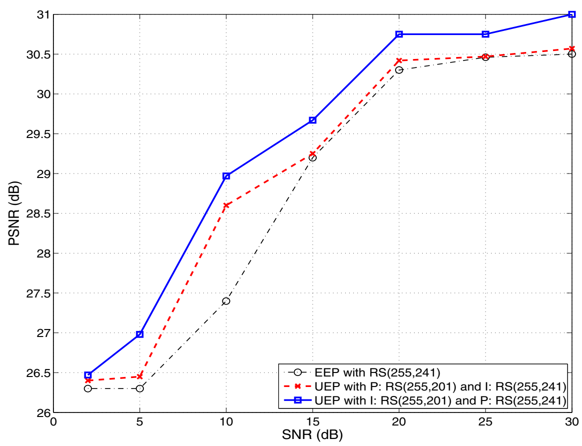
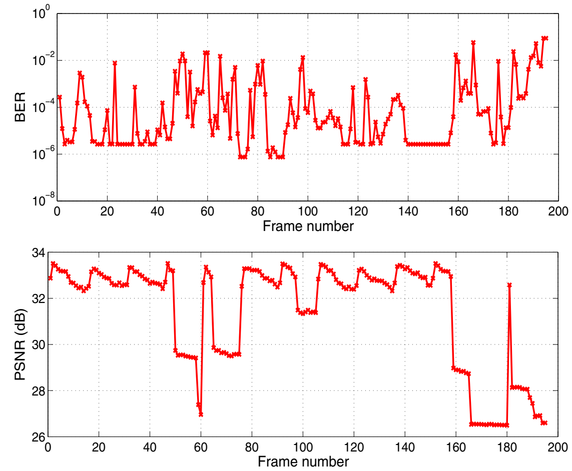


Fig. 5. PSNR variation with different error protection strategies.

For adapting modulation schemes to the available channel rate, we have considered two levels of variation: 900 kbps (8-PSK modulation) at the good channel states and 300 kbps (BPSK) during the bad channel states. As shown in Fig. 6(a), PSNR goes down during the period of channel error but not as much as in the case without adaptation, shown in Fig. 3(a). As shown in Figure 6(b) and Fig. 6(c), the rate adaptation improves the video quality at the cost of buffer underflow. Buffer underflow results from the higher decoding rate of video compared to the low transmission rate during the bad channel conditions.

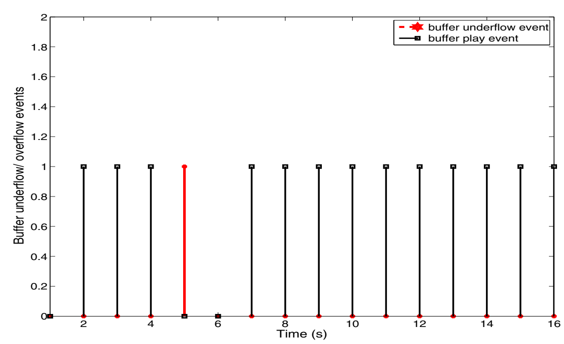
B. With variable encoding and transmission rates

To mitigate the buffer underflow problem, and hence to minimize or eliminate the pauses in video playback, we changed the encoding rate from 600 kbps to 300 kbps at the GOP levels, when the average channel condition is deteriorated. The changes in transmission rate was kept as before (discussed in Section V-A. As shown in Fig. 7(a), this has marginal impact on the resulting PSNR and hence a graceful degradation of video quality (Fig. 7(b)), but it mitigates the buffer underflow situations (Fig. 7(c)).



(a) PSNR variation with BER

(b) Quality of a received video frame during bad channel

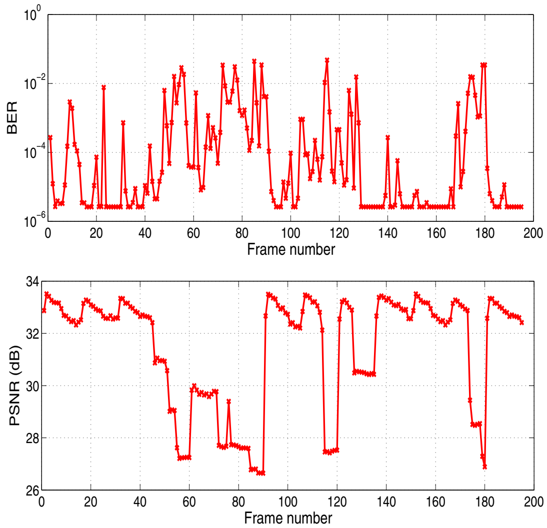


(c) Receiver buffer underflow due to bad channel conditions

Fig. 6. MPEG-2 streaming video delivery performance under time-varying channel conditions. Encoding rate is fixed, transmission rate is adapted according to the instantaneous channel capacity.

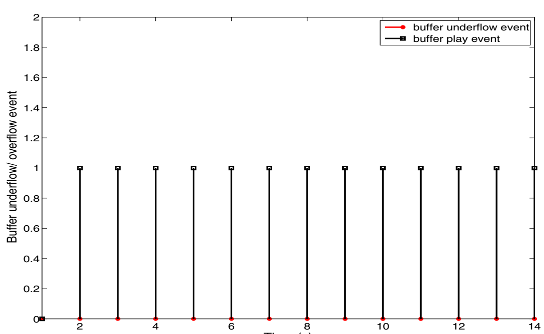
VI. Conclusion

In this paper we have presented a cross-layer adaptive video streaming mechanism for wireless networks with an aim to improve the quality of user experience. Considering an example of single-user transmission, we have shown that, it is possible to adjust the video encoding rate at the application



(a) PSNR variation with BER

(b) Quality of a received video frame during bad channel



(c) Receiver buffer underflow due to bad channel conditions

Fig. 7. MPEG-2 streaming video delivery performance under time-varying channel conditions. Encoding and transmission rates are adapted respectively according to the long-term channel statistics and instantaneous channel capacity.

layer and vary the channel modulation and coding schemes at the physical layer during pathological cases of the wireless channel, such as wireless errors. As a result, this channel aware rate adaptations at the transmitter, allows a gracefully degraded but freeze- or pause-free streaming video playback. In this study, we have used only two levels of adaptation in video encoding as well as two different modulation schemes to provide a proof of the concept of the proposed adaptive streaming technique. In our future work, we plan to quantify the performance gain more extensively, with more fine-grained adaptation scenarios, i.e., with higher levels of variation in encoding rate and modulation and coding schemes. The evaluation of energy saving performance is also an important aspect to be investigated.

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