An Access Point-Based FEC Mechanism for Video Transmission Over Wireless LANs

Cheng-Han Lin, Ce-Kuen Shieh, *Member, IEEE*, and Wen-Shyang Hwang, *Member, IEEE*

***Abstract*—Forward Error Correction (FEC) is one of the most common means of performing packet error recovery in data transmissions. FEC schemes typically tune the FEC rate in accordance with feedback information provided by the receiver. However, the feedback and FEC rate calculation processes inevitably have a finite duration, and thus the FEC rate implemented at the sender may not accurately reflect the current state of the network. Thus, this paper proposes an Enhanced Random Early Detection Forward Error Correction (ERED-FEC) mechanism to improve the quality of video transmissions over Wireless Local Area Networks (WLANs). In contrast to most FEC schemes, the FEC redundancy rate is calculated directly at the Access Point (AP). Moreover, the redundancy rate is tuned in accordance with both the wireless channel condition (as indicated by the number of packet retransmissions) and the network traffic load (as indicated by the AP queue length). The experimental results show that the proposed ERED-FEC mechanism achieves a significant improvement in the video quality compared to existing FEC schemes without introducing an excessive number of redundant packets into the network.**

***Index Terms*—FEC, video quality, wireless access point.**

I. Introduction

THE use of wireless devices such as laptop computers and PDAs to connect to Internet services is becoming increasingly common nowadays. However, wireless communication channels are prone to serious transmission errors due to attenuation, fading, scattering or interference [1]–[4]. Packet losses in wireless environments are generally recovered using either Automatic Repeat reQuest (ARQ) or Forward Error Correction (FEC) methods [5]–[8]. ARQ schemes automatically retransmit the lost packets during timeouts, or in response to explicit receiver requests. By contrast, in FEC schemes, the effects of potential packet losses are mitigated in advance by transmitting redundant packets together with the source packets such that a block of packets can be successfully reconstructed at the receiver end even if some of the packets within the block are lost during transmission. Of the two approaches, FEC schemes result in a lower retransmission latency, and are therefore widely preferred for the delivery of video streams over wireless networks [9].

Conventional FEC mechanisms are sender-based, i.e., the redundant packets are generated and encoded at the sender end. Broadly speaking, sender-based FEC schemes can be categorized as either Static FEC (SFEC) or Dynamic FEC (DFEC). In SFEC schemes, the number of redundant packets added to the source packets remains constant irrespective of changes in the network condition. The recovery performance of SFEC schemes is therefore somewhat unpredictable because they fail to capture the real-time network conditions and adjust the FEC redundancy rate accordingly. Thus, various DFEC schemes have been proposed in recent years [10]–[16]. In such schemes, the FEC rate is tuned dynamically in accordance with changes in the channel condition or network load. In most DFEC schemes, the FEC rate is tuned at the sender based on information provided by the receiver. For example, in [11], the packet error rate is measured periodically at the receiver side and fed back to the sender, which then calculates the FEC rate required to maintain a constant packet error rate at the receiver end. In the FEC mechanism proposed in [12], the FEC rate is adjusted incrementally in such a way as to preserve a pre-determined value of the Peak Signal-to-Noise Ratio (PSNR) at the receiver end. Meanwhile, in [13], the FEC scheme modifies the FEC rate in accordance with changes in the network delay.

The FEC redundancy rate is traditionally calculated at the application layer based on feedback information such as that provided by acknowledgement messages (ACKs). However, the feedback and FEC rate calculation processes have a finite duration, and thus there is no guarantee that the FEC rate implemented at the sender end accurately reflects the current network condition. Accordingly, in the studies presented in [17]–[21], the FEC mechanism was implemented at the wireless Access Point (AP) and the FEC redundancy rate was calculated directly without feedback information from the receiver. In [17], Lin *et al.* proposed an AP-based FEC mechanism designated as Enhanced Adaptive FEC (EAFEC), in which the FEC rate was determined dynamically in accordance with both the network traffic load (as indicated by the queue length at the AP) and the wireless channel state (as indicated by the number of packet retransmissions). In a later study [20], the same group proposed an alternative AP-based FEC mechanism designated as Random Early Detection Forward Error Correction (RED-FEC) based on a random early detection algorithm. In the proposed approach, the redundancy rate was gradually reduced as the AP queue

Manuscript received June 27, 2011; revised November 30, 2011 and April 16, 2012; accepted June 14, 2012. Date of publication October 16, 2012; date of current version December 12, 2012. This work was supported by the National Science Council (NSC), Taiwan under Contract No. NSC 100-2221-E-151-036. The corresponding author is W.-S. Hwang. The associate editor coordinating the review of this manuscript and approving it for publication was Dr. Zhihai (Henry) He.

C.-H. Lin and C.-K. Shieh is with the Department of Electrical Engineering, National Cheng Kung University, Tainan 701, Taiwan (e-mail: jhlin5@hpds.ee.ncku.edu.tw; shieh@ee.ncku.edu.tw).

W.-S. Hwang are with the Department of Electrical Engineering, National Kaohsiung University of Applied Sciences, Kaohsiung, Taiwan (e-mail: wshwang@mail.ee.kuas.edu.tw).

Color versions of one or more of the figures in this paper are available online at http://ieeexplore.ieee.org.

Digital Object Identifier 10.1109/TMM.2012.2225028

length increased. However, in determining the FEC rate, the network loss rate was ignored. In practice, it is crucial for FEC mechanisms to have the ability to accurately detect channel fluctuations and to manipulate the FEC redundancy rate accordingly. Consequently, Han *et al.* [21] proposed an Adaptive Cross-layer FEC mechanism (ACFEC) in which loss information was retrieved from the ARQ function of the MAC layer and the redundancy rate was controlled adaptively in accordance with changes in the network condition. However, ACFEC does not take the effect of the network traffic load into consideration. As a result, packets may be lost at the wireless AP under heavy network loads due to a self-induced congestion problem.

This paper proposes an Enhanced Random Early Detection Forward Error Correction (ERED-FEC) mechanism for improving the quality of video transmissions over wireless LANs (WLANs). In the proposed approach, redundant FEC packets are generated dynamically at the AP in accordance with both the condition of the wireless channel and the current network traffic load. The channel condition is evaluated by monitoring the number of packet retransmissions. As the number of retransmissions increases (i.e., the condition of the wireless channel deteriorates), a greater number of redundant FEC packets are generated. Conversely, as the channel condition improves, the number of FEC packets is reduced. The network traffic load is evaluated by monitoring the queue length at the wireless AP. If the queue is almost empty, i.e., the network is only lightly loaded, the number of redundant FEC packets is increased. By contrast, if the queue is nearly full, i.e., the network is heavily loaded, the number of FEC packets is reduced. By adopting this approach, the ERED-FEC algorithm significantly improves the video quality without overloading the network with an excessive number of redundant packets. An analytical model is proposed for predicting the quality of MPEG-4 video streams delivered over WLANs with FEC protection in terms of the effective packet loss rate and the Decodable Frame Rate (DFR) [22], [23]. It is shown that the model provides the ERED-FEC mechanism with the means to determine the FEC redundancy rate required to guarantee the QoS requirements of video transmissions over lossy wireless networks.

The remainder of this paper is organized as follows. Section II reviews the basic concepts of the FEC recovery technique and discusses previous related works in the field. In addition, the contributions of the present study are formally stated. Section III outlines the proposed ERED-FEC mechanism and describes the workflow of the FEC rate determination process. Moreover, an analytical model is proposed for evaluating the loss recovery performance of the ERED-FEC scheme. Section IV compares the performance of the ERED-FEC mechanism with that of three other AP-based FEC schemes, i.e., SFEC, RED-FEC and ACFEC. Finally, Section V provides some brief concluding remarks and indicates the intended direction of future research.

II. Related Works

A. Forward Error Correction (FEC)

The basic principle of FEC entails injecting redundant packets (h) into the video stream together with the source transmission packets (k) such that packet losses can be recovered at the receiver end without the need for retransmission. In other words, as shown in Fig. 1, the original block is encoded as (n, k) packets, where n is the summation of source packets (k) and redundant packets (h). Thus, provided that no more than h packets are lost in transmission, the source transmission packets can be successfully recovered at the receiver. Since FEC schemes enable the recovery of source packets which would otherwise be lost, the effective loss rate in the transmission network is lower than the actual loss rate.

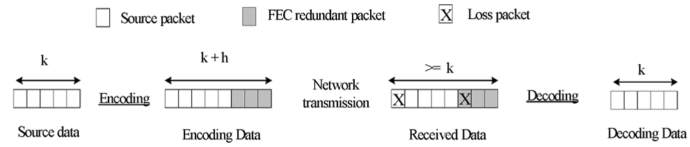

Fig. 1. FEC encoding and decoding.

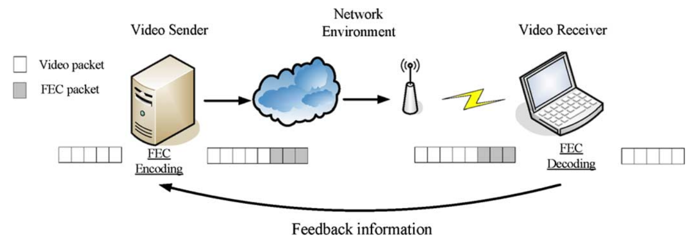

Fig. 2. Sender-based FEC scheme.

In FEC codec, redundant packets are derived from the original packet using conventional coding theory techniques. Of the various traditional error correcting codes available for this purpose, Reed-Solomon (RS) code [24], [25] has attracted particular interest. RS code provides an ideal error protection capability against packet losses since it is a maximum distance separable code, i.e., no other coding scheme exists capable of recovering lost source data symbols from a lesser number of received code symbols.

B. Sender-Based FEC Mechanisms

1) Constant Error Rate FEC (CER-FEC): Takahata *et al.* [11] proposed a sender-based Constant Error Rate FEC (CER-FEC) scheme for enabling the dynamic QoS control of real-time multimedia streams over heterogeneous environments comprising wired and wireless connections. As shown in Fig. 2 in the proposed scheme, the packet error rate is periodically observed at the receiver side and any change in the error rate is fed back to the sender. Upon receiving this information, the sender calculates the number of redundant packets required to restore the error rate to its original value. In other words, the FEC redundancy rate is dynamically controlled in such a way as to maintain a constant packet error rate at the receiver end.

2) Cross-Layer FEC (CL-FEC): Bajic *et al.* [12] proposed an efficient Cross-Layer FEC (CL-FEC) scheme for wireless video multicasting designed to maintain the received video quality for all the users above a certain pre-specified level. In the proposed scheme, each user periodically reports the number of packets received out of the previously transmitted k packets. The sender then calculates the number of packets which each user has lost

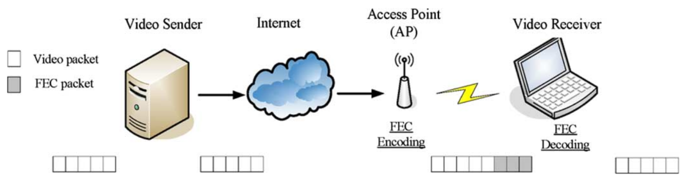

Fig. 3. AP-based FEC scheme.

and determines the maximum number of packets which can be decoded by all the users (i.e., the number of decodable packets for the user with the greatest number of packet losses).

3) Adaptive FEC (AFEC): Park *et al.* [13] presented an adaptive FEC (AFEC) protocol for facilitating the end-to-end transport of real-time traffic whose timing constraints rule out the use of retransmission-based congestion control or QoS provisioning schemes. In the proposed approach, the degree of FEC redundancy is tuned in accordance with the current network delay. Specifically, the number of redundant packets is increased as the network delay decreases, but is reduced as the delay increases.

C. AP-Based FEC Mechanisms

1) Random Early Detection FEC (RED-FEC): In heavily-congested networks, traditional FEC-based error recovery schemes increase the redundancy rate in order to compensate for the greater number of packet losses. However, the redundant packets worsen the network congestion, and therefore further degraded the network performance. To address this problem, Lin *et al.* [20] proposed a Random Early Detection FEC (RED-FEC) scheme in which the redundant FEC packets are generated dynamically at the wireless AP in accordance with the current network traffic load, as indicated by the AP queue length (see Fig. 3). Specifically, the number of redundant packets is increased as the queue length shortens, but is reduced as the queue length grows. Importantly, when the queue is near to full, no FEC packets are generated in order to avoid overloading the network. By adopting this approach, the RED-FEC mechanism improves the quality of the delivered video stream without injecting an excessive number of redundant packets into the network.

2) Adaptive Cross-Layer FEC (ACFEC): Han *et al.* [21] proposed an Adaptive Cross-layer FEC (ACFEC) scheme for enhancing the quality of video transmissions over IEEE 802.11 WLANs. The cross-layer design enables the ACFEC mechanism to leverage the functionalities of the different network layers. For example, packet loss information is retrieved using the ARQ function of the MAC layer, while the FEC redundancy rate is controlled adaptively at the application layer utilizing the UDP protocol. Specifically, as the source packets are transmitted through the wireless AP, the ACFEC mechanism monitors the transmission performance continuously via the failure information from the MAC layer. After transmitting one block of video packets, the failure counter is used to adjust the FEC redundancy rate accordingly.

D. Contribution of Present Study

The major contribution of the present study is to propose a new AP-based FEC mechanism (ERED-FEC) for improving the quality of video transmissions over wireless LANs (WLANs). The literature contains many proposals for sender-based FEC schemes [11]–[13], which have a finite duration to feedback information from the receiver. Thus, the FEC rate determined at the sender end may not accurately reflect the current network condition. The proposed ERED-FEC mechanism is AP-based and the FEC rate is calculated at the AP directly without feedback information from the receiver. Moreover, while the literature also contains various proposals for AP-based FEC schemes [20], [21], these schemes consider only single metric, such as the wireless error rate or only the traffic load to determine the FEC rate. By contrast, in the ERED-FEC mechanism proposed in this study, the FEC rate is controlled adaptively in accordance with both the wireless channel condition and the network traffic load. By adopting this approach, the ERED-FEC mechanism significantly improves the video quality and avoids overloading the network with an excessive number of redundant packets.

In addition, this paper proposed an analytical model for predicting the performance of video transmissions over a WLAN with FEC protection. In fact, the video quality is determined not only by the loss effect of wireless network but also the coding dependency of MPEG-4 video frames. However, the analytical models in previous related works [22], [23], [26]–[28] did not take the FEC recovery performance and frame coding dependency aspects into consideration. In [26]–[28], the video quality cannot be directly evaluated using these models because these models did not include coding dependency of video frames. Moreover, the Decodable Frame Rate (DFR) which is proposed in [22], [23] is a performance metric to assess the video quality of streaming MPEG-4 video. However, the loss effect of wireless transmissions on video quality using DFR is measured by a simple parameter (such average packet loss rate in wireless network) without considering the actual behavior of the FEC recovery performance. Therefore, the advantage of the proposed model is to consider not only effects of FEC recovery performance but also the impact of the loss of specific MPEG-4 video frames.

III. Enhanced Random Early Detection FEC (ERED-FEC) Mechanism

A. Basic Concept of ERED-FEC Mechanism

Fig. 4 illustrates the basic architecture of the AP-based ERED-FEC mechanism proposed in this study. (Note that an assumption is made that the wired segment of the video delivery path is loss free.) As shown, the ERED-FEC mechanism consists of five components, namely (1) a packet type classifier, (2) a packet loss monitor, (3) a video quality model, (4) a network load monitor and (5) a FEC packet generator. During video streaming, the streaming server encapsulates the video data in Real-time Transport Protocol (RTP) packets, and delivers them to the receiver through the wireless AP. When a packet arrives at the AP, the ERED-FEC controller retrieves the packet header from the UDP, and identifies the packet type by checking the RTP header. Once a complete block of video packets has arrived, the packet loss monitor estimates the packet loss rate by examining the number of packet retransmissions associated with the block. An appropriate FEC redundancy rate is then

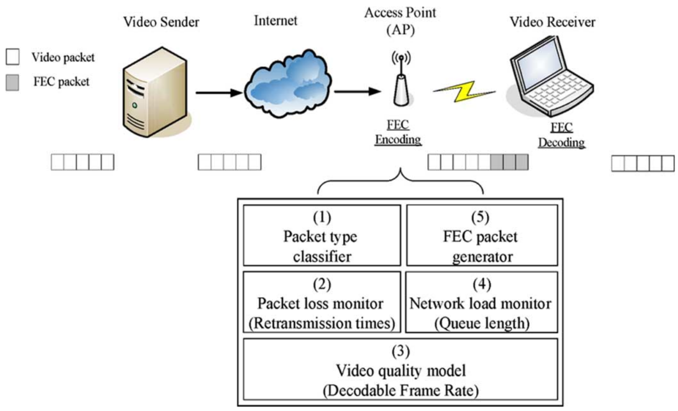

Fig. 4. Architecture of ERED-FEC controller.

determined via the video quality model (i.e., the DFR). Finally, the ERED-FEC mechanism checks the queue length at the AP in order to evaluate the current network traffic load, and then uses this information to adjust the FEC redundancy rate (if required).

B. ERED-FEC Algorithm

1) Estimation of Packet Loss Rate in Wireless Network: In designing the ERED-FEC mechanism, it is assumed that the wireless errors are generated at a rate of $P_{\rm pkt}$ in accordance with a random uniform model. In addition, an assumption is made that the MAC layer senders transmit a packet a maximum of $T_{\max}$ times before discarding it. $P_{\rm pkt}$ represents the failure probability of a packet transmitted just one time. However, packet retransmission increases the probability of the packet being successfully received. Therefore, the perceived probability of a packet being correctly received is given as

$$P_{\rm correct} = \sum_{i=1}^{T_{\max}} (1 - P_{\rm pkt}) \times (P_{\rm pkt})^{i-1} = 1 - (P_{\rm pkt})^{T_{\max}} \tag{1}$$

Consequently, the effective failure probability of a packet is equal to

$$P_{T_{\max}} = 1 - P_{\rm correct} = (P_{\rm pkt})^{T_{\max}} \tag{2}$$

In general, when the packet loss rate is low, the number of packet retransmissions is also low, and vice versa. For example, assume that a total of $(N_{{\rm pkt}\_i})$ packets of the ith video block are to be sent by the sender. Furthermore, assume that to successfully transmit all of these video packets, the total number of packet retransmissions is equal to $({\rm RT}_{{\rm total}\_i})$. According to the (2), with the limit of packet retransmission time $(T_{\max})$, the number of lost packet is $(N_{{\rm pkt}\_i} \times P_{\rm pkt}^{T\max})$. We can express the equation of the total number of correctly received packet as below:

$$\begin{aligned}(N_{{\rm pkt}\_i} + {\rm RT}_{{\rm total}\_i}) \times (1 - P_{\rm pkt})& \\ = N_{{\rm pkt}\_i} - N_{{\rm pkt}\_i} \times (P_{\rm pkt})^{T_{\max}}&\end{aligned} \tag{3}$$

Following a process of manipulation, it can be shown that

$$\frac{(N_{{\rm pkt}\_i} + {\rm RT}_{{\rm total}\_i})}{(N_{{\rm pkt}\_i})} = \frac{\left(1 - P_{\rm pkt}^{T_{\max}}\right)}{(1 - P_{\rm pkt})} = \sum_{i=0}^{T_{\max}-1} (P_{\rm pkt})^i \tag{4}$$

Equation (4) provides the means to estimate the packet loss rate of the ith video block given a knowledge of the total number of packets sent $(N_{{\rm pkt}\_i})$, the total number of packet retransmissions $({\rm RT}_{{\rm total}\_i})$, and the upper bound on the maximum number of packet retransmissions by the MAC senders $(T_{\max})$. To prevent excessive oscillations of the estimated packet loss rate in response to changes in the wireless network condition, the ERED-FEC controller weights the estimated packet loss rate as follows:

$$\begin{aligned}P_{{\rm estimation},i} = {\rm RT}_{\rm weight} \times P_i + (1 - {\rm RT}_{\rm weight})& \\ \times P_{{\rm estimation},i-1}&\end{aligned} \tag{5}$$

where ${\rm RT}_{\rm weight}$ is a weighting factor and P_i is the estimated packet loss rate of the ith video block. Note that in implementing the ERED-FEC scheme, $P_{\rm estimation}$ is computed following the arrival of a complete block of video packets as the current weighted moving average of the number of retransmission times.

2) Analytical Model for Estimating Effective Packet Loss Rate With FEC Recovery: Consider a video streaming file containing a total of $N\_{\rm pkt}$ video packets. Assume that each transmission block comprises k video packets. As a result, the video file is transmitted in a total of approximately $({\rm N\_pkt}/{\rm k})$ blocks. Given an effective packet loss rate of $P_{T\max}$, the probability that a block cannot be recovered at the receiver end is given by

$$P_{\rm block} = 1 - \sum_{i=k}^{k+h} C_i^{k+h} \times (1 - P_{T\max})^i \times (P_{T\max})^{k+h-i} \tag{6}$$

where C_i^{k+h} denotes all possible combinations of the i packets successfully received within a block. Since there are approximately $({\rm N\_pkt}/{\rm k})$ blocks in the video file, the expected number of packets received from the successfully recovered blocks at the receiver end can be determined as

$$\frac{N\_{\rm pkt}}{k} \times (1 - P_{\rm block}) \times k \tag{7}$$

As shown in (7), a successfully recovered block is assumed to yield k video packets, irrespective of whether the actual number of packets received is equal to or greater than k. Furthermore, the expected number of source packets received from the unrecovered blocks is given by

$$\begin{aligned}\frac{N\_{\rm pkt}}{k} \times P_{\rm block} \times \Bigg(\Bigg(\sum_{i=0}^{k-1} C_i^{k+h} \times (1 - P_{T\max})^i& \\ \times (P_{T\max})^{k+h-i} \times i\Bigg) \times \frac{k}{k+h}\Bigg)&\end{aligned} \tag{8}$$

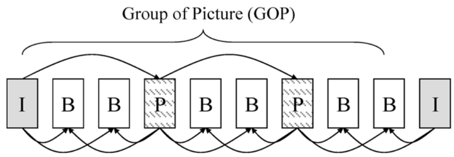

Fig. 5. Illustrative MPEG Group of Picture ($\mathrm{N}=9$ and $\mathrm{M}=3$).

Note that there could be some of source and redundant packets received in the unrecovered blocks, but the received redundant packets are assumed to be discarded. Therefore, in (8), the term $\mathrm{k}/(\mathrm{k}+\mathrm{h})$ means the source packets are considered only into the amount of received packets.

The expected number of packets lost during transmission is given by $[N\_\mathrm{pkt}-(\mathrm{Eq.}\ (7)+\mathrm{Eq.}\ (8))]$, and thus the effective packet loss rate (P_{FEC}) for a video file delivered over a wireless link with FEC protection can be formulated as shown in (9) at the bottom of the page. Equation (9) provides a simple model for predicting the effective packet loss rate (P_{FEC}) in a wireless channel with FEC protection given a knowledge of the packet loss rate and the FEC rate (n, k).

3) Analytical Model for Evaluating Perceived Quality of MPEG-4 Video Stream: This section presents a model for evaluating the impact of packets losses on the percentage of successfully decoded frames at the receiver end. To calculate the error propagation due to packet losses, the interdependencies of the coded frames must be considered. The MPEG-4 standard defines three frame types for compressed video streaming, namely I frame (Intra-coded), P frame (Predictive-coded) and B frame (Bi-directionally coded) [27]. I frames are encoded and decoded independently of any other frames in the sequence. However, P frames are encoded based on the preceding I or P frame in the video sequence, while B frames are encoded based on both the preceding and the succeeding I or P frames. In MPEG-4 coding, the video sequence is decomposed into small units designated as Groups of Pictures (GOPs), where each GOP contains consecutive pictures and begins with an I frame. The GOP structure is defined by two parameters, i.e., G (N, M), where N is the I-to-I frame distance and M is the I-to-P frame distance. Fig. 5 presents an illustrative example of a GOP with parameters G (9, 3). The non-recoverable frames in MPEG-4 video transmissions can be classified as either directly undecodable or indirectly undecodable. In the former case, the video frame cannot be reconstructed since an insufficient number of packets within the frame have been received. By contrast, in the latter case, the video frame cannot be reconstructed since some of the frames on which it depends are directly undecodable.

The delivered quality of MPEG video streaming over lossy networks is commonly evaluated using the Decodable Frame Rate (DFR) metric, defined as the sum of the decodable video frames over the total number of video frames sent by the video source, i.e.,

$$\mathrm{DFR}=\frac{N_{\mathrm{total\_dec}}}{N_{\mathrm{total\_send}}}=\frac{N_{\mathrm{dec}\_I}+N_{\mathrm{dec}\_P}+N_{\mathrm{dec}\_B}}{N_{\mathrm{total}\_I}+N_{\mathrm{total}\_P}+N_{\mathrm{total}\_B}} \tag{10}$$

Assuming that an I frame consists of approximately C_I packets, the probability that the I frame is decodable is given by

$$S(I)=(1-P_{\mathrm{FEC}})^{C_I} \tag{11}$$

where P_{FEC} is the effective packet loss rate for a video file delivered over a wireless link with FEC protection (9).

Consequently, the expected number of decodable I frames over the entire video sequence is given by

$$N_{\mathrm{dec}-I}=(1-P_{\mathrm{FEC}})^{C_I}\times N_{\mathrm{GOP}} \tag{12}$$

where N_{GOP} is the total number of GOPs in the video sequence.

In a GOP, the P frame is decodable only if the preceding I or P frame is decodable and all the packets which belong to the P frame are also decodable. Assuming that the total number of P frames in a GOP is denoted as N_p, the probability of each P frame in the GOP being decodable is given as

$$\begin{aligned}
\mathrm{S}(\mathrm{P}_1)&=(1-P_{\mathrm{FEC}})^{C_I}\times(1-P_{\mathrm{FEC}})^{C_P}\\
&=(1-P_{\mathrm{FEC}})^{C_I+C_P}\\
\mathrm{S}(\mathrm{P}_2)&=(1-P_{\mathrm{FEC}})^{C_I}\times(1-P_{\mathrm{FEC}})^{C_P}\times(1-P_{\mathrm{FEC}})^{C_P}\\
&=(1-P_{\mathrm{FEC}})^{C_I+2\times C_P}\\
&\cdots\\
\mathrm{S}(\mathrm{P}_{\mathrm{N_P}})&=(1-P_{\mathrm{FEC}})^{C_I}\times(1-P_{\mathrm{FEC}})^{N_P\times C_P}\\
&=(1-P_{\mathrm{FEC}})^{C_I+N_P\times C_P}
\end{aligned} \tag{13}$$

where C_p denotes the average number of packets within each P frame. Thus, the total expected number of decodable P frames over the entire video sequence is equal to

$$N_{\mathrm{dec}-P}=(1-P_{\mathrm{FEC}})^{C_I}\times\sum_{j=1}^{N_P}(1-P_{\mathrm{FEC}})^{j\times C_P}\times N_{\mathrm{GOP}}. \tag{14}$$

In a GOP, the B frame is decodable only if the preceding and succeeding I or P frames are both decodable and all of the

$$\begin{aligned}
P_{\mathrm{FEC}}&=\frac{[N_\mathrm{pkt}-(\mathrm{Eq.}\ (7)+\mathrm{Eq.}\ (8))]}{N_\mathrm{pkt}}\\
&=P_{\mathrm{block}}-\frac{\left(\sum_{i=0}^{k-1}C_i^{k+h}\times(1-P_{T\max})^i\times(P_{T\max})^{k+h-i}\times i\right)}{k+h}
\end{aligned} \tag{9}$$

packets belonging to the B frame are also decodable. Since consecutive B frames have the same dependency throughout the GOP structure, consecutive B frames can be regarded for convenience as belonging to the same B group. The last B frame/group in a GOP is encoded based on the preceding P frame and succeeding I frame. In other words, it is influenced by both the preceding I frame and the succeeding I frame. Therefore, the probability that the B frame/group can be successfully decoded is given by

$$\begin{aligned} \mathrm{S}(\mathrm{B}_1) &= (1-P_{\mathrm{FEC}})^{C_I} \\ &\quad \times (1-P_{\mathrm{FEC}})^{C_P} \times (1-P_{\mathrm{FEC}})^{C_B} \\ \mathrm{S}(\mathrm{B}_2) &= (1-P_{\mathrm{FEC}})^{C_I} \times (1-P_{\mathrm{FEC}})^{2\times C_P} \\ &\quad \times (1-P_{\mathrm{FEC}})^{C_B} \\ &\cdots \\ \mathrm{S}\left(\mathrm{B}_{\frac{\mathrm{N}}{\mathrm{M}}-1}\right) &= (1-P_{\mathrm{FEC}})^{C_I} \times (1-P_{\mathrm{FEC}})^{\left(\frac{N}{M}-1\right)\times C_P} \\ &\quad \times (1-P_{\mathrm{FEC}})^{C_B} \\ \mathrm{S}\left(\mathrm{B}_{\frac{\mathrm{N}}{\mathrm{M}}}\right) &= (1-P_{\mathrm{FEC}})^{2\times C_I} \times (1-P_{\mathrm{FEC}})^{\left(\frac{N}{M}-1\right)\times C_P} \\ &\quad \times (1-P_{\mathrm{FEC}})^{C_B} \end{aligned} \tag{15}$$

where C_B is the average number of packets within each B frame. Within each GOP (N, M), there exist $(\mathrm{M}-1)$ subsequent B frames (considered in the analysis above as a single frame). By aggregating the respective probabilities of all the B frames, the total expected number of correctly decodable B frames over the entire video sequence is obtained as

$$\begin{aligned} N_{\mathrm{dec}-B} = &\left((1-P_{\mathrm{FEC}})^{C_I+N_P\times C_P} \right. \\ &\left. + \sum_{j=1}^{N_P} (1-P_{\mathrm{FEC}})^{j\times C_P} \times (1-P_{\mathrm{FEC}})^{C_B} \right) \\ &\times (M-1) \times (1-P_{\mathrm{FEC}})^{C_I+C_B} \times N_{\mathrm{GOP}} \end{aligned} \tag{16}$$

The results obtained from (12), (14) and (16) for the total number of decodable I, P and B frames, respectively, can be substituted into the analytical model given in (10) to evaluate the delivered quality of the video stream. Moreover, the analytical model can be used by the ERED-FEC mechanism to determine the FEC parameters (n, k) required to guarantee the specified video quality over a lossy wireless network.

4) Adjustment of FEC Packets via RED Mechanism: Having determined the FEC redundancy rate (n, k) in accordance with the channel condition, the ERED-FEC mechanism dynamically adjusts the number of redundant packets in accordance with the current network load so as to avoid congestion-induced packet losses. Table I presents the pseudo-code of the ERED-FEC algorithm. (Note that the various notations used in the pseudo-code are summarized in Table II.) As shown, once a complete block of video packets has been received, the AP estimates the packet loss rate and determines the maximum number of FEC redundant packets required to satisfy the specified QoS for the video stream $(\mathrm{Max\_FEC\_pkt})$ using the analytical model given in (9) and (10). The actual number of redundant packets to be injected into the network $(\mathrm{Final\_FEC\_pkt})$ is then determined by computing the current weighted moving average of the AP queue length (Q_{length}) and comparing the result with two threshold values, namely $\mathrm{Th_{low}}$ and $\mathrm{Th_{high}}$. If Q_{length} is less than $\mathrm{Th_{low}}$, the network is lightly loaded, and thus the AP specifies the number of redundant packets as $(Final\_FEC\_pkt = \mathrm{Max}\_FEC\_pkt)$. Conversely, if Q_{length} exceeds $\mathrm{Th_{high}}$, the network is heavily loaded, and thus the AP specifies $(Final\_FEC\_pkt = 0)$ in order to minimize the risk of packets being dropped at the AP queue. If Q_{length} falls between $\mathrm{Th_{low}}$ and $\mathrm{Th_{high}}$, the AP determines the appropriate number of redundant packets by scaling $\mathrm{Max\_FEC\_pkt}$ in accordance with the extent by which Q_{length} exceeds $\mathrm{Th_{low}}$.

TABLE I
PSEUDO-CODE OF ERED-FEC MECHANISM

```
When a block of packets arrives:
/* estimation of packet loss rate in wireless network */
P_estimation,i = RT_weight × P_i + (1 − RT_weight) × P_estimation,i-1
/* determination of FEC redundancy rate */
Max_FEC_pkt = FEC_model (QoS requirement)
/* adjust the number of redundant FEC packets in accordance with the
network load */
Q_length,i = Q_weight × Q_i + (1 − Q_weight) × Q_length,i-1
if (Q_length < Th_low)
  Final_FEC_pkt = Max_FEC_pkt;
else if (Q_length < Th_high)
   Final_FEC_pkt = Max_FEC_pkt × (Th_high − Q_length) / (Th_high − Th_low);
else
   Final_FEC_pkt = 0;
```

To conclude, by dynamically adjusting the FEC rate in accordance with not only the packet loss rate (as indicated by the number of MAC layer retransmissions), but also the wireless network load (as indicated by the AP queue length), the ERED-FEC mechanism enhances the quality of the delivered video while simultaneously maintaining the network performance.

IV. RESULTS AND DISCUSSION

A. Experimental Environment and Setting

The validity of the proposed ERED-FEC mechanism was demonstrated by comparing the analytical results for the estimated packet loss rate and DFR with the results obtained from NS-2 simulations [28]. A further series of simulations was then performed using the topology shown in Fig. 6 to compare the performance of the proposed scheme with that of a conventional static FEC (SFEC) scheme and two AP-based FEC schemes, namely RED-FEC [20] and ACFEC [21].

In performing the simulations, the video server transmitted the video stream over the Internet, and the video packets were delivered to the receiver by a wireless AP. The video trace, "Highway" [29], was encoded using the MPEG-4 standard in a QCIF format with a GOP structure of IBBPBBPBB $(\mathrm{N}=9, \mathrm{M}=3)$. The video was streamed at a rate of 30 frames/sec with each frame divided into transmission units of 1000 bytes. Table III shows the total number of I, P and

TABLE II
NOTATIONS USED IN ERED-FEC ALGORITHM

Terms	Definitions
P_{pkt}	Packet loss rate in wireless network
$P_{correct}$, P_{Tmax}	Effective received rate, loss rate of packet given maximum number of retransmissions equal to T_{max}
$P_{estimation}$	Estimated packet loss rate
P_{block}	Probability of block being unrecovered
P_{FEC}	Effective packet loss rate with FEC recovery
RT_{total_i}	*Total number of packet retransmissions of the ith video block*
T_{max}	Maximum number of retransmissions for a packet
RT_{weight}	Weighted value of number of retransmissions
k, h	Number of source packets, redundant packet in block
$N_{pkt\text{-}i}$	Total number of video packets of the ith video block
N_pkt	Total number of video packets
N_{dec_I}, N_{dec_P}, N_{dec_B}	Expected number of decodable I, P and B frames
N_{total_send}	Total number of sent video frames
N_{GOP}	Total number of GOPs in video sequence
N_P, N_B	Total number of P and B frames in GOP
C_I, C_P, C_B	Mean number of packets in I, P and B frames
Q_{length}	Weighted moving average of AP queue length
Q_{weight}	Weighted value of queue length
$Q_{current}$	Current queue length
Q_{max}	Maximum queue size
Th_{low}, Th_{high}	Lower threshold, upper threshold for queue length
Max_FEC_pkt, $Final\_FEC\_pkt$	Maximum number, final number of injected FEC packets

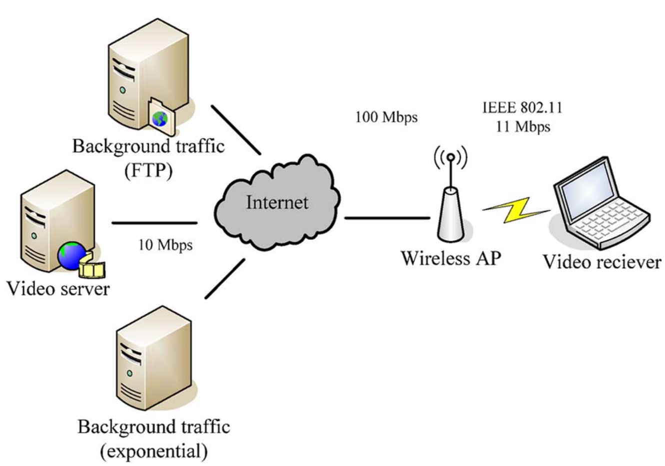

Fig. 6. Simulation topology.

B frames and packets in the video trace. The trace was transmitted using a unicast transmission technique ($T_{\max} = 4$). The number of video packets within each FEC block was specified as eight ($k = 8$). In addition to the video source, two background traffic flows were also streamed, namely FTP traffic, transmitted using TCP packets, and exponential traffic, transmitted using UDP packets. Finally, it was assumed that the wired segment of the video delivery path was loss free and that no packets were lost as a result of channel contentions in the wireless network.

TABLE III
NUMBER OF VIDEO FRAMES AND PACKETS IN VIDEO TRACE (HIGHWAY)

Total frame	2000	Total packet	2684	Format	QCIF
I frame	223	I packet	828	C_I	3.71
P frame	445	P packet	514	C_P	1.15
B frame	1332	B packet	1342	C_B	1.01

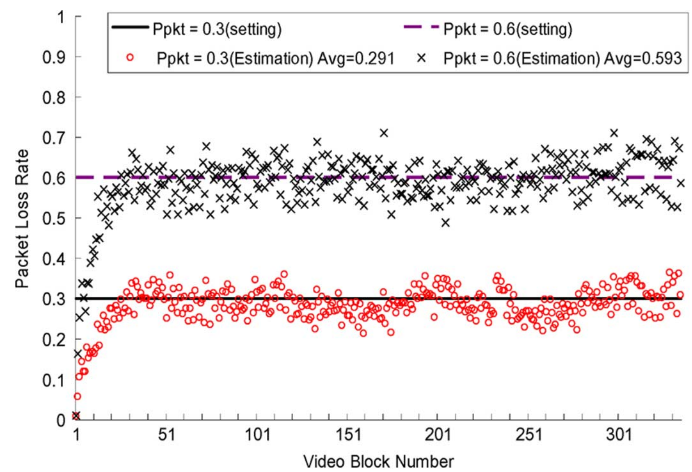

Fig. 7. Estimated packet loss rate in wireless network with constant loss rate. (Constant loss rate, $P_{\rm pkt} = 0.3, 0.6$.)

B. Validation of FEC Analytical Model

Estimated Packet Loss Rate: The performance of the packet loss rate estimation model was verified by comparing the results obtained from (4) with those obtained from NS-2 simulations. As shown in Fig. 7, a good agreement was obtained between the two sets of results at both low ($P_{\rm pkt} = 0.3, P_{\rm Tmax} = 0.0081$) and high ($P_{\rm pkt} = 0.6, P_{\rm Tmax} = 0.1296$) packet loss rates.

Figs. 8 and 9 show the estimated packet loss rate in wireless networks with a varying channel condition. In Fig. 8, the wireless channel is in a good condition ($P_{\rm pkt} = 0.3$) for the first half of the simulation period, but then changes to a bad condition ($P_{\rm pkt} = 0.6$) for the remainder of the simulation. In Fig. 9, the wireless channel condition commences in a good condition ($P_{\rm pkt} = 0.3$), but then changes to a bad condition ($P_{\rm pkt} = 0.6$) at $t = 18$ s. The channel remains in a bad condition until $t = 35$ s, at which point it returns to a good condition. The channel condition remains good until $t = 52$ s, but then reverts to a bad condition for the remainder of the simulation period. The results presented in Figs. 8 and 9 show that irrespective of the rate at which the channel condition varies, the estimated packet loss rate obtained using the proposed analytical model is in good agreement with the actual packet loss rate observed in the simulations.

Effect of Packet Losses on Delivered Video Quality: In this section, the validity of the analytical model given in (9) and (10) is examined by comparing the analytical results for the effective packet loss rate ($P_{\rm FEC}$) and Decodable Frame Rate (DFR) with those obtained from NS-2 simulations. Note that in performing

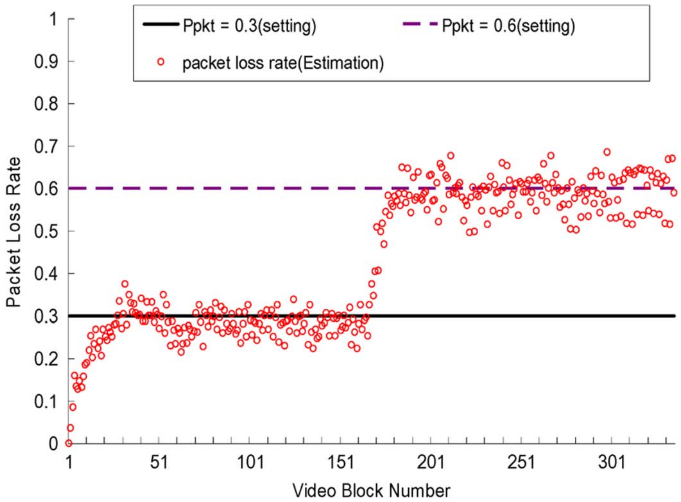

Fig. 8. Estimated packet loss rate in wireless network with variable loss rate. (Variable loss rate, t: 0–33 s: $P_{\rm pkt} = 0.3$; t: 34–66 s: $P_{\rm pkt} = 0.6$.)

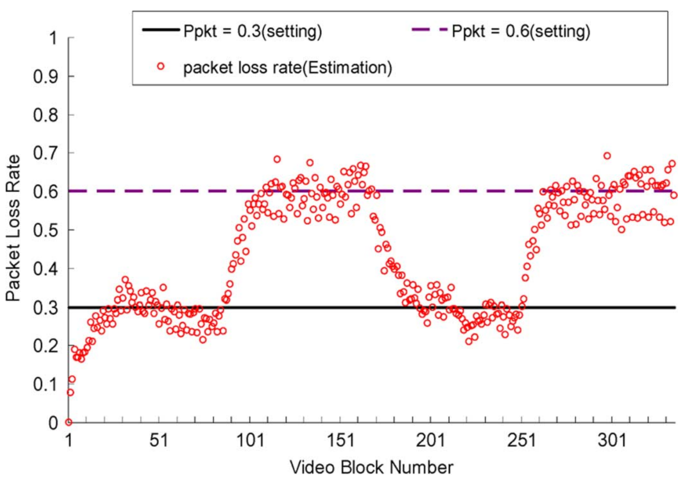

Fig. 9. Estimated packet loss rate in wireless network with variable loss rate. (Variable loss rate, t: 0–17 s and 35–51 s: $P_{\rm pkt} = 0.3$; t: 18–34 s and 52–66 s: $P_{\rm pkt} = 0.6$.)

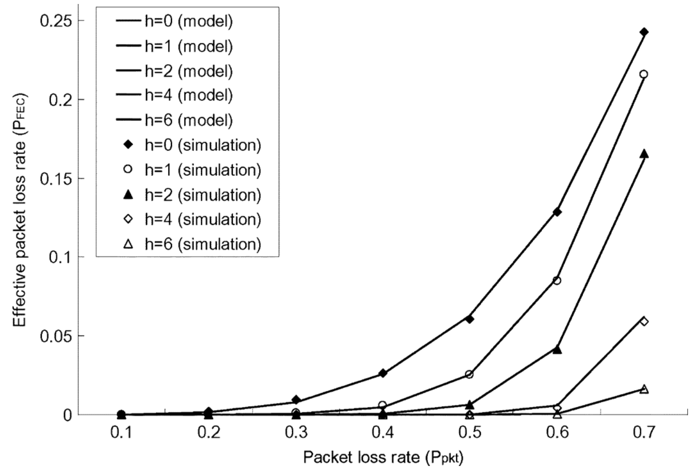

Fig. 10. Variation of effective packet loss rate with actual packet loss rate as function of FEC redundancy rate.

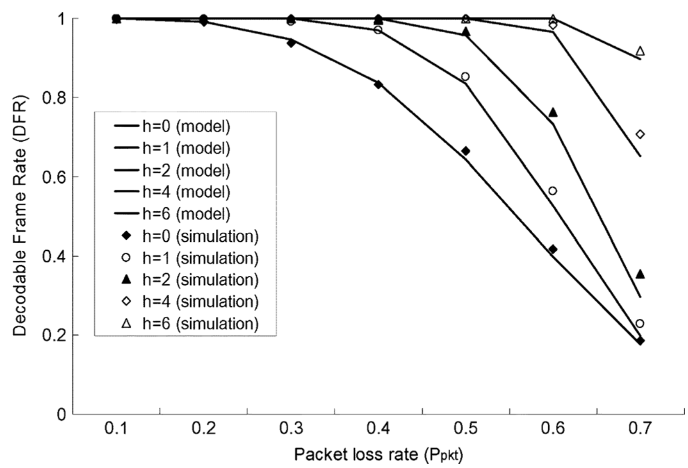

Fig. 11. Variation of DFR with packet loss rate as function of FEC redundancy rate.

the simulations, the video stream was transmitted without any background traffic. Fig. 10 compares the analytical results (solid lines) and numerical results (symbols) for the variation of the effective packet loss rate ($P_{\rm FEC}$) with the actual packet loss rate ($P_{\rm pkt}$). Note that the results are presented for five different FEC redundancy rates (i.e., $h = 0, 1, 2, 4$ and 6). Note also that each video block contains eight packets in every case (i.e., $k = 8$). As expected, the results show that the effective packet loss rate ($P_{\rm FEC}$) increases with an increasing actual packet loss rate ($P_{\rm pkt}$). In addition, for a given packet loss rate ($P_{\rm pkt}$), the effective packet loss rate ($P_{\rm FEC}$) decreases with an increasing number of FEC redundant packets.

Fig. 11 shows the results obtained from the analytical model and simulations for the DFR (10) in a wireless network with FEC protection. As expected, a higher FEC redundancy rate results in a lower effective packet loss rate, and thus gives rise to an improved transmission quality. A good agreement is observed between the analytical results and the simulation results. As a result, the validity of the analytical model is confirmed. Accordingly, as shown in Table IV, the analytical model enables the ERED-FEC controller to determine the FEC redundancy rate required to satisfy a given QoS for video streams over lossy wireless networks. For example, given an actual packet loss rate of $P_{\rm pkt} \leq 0.5$ ($P_{\rm Tmax} \leq 0.0625$), a minimum DFR of 0.8 can be guaranteed by setting the FEC redundancy rate to $h = 1$. Moreover, if the packet loss rate ($P_{\rm pkt}$) increases to 0.7 ($P_{\rm Tmax} \geq 0.2401$), the FEC redundancy rate should be assigned a value of h > 5 in order to maintain the same video quality (i.e., DFR > 0.8). Table IV summarizes the results obtained from the ERED-FEC mechanism for the FEC settings, effective packet loss rate, and DFR given various values of $P_{\rm pkt}, T_{\rm max}$, and $P_{\rm Tmax}$. It can be seen that for an actual packet loss rate ($P_{\rm pkt}$) of 0.3 ($P_{\rm Tmax} = 0.0081$), the ERED-FEC scheme generates two FEC packets per block ($h = 2$). Moreover, the number of FEC packets per block is increased to seven ($h = 7$) when the packet loss rate increases to 0.6 ($P_{\rm Tmax} = 0.1296$) in order to maintain the same video quality.

C. Performance Analysis of ERED-FEC Mechanism

The performance of the ERED-FEC mechanism is evaluated using the parameter settings shown in Table V. Figs. 12 to 14

TABLE IV
FEC SETTINGS AND VIDEO QUALITY METRICS OBTAINED FROM ERED-FEC MECHANISM FOR VARIOUS NETWORK CONDITIONS

P_{pkt}	T_{max}	P_{Tmax}	(n, k)	P_{FEC}	DFR
0.1	4	0.0001	(8, 8)	0.0001	1.0
0.2	4	0.0016	(9, 8)	0.00002	1.0
0.3	4	0.0081	(10, 8)	0.000018	1.0
0.4	4	0.0256	(11, 8)	0.000045	1.0
0.5	4	0.0625	(13, 8)	0.000032	1.0
0.6	4	0.1296	(15, 8)	0.000118	1.0
0.7	4	0.2401	(16, 8)	0.003245	0.981

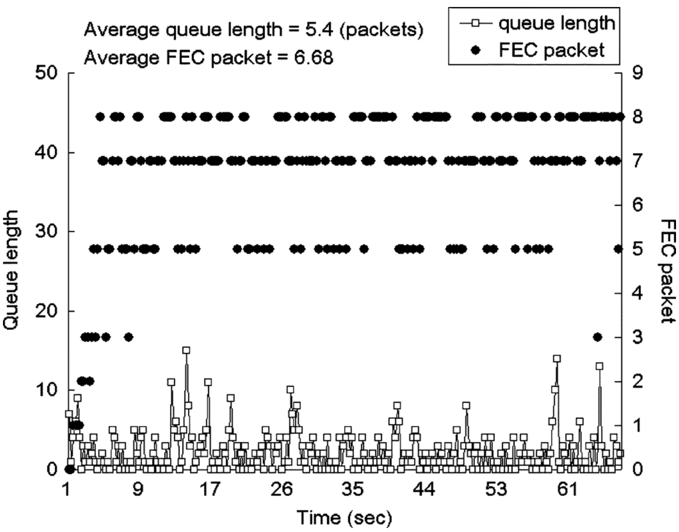

Fig. 12. Correlation between queue length and number of injected FEC redundant packets under light load condition.

show the variation of the AP queue length and the number of generated FEC redundant packets under three different network load conditions, i.e., light, heavy and variable, respectively. Note that the packet loss rate $(P_{\rm pkt})$ is specified as 0.6 and the FEC rate (n, k) is set as (15, 8) in every case. As shown in Fig. 12, under a light load, the average queue length is approximately 5.4 packets and the mean number of injected FEC redundant packets per block is equal to 6.68. Under a heavy load, the average queue length is 39.8 packets and the mean number of FEC redundant packets per block is reduced to 1.54 (see Fig. 13). Finally, under a variable load, the mean queue length is 15.8 packets and the mean number of FEC redundant packets per block is equal to 4.66 (see Fig. 14).

Fig. 15 shows the correlation between the AP queue length and the number of FEC redundant packets when the FEC redundancy rate is adjusted in accordance with changes in the channel condition. Specifically, for a lower channel error rate of $P_{\rm pkt} = 0.3$ ($t =$ 0–35 s), the ERED-FEC mechanism specifies the FEC rate (n, k) as (10, 8) in accordance with Table IV, while for a higher channel error rate of $P_{\rm pkt} = 0.6$ ($t =$ 35–66 s), the FEC rate (k, h) is adjusted to (15, 8). The results confirm the ability of the proposed ERED-FEC mechanism to modify the FEC redundancy rate adaptively in accordance with changes in the network condition.

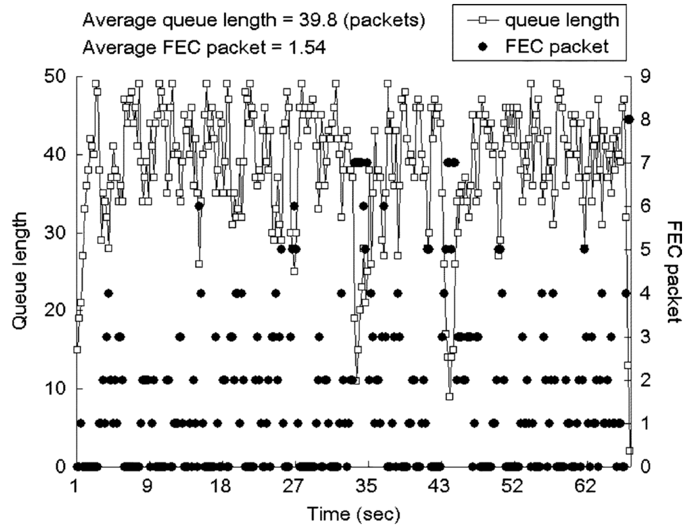

Fig. 13. Correlation between queue length and number of injected FEC redundant packets under heavy load condition.

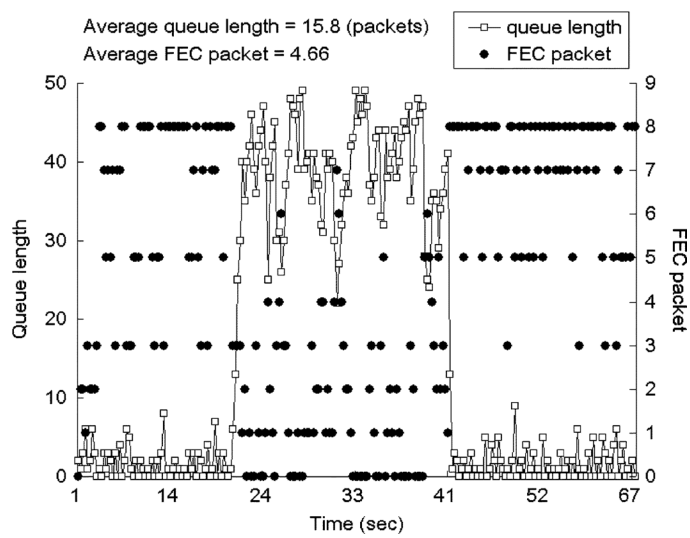

Fig. 14. Correlation between queue length and number of injected FEC redundant packets under variable load condition.

TABLE V
FEC SETTINGS AND VIDEO QUALITY METRICS OBTAINED FROM ERED-FEC MECHANISM FOR VARIOUS NETWORK CONDITIONS

RT_{weight}	0.9	$Q_{\_max}$	50 packets
Q_{weight}	0.9	Th_{low}	25 packets (50% $Q_{\_max}$)
T_{max}	4	Th_{high}	40 packets (80% $Q_{\_max}$)

D. *Performance Comparison of AP-Based FEC Mechanisms*

In this section, the performance of the proposed ERED-FEC scheme is compared with that of three existing AP-based schemes, namely SFEC, RED-FEC [20] and ACFEC [21]. The basic characteristics of the four schemes are compared in Table VI. As described previously, the ERED-FEC mechanism determines the number of FEC redundant packets in accordance with both the network loss rate and the traffic load. By contrast, the SFEC scheme ignores both the packet loss rate and the

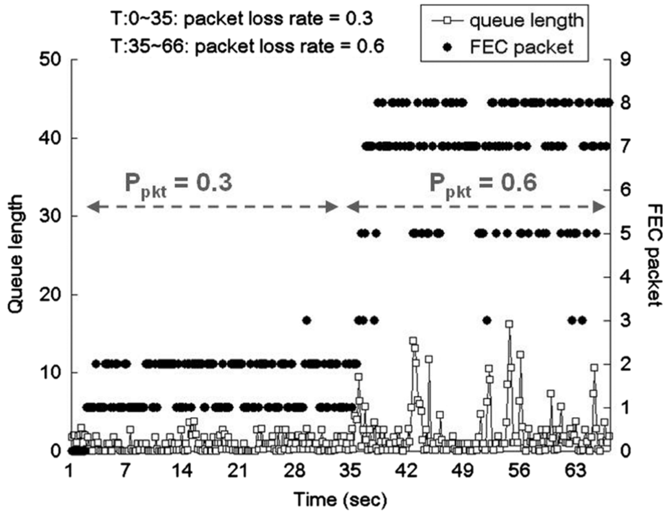

Fig. 15. Correlation between queue length and number of injected FEC redundant packets when redundancy rate is tuned adaptively in accordance with changes in the channel condition.

TABLE VI
COMPARISON OF AP-BASED FEC MECHANISM AND CORRESPONDING PARAMETER SETTINGS

FEC mechanism	Packet loss rate	Network traffic load	Parameter setting
SFEC	Not considered	No congestion avoidance	(n, k) = (11, 8)
RED-FEC [20]	Not considered	Congestion avoidance	(n, k) = (11, 8) Th_{low} = 50%Q_max Th_{high} = 80%Q_max
ACFEC [21]	Considered	No congestion avoidance	k = 8 h = (0, 1, 2, 3, 5, 7, 8)
ERED-FEC	Considered	Congestion avoidance	k = 8 h = (0, 1, 2, 3, 5, 7, 8) Th_{low} = 50%Q_max Th_{high} = 80%Q_max

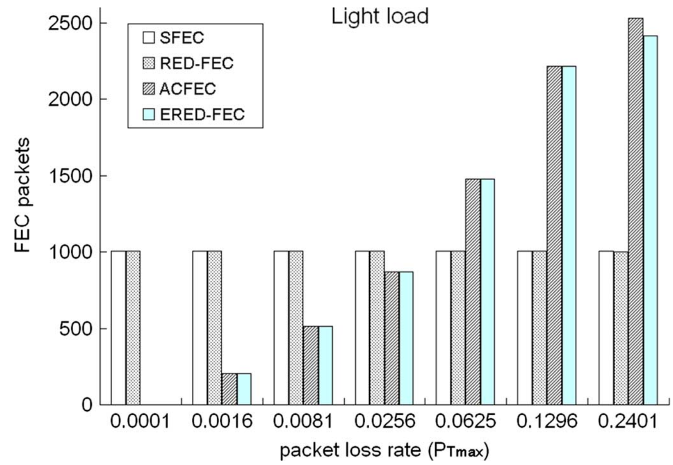

Fig. 16. Variation of FEC redundant packets with packet loss rate in various AP-based FEC mechanisms under light load condition.

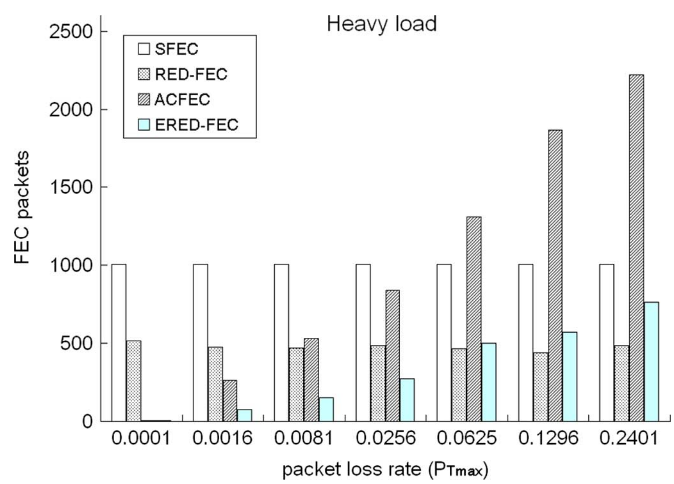

Fig. 17. Variation of FEC redundant packets with packet loss rate in various AP-based FEC mechanisms under heavy load condition.

traffic load, while the RED-FEC and ACFEC schemes consider either the packet loss rate or the traffic load, but not both.

Figs. 16 and 17 compare the number of FEC redundant packets generated by the four schemes under light and heavy traffic loads, respectively. In the case of a light load, the number of FEC redundant packets generated by the SFEC and RED-FEC schemes remains approximately constant as the packet loss rate increases since neither scheme considers the channel condition when evaluating the FEC redundancy rate (see Fig. 16). By contrast, the ACFEC and ERED-FEC schemes both consider the packet loss rate when determining the FEC redundancy rate, and thus for both algorithms, the number of redundant packets increases with an increasing packet loss rate. In the case of a heavy load, the number of FEC redundant packets generated by the SFEC and ACFEC algorithms is the same as that generated for a light traffic load since both algorithms ignore the effects of congestion when determining the FEC redundancy rate (see Fig. 17). However, in both the RED-FEC scheme and the ERED-FEC scheme, the number of redundant packets is reduced in order to avoid overloading the network.

Figs. 18 to 21 compare the DFR and Peak Signal-to-Noise Ratio (PSNR) performance of the four schemes under light and heavy load conditions. Under light loads, the DFR and PSNR obtained using the proposed ERED-FEC scheme at low values of the packet loss rate are similar to those obtained using the SFEC, RED-FEC and ACFEC schemes, respectively (see Figs. 18 and 19). At higher values of the loss rate ($P_{\mathrm{Tmax}} = 0.1296$ and 0.2401), the DFR and PSNR obtained using the SFEC and RED-FEC schemes reduce significantly since neither scheme considers the packet loss rate when determining the FEC redundancy rate. However, in the ACFEC and ERED-FEC schemes, no appreciable degradation in the DFR or PSNR occurs since both schemes take account of the packet loss rate when computing the number of FEC redundant packets per block. Under heavy loads, the DFR and PSNR performance of the ERED-FEC scheme is notably better than that of the SFEC, RED-FEC and ACFEC schemes since, in contrast to these schemes, the ERED-FEC mechanism determines the

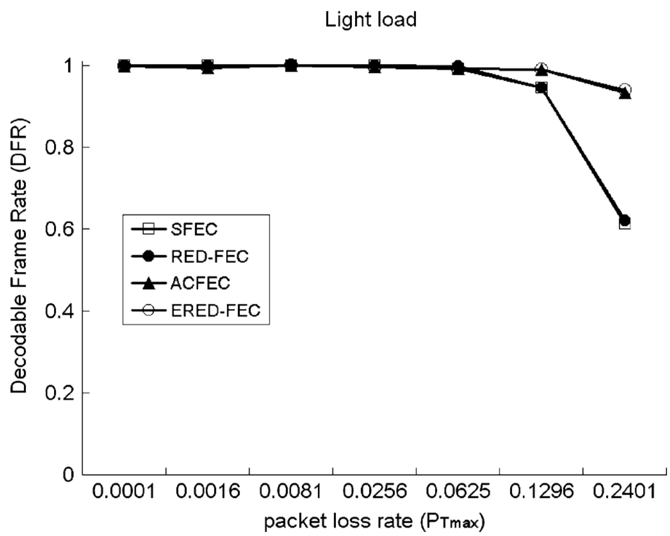

Fig. 18. Variation of DFR with packet loss rate for various AP-based FEC mechanisms under light load condition.

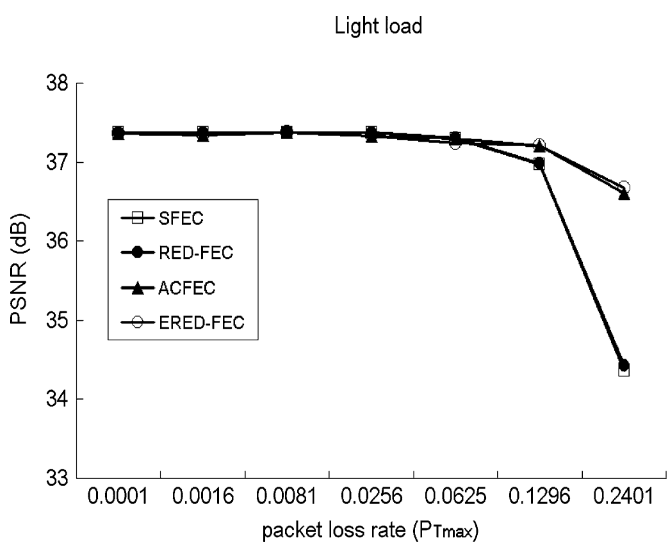

Fig. 19. Variation of PSNR with packet loss rate for various AP-based FEC mechanisms under light load condition.

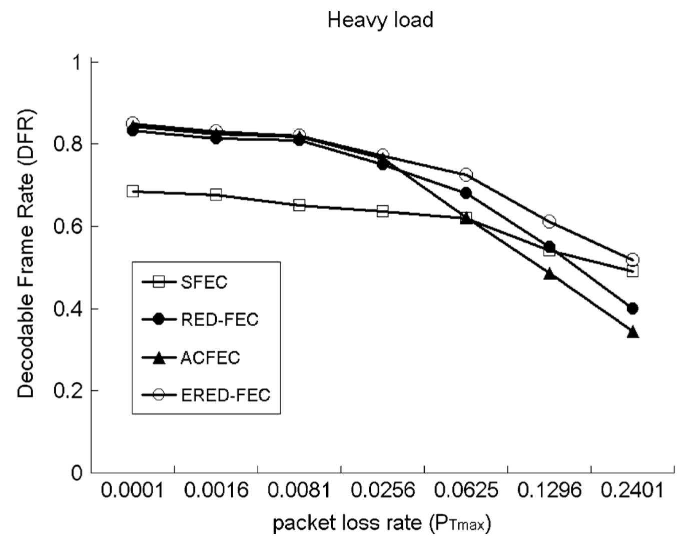

Fig. 20. Variation of DFR with packet loss rate for various AP-based FEC mechanisms under heavy load condition.

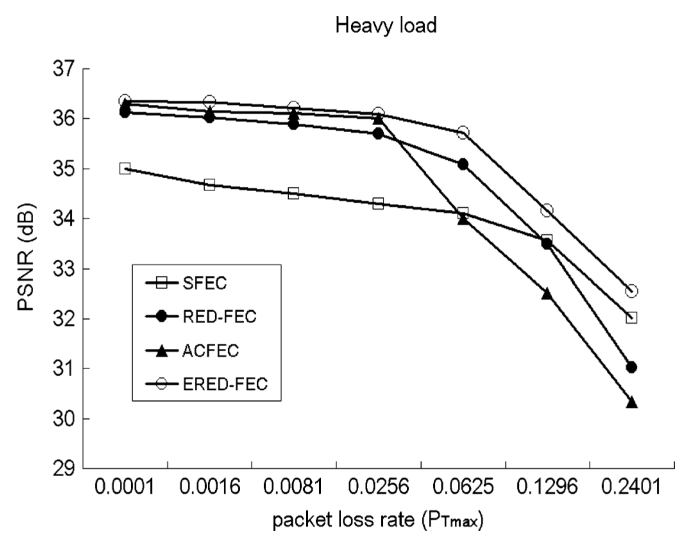

Fig. 21. Variation of PSNR with packet loss rate for various AP-based FEC mechanisms under heavy load condition.

FEC redundancy rate in accordance with both the packet loss rate and the network traffic load.

V. Conclusion

This paper has presented an AP-based FEC mechanism (ERED-FEC) for improving the quality of video transmissions over WLANs. In contrast to many FEC schemes, in which the FEC rate is determined at the sender end on the basis of information provided by the receiver, in the FEC mechanism proposed in this study, the FEC redundancy rate is determined at the wireless access point (AP). Moreover, the FEC redundancy rate is calculated in accordance with both the wireless channel condition and the network traffic load. As a result, the ERED-FEC mechanism significantly improves the video quality without overloading the network with redundant packets. The experimental results have shown that the ERED-FEC scheme yields a higher Decodable Frame Rate (DFR) and Peak Signal-to-Noise Ratio (PSNR) than existing AP-based FEC mechanisms under both light and heavy network traffic loads.

In a future study, the recovery performance of the ERED-FEC mechanism will be further enhanced by utilizing an FEC interleaving/de-interleaving strategy. In addition, the feasibility of extending the ERED-FEC scheme to IEEE 802.11e [32], [33] and IEEE 802.16 (WiMAX) [34], [35] networks will also be addressed.

References

[1] D. J. Deng, C. H. Ke, Y. M. Huang, and H. H. Chen, "Contention window optimization for IEEE 802.11 DCF access control," *IEEE Trans. Wireless Commun.*, vol. 7, no. 12, pp. 5129–5135, Dec. 2008.

[2] K. J. Geun and M. M. Krunz, "Bandwidth allocation in wireless networks with guaranteed packet-loss performance," *IEEE/ACM Trans. Netw.*, vol. 8, pp. 337–349, 2000.

[3] H. Zhihai and X. Hongkai, "Transmission distortion analysis for real-time video encoding and streaming over wireless networks," *IEEE Trans. Circuits Syst. Video Technol.*, vol. 16, pp. 1051–1062, 2006.

[4] A. Nafaa, Y. Hadjadj-Aoul, and A. Mehaoua, "On interaction between loss characterization and forward error correction in wireless multimedia communication," in *Proc. IEEE Int. Conf. Communications, ICC 2005*, 2005, pp. 1390–1394.
[5] E. Maani and A. Katsaggelos, "Unequal error protection for robust streaming of scalable video over packet lossy networks," *IEEE Trans. Circuits Syst. Video Technol.*, vol. 20, pp. 407–416, 2010.
[6] A. Argyriou, "Cross-layer error control for multimedia streaming in wireless/wireline packet networks FEC scheme for TDM-OFDM based satellite radio broadcasting system," *IEEE Trans. Multimedia*, vol. 10, pp. 1121–1127, Oct. 2008.
[7] J. Dan, F. Pascal, and J. Aleksandar, "Forward error correction for multipath media streaming," *IEEE Trans. Circuits Syst. Video Technol.*, vol. 19, pp. 1315–1326, 2009.
[8] J. Paavola, H. Himmanen, T. Joela, J. Poikonen, and V. Ipatov, "The performance analysis of MPE-FEC decoding methods at the DVB-H link layer for efficient IP packet retrieval," *IEEE Trans. Broadcast.*, vol. 53, pp. 263–275, Mar. 2007.
[9] A. Nafaa, T. Ahmed, and A. Mehaoua, "Unequal and interleaved FEC protocol for robust MPEG-4 multicasting over wireless LANs," in *Proc. IEEE Conf. Computer and Communications, ICC2004*, Jun. 2004, vol. 3, pp. 1431–1435.
[10] I. V. Bajic, "Efficient cross-layer error control for wireless video multicast," *IEEE Trans. Broadcast.*, vol. 53, pp. 276–285, Mar. 2007.
[11] K. Takahata, N. Uchida, and Y. Shibata, "Packet error and frame rate controls for real time video stream over wireless LANs," in *Proc. Int. Conf. Distributed Computing Systems Workshops, ICDCSW2003*, May 2003, pp. 594–599.
[12] I. V. Bajic, "Efficient cross-layer error control for wireless video multicast," *IEEE Trans. Broadcast.*, vol. 53, pp. 276–285, 2007.
[13] K. Park and W. Wang, "AFEC: An adaptive forward error correction protocol for end-to-end transport of real-time traffic," in *Proc. Int. Conf. Computer Communications and Networks, ICCCN1998*, Oct. 1998, pp. 196–205.
[14] S.-H. G. Chan, X. Zheng, Q. Zhang, W.-W. Zhu, and Y.-Q. Zhang, "Video loss recovery with FEC and stream replication," *IEEE Trans. Multimedia*, vol. 8, pp. 370–381, Apr. 2006.
[15] X. Yang, C. Zhu, Z. G. Li, X. Lin, and N. Ling, "An unequal packet loss resilience scheme for video over the internet," *IEEE Trans. Multimedia*, vol. 7, pp. 753–765, Aug. 2006.
[16] M. Tun, K. K. Loo, and J. Cosmas, "Error-resilient performance of Dirac video codec over packet-erasure channel," *IEEE Trans. Broadcast.*, vol. 53, pp. 649–659, Sep. 2007.
[17] C. H. Lin, C. H. Ke, C. K. Shieh, and N. Chilamkurti, "An enhanced adaptive FEC mechanism for video delivery over wireless networks," in *Proc. Int. Conf. Networking and Services, ICNS2006*, 2006.
[18] H. Du, Y. Liu, C. Guo, and Y. Liu, "Research on adaptive FEC for video delivery over WLAN," in *Proc. Int. Conf. Wireless Communications, Networking and Mobile Computing, WiCOM2009*, 2009, pp. 4808–4811.
[19] P. C. Huang, K. C. Chu, H. F. Lo, W. T. Lee, and T. Y. Wu, "A novel adaptive FEC and interleaving architecture for H.264/SVC wireless video transmission," in *Proc. Int. Conf. Intelligent Information Hiding and Multimedia Signal Processing, IIHMSP2009*, 2009, pp. 989–992.
[20] C. H. Lin, C. K. Shieh, N. Chilamkurti, C. H. Ke, and W. S. Hwang, "A RED-FEC mechanism for video transmission over WLANs," *IEEE Trans. Broadcast.*, vol. 54, no. 3, pp. 517–524, Sep. 2008.
[21] L. Han, S. Park, S. Kang, and H. P. In, "An adaptive cross-layer FEC mechanism for video transmission over 802.11 WLANs," in *Proc. Int. Conf. Internet, ICI 2009*, Dec. 2009, pp. 209–215.
[22] C. H. Ke, C. H. Lin, C. K. Shieh, W. S. Hwang, and A. Ziviani, "Evaluation of streaming MPEG video over wireless channels," *J. Mobile Multimedia*, vol. 3, Mar. 2007.
[23] H. Koumaras, C. H. Lin, C. K. Shieh, and A. Kourtis, "A framework for end-to-end video quality prediction of MPEG video," *J. Visual Commun. Image Represent.*, vol. 21, no. 2, pp. 139–154, Feb. 2010.
[24] V. Guruswami and M. Sudan, "Improved decoding of Reed-Solomon and algebraic-geometric codes," *IEEE Trans. Inf. Theory*, vol. 45, pp. 1757–1767, 1999.
[25] V. Roca, Design, Evaluation and Comparison of Four Large Block FEC Codecs, LDPC, LDGM, LDGM Staircase and LDGM Triangle, Plus a Reed-Solomon Small Block FEC Codec, INRIA Res., rep. RR-5225, Jun. 2004.
[26] H. Wu, Y. Peng, K. Long, S. Cheng, and J. Ma, "Performance of reliable transport protocol over IEEE 802.11 wireless LAN: Analysis and enhancement," in *Proc. IEEE Int. Conf. Computer Communications, INFOCOM2002*, 2002, pp. 599–607.
[27] X. J. Dong and P. Varaiya, "Saturation throughput analysis of IEEE 802.11 wireless LANs for a lossy channel," *IEEE Commun. Lett.*, vol. 9, no. 2, pp. 100–102, 2005.
[28] Q. Ni, T. Li, T. Turletti, and Y. Xiao, "Saturation throughput analysis of error-prone 802.11 wireless networks," *Wiley J. Wireless Commun. Mobile Comput.*, vol. 5, no. 8, pp. 945–956, 2005.
[29] J. Mitchell and W. Pennebaker, *MPEG Video: Compression Standard*. London, U.K.: Chapman and Hall, 1996, ISBN 0412087715.
[30] NS-2 Simulator. [Online]. Available: http://hpds.ee.ncku.edu.tw/smallko/ns2/ns2.htm.
[31] YUV Video Sequences. [Online]. Available: http://trace.eas.asu.edu/yuv/index.html.
[32] *Wireless LAND Medium Access Control (MAC) and Physical Layer (PHY) Specifications Amendment 8: Medium Access Control Quality of Service Enhancements*, IEEE Standard 802.11e-2005, 2005.
[33] C. H. Lin, C. K. Shieh, C. H. Ke, N. Chilamkurti, and S. Zeadally, "An adaptive cross-layer mapping algorithm for MPEG-4 video transmission over IEEE 802.11e WLAN," *TeleCommun. Syst. J.*, vol. 42, no. 3, pp. 223–234, Dec. 2009.
[34] *IEEE 802.16 Standard-Local and Metropolitan Area Networks-Part 16: Air Interface for Fixed Broadband Wireless Access Systems*, IEEE Standard 802.16-2004.
[35] D. J. Deng, L. W. Chang, C. H. Ke, Y. M. Huang, and J. M. Chang, "Delay constrained uplink scheduling policy for rtPS/nrtPS service in IEEE 802.11e BWA systems," *Int. J. Commun. Syst.*, vol. 22, no. 2, pp. 119–133, Feb. 2009.

Cheng-Han Lin is currently a Post Doctor in National Cheng Kung University, Taiwan. He was a Post Doctor from 2010 to 2012 in National Kaohsiung University of Applied Sciences, Taiwan. He received the Ph.D. degree from the Department of Electrical Engineering, National Cheng Kung University in 2010 and M.S. and B.S. degrees from the Electrical Engineering Department of National Chung Cheng University in 2002 and 2004, respectively. His current research interests include wireless MAC protocols, multimedia communications, and QoS network.

Ce-Kuen Shieh (M'02) is currently a professor teaching in the Department of Electrical Engineering, National Cheng Kung University. He received his Ph.D., M.S., and B.S. degrees from the Electrical Engineering Department of National Cheng Kung University, Tainan, Taiwan. His current research areas include distributed and parallel processing systems, computer networking, and operating systems.

Wen-Shyang Hwang (M'97) received B.S., M.S., and Ph.D. degrees in electrical engineering from National Cheng Kung University, Taiwan, in 1984, 1990, and 1996, respectively. He is currently a professor of Electrical Engineering, and the chairman of the department of computer science and information engineering in National Kaohsiung University of Applied Sciences, Taiwan. His current research interests are in the fields of multimedia wireless communication, wireless mesh networks, storage area networks, WDM Metro-ring networks, performance evaluation, software design for embedded systems, Internet QoS, and Internet applications.