Bandwidth estimation and robust video streaming over 802.11e wireless LANs

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BANDWIDTH ESTIMATION AND ROBUST VIDEO STREAMING OVER 802.11e WIRELESS LANS

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ABSTRACT

Streaming high quality audio/video (AV) from home media sources to TV sets over a wireless local area network (WLAN) is a challenging problem because of the fluctuating bandwidth caused by interference. Our approach is to adjust the video bit-rate dynamically in order to improve the experienced audiovisual quality. The effectiveness of rate adaptation depends on the accurate and timely estimation of the available bandwidth. In this paper we specifically focus on robust video streaming with IEEE 802.11e medium access control (MAC) enhancements for quality-of-service (QoS). These enhancements to 802.11e increase the WLAN throughput and decrease packet latencies. However, bandwidth estimation methods used in prior work, based on packet dispersion measurements, cannot be used in this context. To address this problem, we propose a new on-line application-layer bandwidth measurement method that runs at the sender application and uses the differences of packet send times and feedback receive times. The bandwidth estimation technique is implemented in a video streaming simulation environment that also includes a delay-constrained rate adaptation algorithm at the sender. Experimental results show that streaming with bandwidth estimation and rate adaptation achieves a PSNR gain up to 3.3 dB compared to streaming without rate adaptation.

1. INTRODUCTION

Wireless entertainment systems allow consumers to connect their DVD players, personal video recorders, cable/satellite receivers, and PCs to their TVs without using cables. Widely used IEEE 802.11 WLANs may provide economical, portable, and effective solutions for wireless video networking. The maximum physical data rate of 802.11b (11 Mbps) is able to support MPEG-2 encoded standard-definition (SD) video, while 802.11a/g (54 Mbps) can carry high-definition (HD) video. However, other devices and technologies operating at the same unlicensed frequency spectrum, such as cordless phones and Bluetooth, cause interference for WLANs. Interference increases the packet loss rate at the PHY layer. While retransmission-based error-control methods incorporated into the MAC recover most of the losses, multiple transmission retries at the MAC layer cause drops in the bandwidth observed at the application layer. Since video streaming applications are delay-sensitive, reduced bandwidth causes receiver buffer underruns that potentially result in frame freezes. Mobility and background traffic are other factors that can cause fluctuating bandwidth for multimedia streaming applications.

One approach to improve the experienced audiovisual quality is to dynamically adjust the video rate. In other work [1] [2], we proposed a video rate adaptation technique by performing video trans-rating. This technique adapts the bit-rate on a frame by frame basis considering the available bandwidth and the timeliness requirements of the stream.

The effectiveness of rate adaptation depends on the accurate and timely inference of the available bandwidth. In [1], the available bandwidth is estimated based on measurements of the packet inter-arrival times at the receiver. Such a technique, also called packet dispersion measurement, is used in several other works to estimate bandwidth as well [3], [4].

The IEEE 802.11e MAC enhances the basic 802.11 MAC to provide quality-of-service support for AV streams. The 802.11e MAC defines a new hybrid coordination function (HCF), which provides an enhanced distributed channel access method (EDCA) and an HCF controlled channel access method (HCCA) [5]. EDCA enables prioritization of audio/video traffic over data traffic. HCCA enables assigning dedicated contention-free transmission opportunities (TXOP) to audio/video traffic.

One feature of the 802.11e MAC we are interested in is the block acknowledgements (block-ACK) mechanism. A single block-ACK message is used to signal that multiple packets have been received, as opposed to using one ACK message for each packet (as in the legacy MAC). The block-ACK mechanism significantly increases the network throughput for bursty traffic streams by reducing the amount of overhead. However, the above-mentioned bandwidth estimation techniques based on measuring packet dispersion are not appropriate for the 802.11e HCF when using block-ACK transmissions. The main contribution of this paper is a new on-line bandwidth estimation method for 802.11e. In the following sections, the block-ACK scheme and the proposed bandwidth estimation technique are explained in detail. Then, we describe our simulation testbed, which includes bandwidth estimation and rate adaptation techniques for robust video streaming over wireless. Simulation results illustrating the performance of our bandwidth estimation and rate adaptation techniques are provided in Section 5.

2. IEEE 802.11e MULTIMEDIA ENHANCEMENTS

The legacy 802.11 MAC protocol uses an acknowledgement message for each transmitted packet. Inter-frame spacings between consecutive transmissions and acknowledgement messages cause extra overhead that reduces the effective throughput. The 802.11e block-ACK feature aims at increasing the bandwidth by minimizing overhead. This feature suits well such applications as video streaming that generate bursty traffic. Figure 1 illustrates the block-ACK mechanism [5].

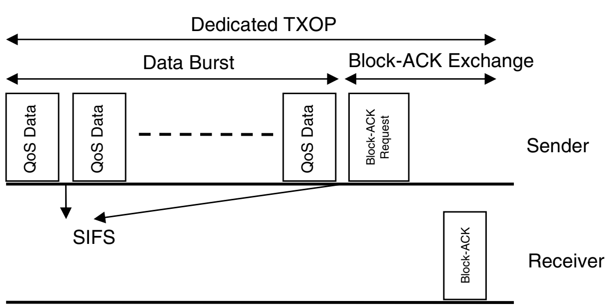

Fig. 1. Block-ACK sequence at the MAC layer.

The HCF allows wireless stations to transmit multiple MAC frames at their dedicated transmission opportunities (TXOPs). These frames are transmitted consecutively with short inter-frame spacing (SIFS). After such a burst of MAC frames ends, the sender transmits a block-ACK request frame that contains the sequence numbers of the frames in the burst. The receiver replies with a block-ACK that indicates the successfully received frames. After sending the block-ACK, the receiver MAC passes a batch of in-order packets to the upper layer. Received packets with higher sequence numbers than that of the first missing packet are buffered at the MAC layer until the missing packet(s) are recovered. However, if the pre-determined packet lifetime is reached, the sender gives up and removes the sequence number of this particular packet from the block-ACK request packet. This way the receiver is informed that the lifetime limit is reached and it forwards the remaining in-order packets to the upper layer. Because of the buffering and batch forwarding of the packets to the upper layer at the receiver, the inter-arrival times of the packets do not reflect the bandwidth of the wireless channel.

Packet lifetime limit is another 802.11e feature that improves video streaming performance. This parameter limits the maximum latency individual packets can incur, and can be related to a given end-to-end delay (or maximum duration that video data is buffered at the receiver). This stands in contrast to the packet retry limit of the legacy MAC.

3. BANDWIDTH ESTIMATION FOR VIDEO STREAMING OVER 802.11e

In this section, we describe a new on-line bandwidth estimation algorithm for 802.11e, when using the block-ACK mechanism. This algorithm runs at the video sender application and counts the number of packets that are transmitted successfully over the wireless link in a small time interval, which is called the sampling interval (SI). It does not consume any additional bandwidth since no extra probing traffic is used. The average transmission rate in the sampling interval is an estimate of the instantaneous bandwidth. The sampling interval should not include the waiting times of the packets at the sender MAC buffer, since the wait times depend on the video traffic characteristics in addition to the channel bandwidth.

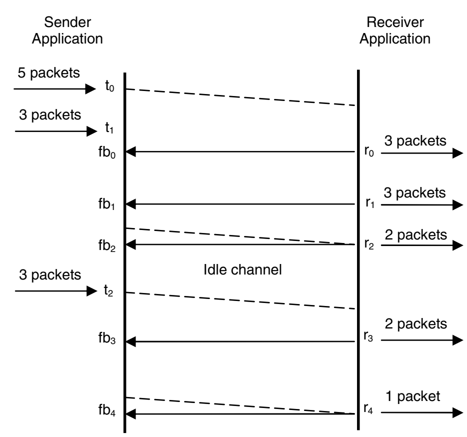

Fig. 2. Example run for the bandwidth estimation method.

In the proposed scheme, the receiver application sends a feedback message immediately after receiving a batch of video packets from the lower layer, as seen in the example in Figure 2. The maximum burst size for 802.11e block-ACK mechanism is chosen as 3 in this example. The feedback message includes the sequence number of the last received packet. A sampling interval starts when a new packet is sent and if all of the previously transmitted packets are acknowledged by the most recent feedback message. Such a packet will be transmitted immediately without buffering at the MAC layer since there is no unacknowledged packet waiting at the MAC buffer(s). As an example, t_0 and t_2 are sampling interval start times, while t_1 is not. The packets transmitted at t_1 are buffered at the sender MAC. Multiple sampling intervals may share common starting times. A bandwidth calculation is performed whenever a feedback message is received by the sender. The receive time of the feedback message (fb_i) is used as the sampling interval end point. Therefore, the duration of the sampling interval is computed by

$$SI_i = fb_i - st_i, \tag{1}$$

where st_i is the sampling interval start time. In the example above, $st_i = t_0$ for $i = 0, 1, 2$, and $st_i = t_2$ for $i = 3, 4$. The difference between the sequence number of the latest received video packet as specified by the feedback ($fbseq$) and the video packet transmitted at the sampling interval start time ($tseq$) is used for calculating the transmitted data size (tds_i), as follows:

$$tds_i = (fbseq_i - tseq_i - nlp_i + 1) \times PS \tag{2}$$

where $tseq_i$ is equal to seq_0 (sequence number of the packet transmitted at time t_0) for $i = 0, 1, 2$ and $tseq_i = seq_2$ for $i = 3, 4$. PS is the size of each video packet and nlp_i is the number of lost packets within the i^{th} sampling interval.

A sample of the instantaneous bandwidth (BWS_i) is computed by

$$BWS_i = tds_i / SI_i. \tag{3}$$

In the example in Figure 2, the length of sampling interval 4 (tds_4) gets longer because of the retransmissions, which will result in a bandwidth sample with decreased value.

Comparison of the proposed method with a packet dispersion based bandwidth measurement technique, which was originally tailored for the legacy 802.11 MAC (e.g., [1]), is illustrated in Figure 3. As previously mentioned in Section 2, buffering and batch forwarding of the packets at the receiver application causes erroneous and high bandwidth values when the packet dispersion measurements are used. On the other hand, instantaneous bandwidth measured with the proposed method results in more realistic values and a smoother trend.

4. RATE-ADAPTIVE VIDEO STREAMING AND EXPERIMENTAL SETUP

The rate-adaptation of encoded video streams is achieved via trans-rating. The utilized trans-rating scheme involves the re-quantization of the discrete cosine transform (DCT) coefficients on a frame by frame basis. The bit budget for a frame is selected based on the wireless bandwidth estimate and the delay sensitivity constraints, as described in [2]. The maximum frame bit-rate that ensures the timely delivery of the video frame is selected. The fullness of the sender buffer is used to determine the waiting time of the packets at the sender before the actual transmission. The sum of the waiting time and the transmission time gives the estimated delivery time of a packet. Interested readers are referred to [2] for the details of the algorithm.

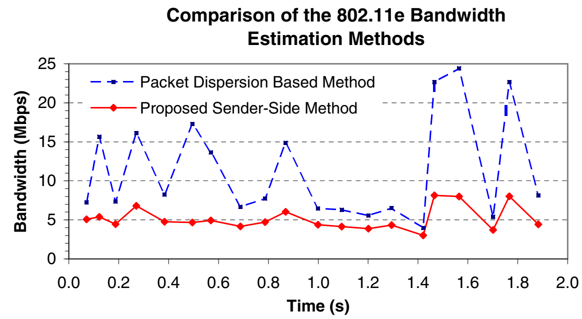

Fig. 3. Comparison of the 802.11e bandwidth estimation methods at 11 Mbps PHY Rate

In the experimental setup, we considered a scenario where the sender and receiver stations are connected with an ad-hoc wireless connection. No background traffic is allowed during the video streaming session. This scenario is a representative example since HCF could provide dedicated TXOPs for the video traffic even if data or any other background traffic is present.

We implemented the 802.11e block acknowledgement feature in the NS-2 network simulator [6] and emulated the HCCA TXOP based packet transmission. We used real channel traces from a 2.4 GHz 802.11b wireless link at 11 Mbps PHY rate to emulate a realistic physical layer in NS-2. These packet loss traces (each 100 seconds) are collected in the presence of an interference source generated by a frequency-hopping spread-spectrum cordless phone operating at the same frequency band. The maximum burst size for block-ACK mechanism is chosen to be 15.

The NS-2 simulator is also provided with a video trace. MPEG-2 encoded 5 Mbps MOBILE video sequence at 352x288 pixels (CIF) resolution and 30 frames per second rate is used throughout the simulations. The sequence is looped 10 times in order to get a total duration of 100 seconds. The encoded input video has an average PSNR of 35.7 dB. The amount of receiver video buffering in the streaming system is chosen as 100 ms in order to compensate for small delay variations. This value also represents the delay tolerance for video frames. A larger receiver buffer would imply reduced interactivity for the application, e.g. when changing the TV channel, and would require more memory, at additional cost.

Late and lost packets are applied to the transrated video at the MPEG slice level, i.e., an entire slice is removed if it overlaps with a lost packet. The quality of the final decoded output video is evaluated in terms of the PSNR and by visual comparison. Furthermore, the severity of video defects caused by late or lost packets (glitches) is quantified by computing the percentage of frames with a PSNR value lower than 20 dB. This measure is a good indicator of the perceived video quality.

5. EXPERIMENTAL RESULTS

The block-ACK mechanism improves the maximum achievable bandwidth of an 802.11b wireless link from 6.2 Mbps to about 7.8 Mbps, when using IP packets of size 1500 bytes. This means that better quality video can be streamed even if the same physical layer transmission rate (11 Mbps) is used. However, the system is still sensitive to interference. Figure 4 shows a 20 seconds sequence of the instantaneous bandwidth measurements acquired with the proposed method. Cordless phone is activated during the last 10 seconds. While the cordless phone is running, the bandwidth significantly drops and fluctuates between 4 Mbps and 6 Mbps. Since the bit-rate of the input video is 5 Mbps, fluctuation causes receiver buffer underruns which result in video freezes (glitches). This behavior can be observed in Figure 5, where the PNSR of video frames over a 300 frame period are compared for rate-adaptive streaming and streaming without rate adaptation. The cordless phone is active during this simulation. Rate-adaptive streaming has better PSNR when glitches occur, which results in a smoother and continuous video viewing experience.

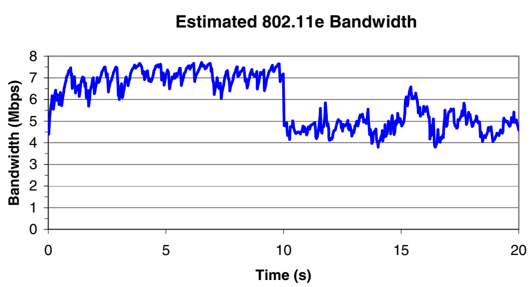

Fig. 4. Estimated wireless channel bandwidth with non-adaptive video stream.

Table 1 shows the average video quality values for a 100 s MOBILE video sequence. Video-rate adaptation results in a 3.34 dB better average PSNR than the 5 Mbps constant rate streaming. This difference can be noticed when the two resulting video outputs are compared visually. The reduction in the PSNR standard deviation means less variation in the quality. The third row of Table 1 represents the glitch percentage (the fraction of the video frames with PSNR less that 20 dB). Results show that approximately 22 percent of the video suffers from glitches when video is not trans-rated. The proposed techniques reduce this percentage to 1.8.

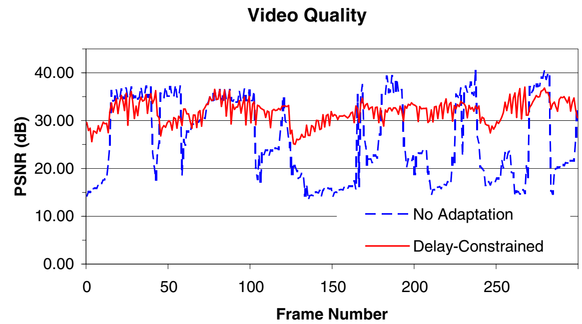

Fig. 5. Frame-level video quality comparison of rate-adaptive and non-adaptive streaming.

	Rate-Adaptive	No Adaptation
PSNR Avg.	31.99	28.65
PSNR Stdev.	3.61	8.48
Glitch %	1.8	22.3

Table 1. Average video quality comparison

6. CONCLUSIONS

In this work we proposed and tested an application layer on-line bandwidth measurement method for 802.11e MAC enhancements together with a rate-adaptive video streaming system. The measurement method is specifically tailored for block acknowledgement mechanism in 802.11e. Our method does not require extra probing traffic since it utilizes the video packets. NS-2 simulations are performed with real wireless channel traces obtained from a 802.11b network and with video test sequences. Video quality is significantly improved for scenarios where an interference source degrades the wireless bandwidth. In future work, we plan to improve the methods proposed by predicting future bandwidth values.

7. REFERENCES

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