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Network-adaptive HD MPEG-2 video streaming with cross-layered channel monitoring in WLAN*

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Abstract: In this paper, we propose a practical design and implementation of network-adaptive high definition (HD) MPEG-2 video streaming combined with cross-layered channel monitoring (CLM) over the IEEE 802.11a wireless local area network (WLAN). For wireless channel monitoring, we adopt a cross-layered approach, where an access point (AP) periodically measures lower layers such as medium access control (MAC) and physical (PHY) transmission information (e.g., MAC layer loss rate) and then sends the monitored information to the streaming server application. The adaptive streaming server with the CLM scheme reacts more quickly and efficiently to the fluctuating wireless channel than the end-to-end application-layer monitoring (E2EM) scheme. The streaming server dynamically performs priority-based frame dropping to adjust the sending rate according to the measured wireless channel condition. For this purpose, the proposed streaming system nicely provides frame-based prioritized packetization by using a real-time stream parsing module. Various evaluation results over an IEEE 802.11a WLAN testbed are provided to verify the intended Quality of Service (QoS) adaptation capability. Experimental results showed that the proposed system can mitigate the quality degradation of video streaming due to the fluctuations of time-varying channel.

Key words: Wireless video, Adaptive video streaming, Cross-layered design, Prioritized packetization, MPEG-2
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INTRODUCTION

Video streaming service nowadays enjoys explosive adoptions over all kinds of underlying networks. To enjoy the freedom of tether-less connection over the wireless local area network (WLAN), we need to overcome many challenges of unstable wireless channel due to fading, interference and so on. The scarce and fluctuating available bandwidth (together with time-varying delays and random/burst losses) can cause the video quality at the streaming client to be seriously degraded. Therefore, many channel adaptive streaming techniques in WLAN have been proposed to enhance the end-to-end Quality of Service (QoS) of video streaming (Girod *et al*., 2002; Pei and Modestino, 2001; Shan and Zakhor, 2002; Li and van der Schaar, 2004; Ahmed *et al*., 2005; van der Schhar and Shankar, 2005).

However, it was reported that the high definition (HD) video streaming of about 20 Mbps MPEG (Moving Picture Experts Groups)-2 transport stream (TS) over the nominal 54 Mbps IEEE 802.11a WLAN is not yet easily achievable due to time-varying channel condition and transmission overheads such as medium access control (MAC)-layer retransmission and request-to-send (RTS)/clear-to-send (CTS) (These overheads make that the actual available bandwidth in IEEE 802.11a WLAN is around 20 Mbps) (IEEE 802.11a WG, 1999; Qiao *et al*., 2002). In addition, this is partly because the IEEE 802.11a WLAN still lacks the pending differentiated QoS support of IEEE 802.11e (IEEE 802.11e/D8.0, 2002). Therefore, when the WLAN suffers severe channel

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degradation, we need to apply a practical network-adaptive streaming technique so that it can adapt to the dynamically varying wireless channel condition of real-life environments.

In (Park *et al*., 2005a; 2005b), we proposed a network-adaptive HD video streaming system employing frame-based prioritized packetization over the IEEE 802.11a WLAN. We adopted real-time parsing and the corresponding frame-based prioritized packetization of HD MPEG-2 TS stream. The streaming server discards frames with low priority based on the MPEG-2 frame coding type (i.e., I, P, and B) when the measured wireless channel condition is not sufficient to transmit the whole video stream. It eventually allows the streaming system to reduce the random packet loss even in poor channel condition. Here, to monitor the wireless channel, we applied an end-to-end monitoring (E2EM) scheme where a feedback message is transmitted from a client to a server periodically. A feedback message contains channel monitoring information such as packet loss rate and average jitter in Real-Time Transport Protocol (RTP) layer (Schulzrinne *et al*., 1996).

However, the E2EM scheme has a limitation in that it does not exactly represent the wireless channel condition. For example, the packet loss rate reported by feedback message only reflects passive observation of underlying wireless network condition. Also, E2EM can cause the timing gap between wireless channel monitoring in a client and actual video adaptation in a server. As a result, it makes the video adaptation to be reactive and sometimes the video adaptation may cause video adaptation to be performed even though the channel is getting better. Thus, in this paper, to make proactive wireless streaming adaptation, we combine a cross-layered channel monitoring (CLM) scheme on a sending device (e.g., access point) with streaming framework. The proposed CLM periodically checks lower layers such as medium access control (MAC) and physical (PHY) channel condition to provide active and timely monitoring results. The video streaming server in application (APP) layer has an interface with the CLM for the adaptive video streaming to match the fluctuating wireless channel.

The proposed streaming system with CLM module is implemented and integrated with the popular VLC software for MPEG-2 video streaming (http://www.videolan.org). To verify the validity of CLM-based video adaptation, we developed both E2EM and CLM modules separately. The implemented system is evaluated over an IEEE 802.11a WLAN testbed. The live or stored HD MPEG-2 TS video is fed into the streaming server and then transmitted. For performance evaluation, we measured the subjective opinion score of video playout and the objective playout discontinuity. The experimental results showed that CLM-based video adaptation can enhance the end-to-end QoS of HD video streaming over the IEEE 802.11a WLAN.

The rest of this paper is organized as follows. Section 2 explains details of the proposed CLM-based wireless video streaming system followed by experimental results in Section 3. After reviewing related work in Section 4, we conclude this paper in Section 5.

CLM-BASED WIRELESS NETWORK-ADAPTIVE HD VIDEO STREAMING

Overall framework

Fig.1 shows the proposed wireless network-adaptive HD streaming framework. The streaming

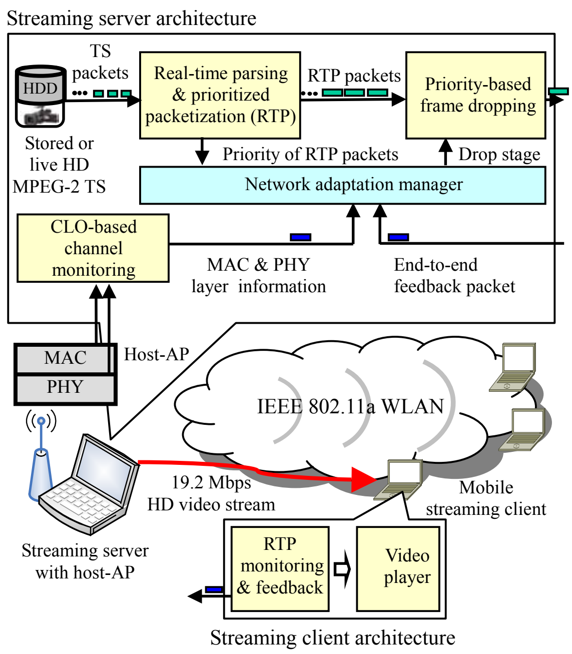


Fig.1 Network-adaptive HD MPEG-2 streaming framework with CLM over WLAN

server transmits 19.2 Mbps HD MPEG-2 TS video from an HD camera or a stored medium (e.g., HDD) to the wireless streaming client over IEEE 802.11a WLAN.

According to the measured wireless channel condition, the transmission rate of video is adaptively adjusted by using the priority-based frame dropping mechanism based on I, B, P frame structure of MPEG. To estimate the wireless channel condition, the CLM realized on top of HostAP implementation (For the implementation, we use a HostAP which is embedded with the streaming server machine. The typical AP can also be employed with modification) measures both MAC and PHY layer transmission information such as MAC-layer loss rate, MAC-layer buffer overflow rate and PHY-layer transmission rate. These are directly conveyed to the network adaptation manager that receives end-to-end feedback from the client in order to assist the CLM. The network adaptation manager determines the amount of discarded frames depending upon the measured wireless channel status (These processes enable the streaming server to implement the temporal scalability mechanism for single-layer layer video stream without special transcoding mechanism). The streaming client at mobile node then plays out the streaming video using Audio/Video decoders and renderers.

Real-time parsing and frame-based prioritized packetization

1. Real-time parsing

The real-time parsing module is designed based on the MPEG-2 TS packet structure (ISO/IEC 13818-1, 1996; ISO/IEC 13818-2, 1996). Fig.2 shows the real-time parsing procedure for MPEG-2 TS stream. When we parse TS and PES headers, both PID and stream_id fields are extracted from the headers to find the type of TS packet. After that, only the video TS packets are passed to the next MPEG-2 video ES parsing stage, since we need to identify the frame coding type for the intended prioritization. For this purpose, the MPEG-2 video ES parser finds *frame_start_code*, which indicates the start of frame. Finally, *frame_coding_type* can be extracted from the frame header. *frame_coding_type* is equivalent to that of the last TS packet in case *frame_start_code* does not exist in the current TS packet.

Note that, a TS packet that spans over multiple

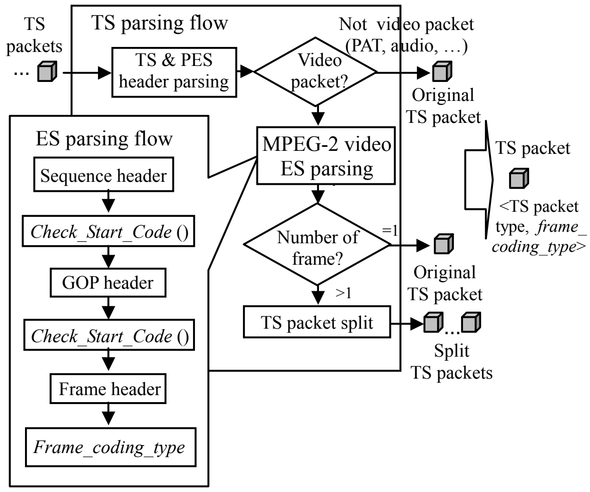


Fig.2 Real-time parsing for MPEG-2 TS

frames should be reorganized for frame-based prioritized packetization. We choose to split these kinds of packets into multiple separate packets so that each packet holds only the same type of frame data [For more detailed split scheme, please refer to (Park *et al*., 2005a)]. Finally, both the TS packet type and *frame_coding_type* in case video packets are output with original TS packet.

2. Prioritized RTP packetization

Fig.3 shows how to make prioritized RTP packets from parsed TS packets. There are five kinds of TS packets as shown in Fig.3. With 1500 bytes MTU (maximum transfer unit) constraint, seven TS packets of the same type are packetized into an RTP packet [Simple RTP packetization of MPEG-2 TS is used that mostly conforms to IETF RFC 2250 (Hoffman *et al*., 2001)]. Then, each RTP packet is prioritized according to the priority table. The priority of each RTP

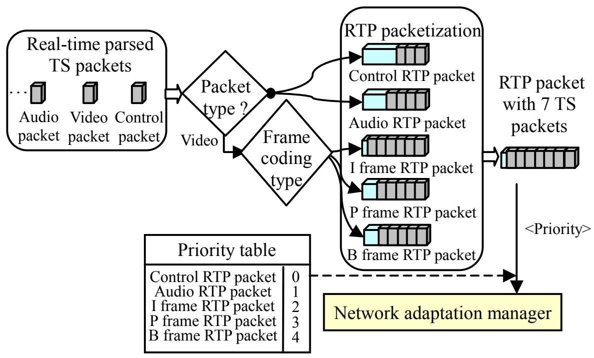


Fig.3 Prioritized RTP packetization

packet is delivered to the network adaptation manager and used to guide the priority-based frame dropping.

Wireless channel monitoring

1. Cross-layered channel monitoring (CLM)

The CLM scheme periodically (e.g., 1 s) (A monitoring period is related with accuracy of channel measurement and monitoring overhead) measures MAC-layer transmission statistics to capture the short-term channel degradation. First, it measures *N_MAC_loss*, the number of lost MAC packets during a monitoring period. *N_MAC_loss* can be obtained as follows:

$$N\_MAC\_loss = N\_L\_excess + N\_B\_over, \quad (1)$$

where *N_L_excess* and *N_B_over* are the numbers of lost packets due to the excess of MAC retransmission limit and MAC buffer overflow, respectively. Both values can be directly counted during the monitoring period. However, *N_MAC_loss* is still time-delayed channel information even though the delay is less than the E2EM. In addition, it cannot detect the temporary wireless link disconnection that may occur when sudden bursts of video streams congest the underlying WLAN.

To solve these problems, we add a proactive (i.e., predictive) channel monitoring aspect. We periodically estimate the available time resource for the transmission (*T_idle*) during an estimation interval (*T_ep*) (An estimation interval can be different with monitoring period to get *N_MAC_loss*. For the sake of simplicity, we assume that they are same). As shown in Fig.4, *T_ep* is divided into three parts where *T_used* and *T_ro* are utilized time for the actual data transmission and retransmission, respectively. Both *T_ro* and *T_used* can be estimated from the MAC-layer transmission information such as the number of retransmission trial, average packet size, and current PHY-TX (transmission) rate. First, *T_idle* can be formulated as follows:

$$\begin{aligned} T\_idle &= T\_ep - T\_used - T\_ro, \\ T\_used &= N\_tx \times T\_success, \\ T\_ro &= N\_retx \times T\_fail, \end{aligned} \quad (2)$$

where *T_success* is required time to transmit a data packet without retransmissions and RTS/CTS protection and *T_fail* is spent time due to a transmission failure. *N_tx* and *N_retx* are the numbers of transmission and retransmission trials, respectively.

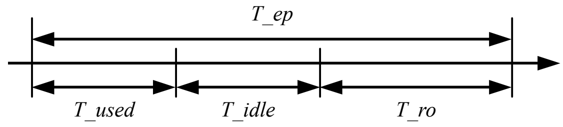


Fig.4 Distribution of the amount of wireless channel time

Since the IEEE 802.11a can support multiple PHY-TX rates (IEEE 802.11a WG, 1999; Qiao *et al.*, 2002), both *T_success* and *T_fail* can be formulated by considering PHY-TX rate, TX-failure rate, and average packet size as follows:

$$\begin{aligned} T\_success = &\left\lceil \frac{B\_mPsize}{BpS(m)} \right\rceil \times tSymbol + cSIFS \\ &+ \left\lceil \frac{B\_mPsize}{BpS(m')} \right\rceil \times tSymbol + T\_wait \\ &+ tPLCPPreamble \times 2 + tPLCPSig \times 2, \end{aligned} \quad (3)$$

where *B_mPsize* is a packet size that covers the sum of application payload, transport, network, and MAC headers. Also, *BpS*(*m*) and *BpS*(*m'*) are the number of bits per symbol for data and control packet, respectively [The IEEE 802.11a specific parameters such as *tSymbol*, *cSIFS*, *tPLCPPreamble*, and *tPLCPSig* in both Eq.(2) and Eq.(3) are stated in the relevant standard (IEEE 802.11a WG, 1999)].

$$\begin{aligned} T\_fail = &\left\lceil \frac{B\_mPsize}{BpS(m)} \right\rceil \times tSymbol + T\_wait \\ &+ tPLCPPreamble + tPLCPSig \\ &+ T\_ACKtimeout, \end{aligned} \quad (4)$$

where *T_wait* is the average value of backoff duration according to the observed rate of transmission failure (*p*) and can be formulated as follows:

$$\begin{aligned} T\_wait = &(1-p)\frac{W}{2} + p(1-p)\frac{2W}{2} + p^2(1-p)\frac{2^2 W}{2} \\ &+ \ldots + p^m(1-p)\frac{2^m W}{2} + p^{m+1}(1-p)\frac{2^m W}{2} \\ &+ \ldots + p^{Retry\_limit-1}(1-p)\frac{2^m W}{2} + cDIFS, \end{aligned} \quad (5)$$

$$p = P\_tx\_failure(t) = \frac{0.9N\_retx(t)}{N\_tx(t) + N\_retx(t)} + 0.1P\_tx\_failure(t-1), \quad (6)$$

where W is $CWmin \times cSlotTime$ that are defined in (IEEE 802.11a WG, 1999).

Finally, from the CLM, *N_MAC_loss* and *T_idle* are delivered periodically to upper-layer streaming application so that it can perform the proactive adaptive video streaming. We have implemented the CLM scheme as a software module using MADWIFI wireless ethernet card driver (http://sourceforge.net/projects/madwifi) and integrated with VLC (http://www.videolan.org) streaming server application located in the HostAP laptop (It is noted that the current implementation status of the CLM scheme has not fully taken an effect of uplink-traffics into consideration. This effect is only captured when uplink-traffic is sufficiently large to increase *N_L_excess* or *N_B_over*. It is our future work to remove this limitation).

2. End-to-end monitoring (E2EM)

To monitor the underlying WLAN, passive end-to-end measurement can also be implemented. For this purpose, we adopt simple RTP packet monitoring in the streaming client. The RTP header is parsed to extract *sequence_number* and *timestamp* fields from all received RTP packets. By using these observations, both packet loss rate (*RTP_loss_rate*) and average delay variation (*avg_jitter*) are measured periodically. Then, a feedback packet is sent to the streaming server.

Priority-based frame dropping

Fig.5 illustrates the interoperation between the network adaptation manager and monitoring modules. According to the measured wireless channel condition, the network adaptation manager in the streaming server adjusts the sending rate by using the priority-based frame dropping. For this purpose, it maintains an internal variable, *drop_stage*, to determine the stage of priority-based dropping. *drop_stage* is initialized as zero value that is equivalent to no frame dropping. That is, the whole video stream is transmitted to the streaming client. As *drop_stage* increases, the number of discarded frames will be increased too. For example, *drop_stage* can be varied from 0 to 3 if the GOP is organized as "I1B2B3P4B5 B6P7B8B9". In this scenario, all B frames in a GOP are discarded if *drop_stage* is equal to 1 while both B and P frames are discarded in case of 2. Based on *drop_stage*, video RTP packets are discarded as a unit of frame. That is, several consecutive RTP packets of the same frame are discarded completely.

Fig.6 shows *drop_stage* adjustment algorithm based on measured channel status. In the E2EM case, first, *drop_stage* is changed by comparing between *net_state* and pre-defined thresholds, where *net_state*

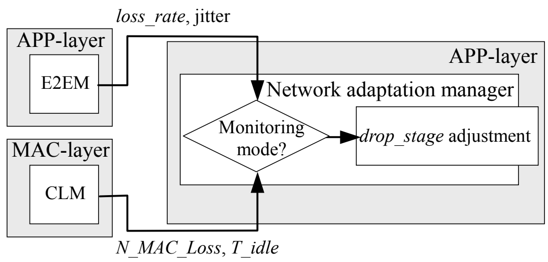
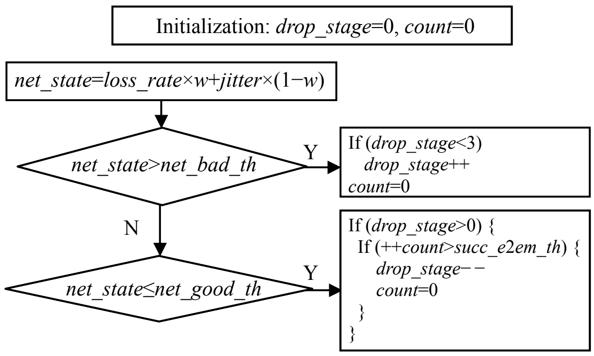
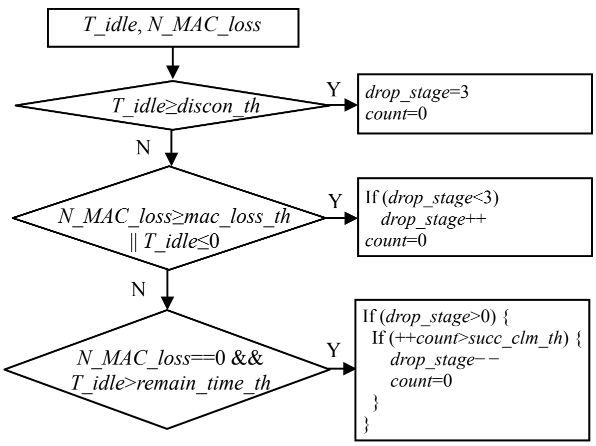


Fig.5 Interoperation among monitoring modules



(a)



(b)

Fig.6 Flowchart for *drop_stage* adjustment. (a) *drop_stage* control in E2EM; (b) *drop_stage* control in CLM

is the weighted sum of *RTP_loss_rate* and *avg_jitter* (Fig.6a). *drop_stage* is incremented by one if *net_state* is higher than *net_bad_th*. It is thus expected that the number of discarded frames are increased. On the contrary, it needs to be decreased by one if it is lower than *good_state_th*. In this case, however, *drop_stage* is decreased only when it is successively repeated as many as *succ_e2em_th* to mitigate the frequent adaptation of sending rate.

On the other hand, Fig.6b shows the algorithm to change *drop_stage* in case of the CLM-based adptation. *T_idle* and *N_MAC_loss* are compared with pre-defined thresholds, where *N_MAC_loss* is sum of *N_B_over* and *N_L_excess*. First of all, *drop_stage* has to be set to 3 when the link connection is unavailable for smooth adaptation after its reconnection. Besides, *drop_stage* is increased by one if either *N_MAC_loss* is higher than *mac_loss_th* or *T_idle* is smaller than zero. On the contrary, it is decreased by one when the *count* is larger than *succ_clm_th* while the *count* is increased by one if and only if *N_MAC_loss* is equivalent to zero and *T_idle* is larger than *remain_time_th*.

PERFORMANCE EVALUATION

Experimental setup

Fig.7 shows the WLAN HD streaming testbed consisting of IEEE 802.11a HostAP-based streaming server, and a wireless streaming client. To generate erroneous scenario, we fix the HostAP with streaming server in the room and move the streaming client and a uniform route for three different cases of streaming.

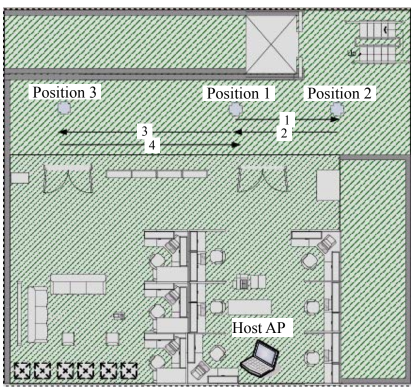


Fig.7 The map of WLAN streaming testbed

The moving pattern of the streaming client is initially set on Position 1 for 15 s and moved to next position for 15 s as shown in Fig.7. This movement is continued until the streaming client returns from Position 3 to Position 1. Table 1 shows the specification of experimental HD MPEG-2 video and parameters.

Table 1 HD video specification and experimental parameters

HD video specification	Value	HD video specification	Value
System layer	MPEG-2 TS	Resolution	1280×720
Video ES	MPEG-2 video	Original frame rate	29.97 fps
Bitrate	19.2 Mbps	GOP	IBBPBB
E2EM parameter	Value	CLM parameter	Value
Monitoring period	1 s	Monitoring period	1 s
Net_bad_th	1	*Discon_th*	0.2
Net_good_th	0.2	*remain_time_th*	0.2
		mac_loss_th	5
Succ_e2em_th	12	*succ_clm_th*	2

Experimental results

For the performance comparison, we evaluate the streaming quality of the E2EM-based adaptive video streaming, CLM-based one, and conventional one in terms of the end-to-end packet loss, playout discontinuity, rendering frames per second (RFPS) which means actually displayed frames during 1 s, and opinion score under the erroneous WLAN environment.

Fig.8 shows *RFPS*, quality of rendering frames, and measured packet loss rate of the conventional video streaming. We can see that the increasing packet loss rate causes quality degradation of the received video as well as lower *RFPS*. When the packet loss rate is higher than 5%, there is serious quality degradation of rendered frames.

Fig.9 shows *RFPS* and *drop_stage* by the E2EM-based adaptive video streaming. We can see several weakness of this scheme. Around 20 second, for example, we can see that it does not increment *drop_stage* to 1 before *RFPS* is fluctuating but after, because of the reactive nature of E2EM-based adaptation. Around 95 second, *RFPS* is severely decreased due to the packet loss. If that is so, the client should transmit feedback information so that the streaming server changes the sending rate. However, this feedback packet could not reach the streaming server due to the packet loss. Therefore, the loss of feedback packet causes serious problem in the E2EM-based adaptation. At around 100 second, the server toggles *drop_stage* to 0 even though the channel condition is not enough to take full rate of HD video stream. Thus, additional packet losses can be inccurred due to increasing sending rate at the 105 second.

Fig.10 shows *RFPS*, sending rate measured at the streaming server, *N_MAC_loss*, *T_idle*, and the intended *drop_stage* by the CLM-based adaptive video streaming. Let us see the proactive adaptability of this scheme. Near the 20 second, *drop_stage* is increased due to *T_idle* is smaller than *remain_time_th*

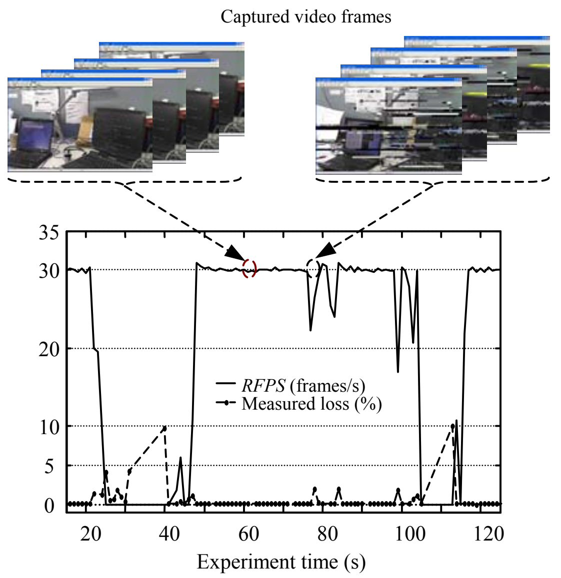


Fig.8 ***RFPS*, snapshot of rendered frame, and measured packet loss rate at the streaming client without adaptive streaming**

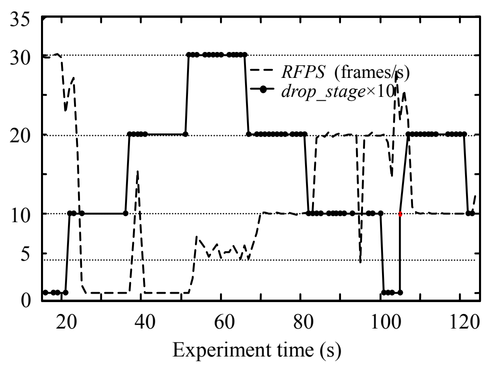


Fig.9 ***RFPS* and intended *drop_stage* by the E2EM-based adaptive video streaming**

(i.e., 0.2 in this experiment). The increasing number of MAC retransmitted packet reduces the available wireless resource even there is no real packet dropping caused by exceeding retransmission limit or sending buffer overflows. This can benefit the received video quality because the frame rate can be smoothly adjusted before playout distortion occurs.

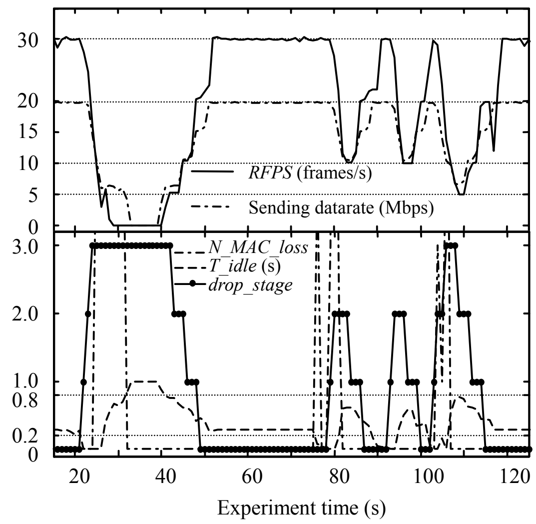


Fig.10 ***RFPS*, sending rate measured at the server, *N_MAC_loss*, *T_idle*, and intended *drop_stage* when the CLM-based adaptation is performed. (a) RFPS and sending rate measured at the server; (b) Number of MAC packet loss, *T_idle*, and intended *drop_stage***

In addition, the CLM-based adaptation has an advantage in the disconnection of wireless link correctly. For both E2EM-based and CLM-based adaptations, they must transmit at least 8 Mbps (i.e., the rate when *drop_stage* is 3. Only I frames are transmitted) when the link connection is alive. In the CLM-based adaptation, if *T_idle* is smaller than *disconnect_th*, the streaming server can set *drop_stage* to 3 when the channel is ready to be recovered at the lowest sending rate. At 32 second, since the link connection is broken, there is no more data transmission and retransmission. However, CLM-based adaptation keeps staying at *drop_stage* 3 and tries to smooth adaptation in contrast with the case of the E2EM-based adaptation.

To show the quantitative performances of the three different streaming, we performed 10 times of experiment for each case and then, took the average of those results. To calculate discontinuity (In this paper, we define the discontinuity as time duration between two consecutive rendered frames) in quantitative form, we define a threshold of 0.2 s to decide whether the video is played out or not. That is, the discontinuity is accumulated if the new frame is not rendered in 0.2 s. The ratio of accumulated time that exceeds the threshold in the total playout time is the "Discontinuity" in Table 2. As the table indicates, the CLM-based adaptation outperforms the others in terms of packet loss ratio, Discontinuity, and opinion score. In the case of no adaptation, however, it has the largest *RFPS* among the three video streams being continuously transmitted at the rate of 30 frames per second regardless of the channel status. Thus it has more chance to render more frames, although they are corrupted and not smoothly played out.

Table 2 Comparison of loss late, discontinuity, RFPS, and "*Opinion_score*" among adaptation schemes

	No_Adapt.	CLM_Adapt.	E2EM_Adapt
Loss_mean	0.46%	0.28%	0.33%
Loss_max	9.93%	9.52%	9.69%
Loss_std.	1.59%	1.25%	1.34%
Discontinuity	18.75%	10.90%	13.10%
RFPS	21.95	18.87	14.50
Opinion_score	1.88	4.21	3.62

To evaluate more subjective quality of reconstructed video, we have preliminary tested the opinion score of 30 video clips. We store all reconstructed video at the rendering time and then, show them randomly to 10 persons, simultaneously at the same place. They note the score of each from 1 to 5 (the higher the score, the better the quality). The "*Opinion_score*" in Table 2 is the average of each streaming strategy. The CLM-based adaptation also outperforms the others in terms of opinion score.

RELATED WORK

In wireless video streaming, the cross-layer approach has been considered as a reasonable choice by many researchers since the traditional wireless network are optimized in the individual layer without explicit considering of continuous media such as video and audio delivery. In addition, the existing adaptive video streaming schemes in wired network cannot be adopted for wireless network without careful modification (van der Schhar and Shankar, 2005). Many video streaming schemes in WLAN have been proposed based on the cross-layer design (Pei and Modestino, 2001; Shan and Zakhor, 2002; Li and van der Schaar, 2004; Ahmed *et al*., 2005; van der Schhar and Shankar, 2005).

In (Pei and Modestino, 2001), the unequal error protection (UEP) scheme for scalable video stream is proposed. Here, the channel coder in PHY-layer uses the priority information (i.e., base and enhancement layer) of scalable video to adaptively select the channel coding rates based on the channel status information (CSI). However, the CSI is statically fixed for the evaluation and dynamic monitoring scheme of CSI is not shown. Shan and Zakhor (2002) proposed the priority-based Automatic Repeat Request (ARQ) and Forward Error Correction (FEC) scheme at APP-layer. To avoid excessive delay in ARQ, the UDP-Lite is employed at transport-layer. In (Li and van der Schaar, 2004), the retry-limit at MAC-layer is adaptively adjusted in real-time to provide different protection strategy for layered coded video. Ahmed *et al*.(2005) proposed the cross-layer MPEG-4 video delivery scheme that adaptively tune transport parameters such as bit rates and QoS mechanisms by using information received from the network. However, time-varying characteristic of wireless network was not dealt with deeply. An excellent review of video streaming based on the cross-layered design is provided in (van der Schhar and Shankar, 2005).

CONCLUSION

In this work, we designed and implemented an HD video streaming system that adaptively transmits the source video depending upon the wireless channel status. For more accurate and fast adaptation, we adopted the cross-layered wireless channel monitoring scheme to the video streaming. Through extensive real-life experiments, we verified the validity of CLM-based video adaptation. However, the current implementation of CLM has a limitation in that we set thresholds such as *disconnect_th* and *remain_time_th* as fixed values even though the PHY-TX rate can be varied. We need to choose more appropriate thresholds that enhance the video streaming performance on various PHY-TX. Also, we expect that the efficient combination of E2EM and CLM can be developed for wireless video streaming.

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