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Using Interleaving Techniques with FEC Mechanisms to Deal with Burst Errors

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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. An Enhanced Random Early Detection Forward Error Correction (ERED-FEC) mechanism to improve the quality of video transmissions over Wireless Local Area Networks (WLANs) is utilized. 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. The channel seen by applications is the physical channel as modified by the error correcting mechanisms used at the physical level. The residual error process is to be considered and the block error process beyond the marginal statistics has to be considered. A Markovian model for the block errors can be used and can be shown adequate.

KEYWORDS: Video Transmission, Forward Error Correction (FEC), QoS, Access Point (AP) Solutions, Redundant Packets

I. INTRODUCTION

Currently, the Wireless Local Area Networks (WLANs) are emerging as the de facto specification for various service providers in order to provide a pervasive and unconstrained mobile internet services. They are indeed expected to provide a desired level of Quality of Service (QoS) for delivering video services. Moreover, the IEEE802.11 WLANs are deployed widely in places like office buildings, railway stations, airports, institutions and the home environments. However, these wireless channels are susceptible to several transmission errors like scattering, interference, fading, etc. [2-3]. In order to recover from the packet losses in these wireless environments, strategies like Automatic Repeat reQuest (ARQ) or Forward Error Correction (FEC) are being used. In the case of ARQ, the lost packets are automatically retransmitted when there is a timeout or an explicit request from the receiver. But this scheme results in higher retransmission latency. When the FEC schemes are considered, the packet losses are avoided to a greater extent by transmitting some amount of redundant packets together with the source packets in such a way that the whole block of packets can be successfully reconstructed even when there is some packet loss within the block at the receiver side. So this scheme results in lower retransmission latency. Hence of these two strategies, FEC is widely preferred for wireless video transmissions. When the FEC mechanisms are considered, there are several ways of implementation like Sender based and AP based. In the Sender based FEC schemes, the FEC redundancy rate is calculated by the sender system according to the feedback information provided by the receiver for the recent packets received pertaining to the desired level of QoS. The traditional Sender based FEC schemes are of two types: Static-Sender based and Dynamic-Sender based. They are characterized based on the calculation of number of redundant packets to be added with the source packets to be transmitted.

In the Static-Sender schemes the redundancy rate is fixed irrespective of the channel conditions and the load on the network. Hence there is an unpredictable recovery performance for this type of schemes. They fail to adjust the redundancy rate according to the condition of the channel. There are several implementations in the Dynamic-Sender techniques. They include the parameters like Packet loss rate, Peak Signal-to-Noise Ratio (PSNR), user feedback parameters, MAC layer feedback parameter and so on. Accordingly, they are classified as single layer and cross layer solutions.

In most cases of Sender solutions the redundancy rate is calculated at the application layer according to the Acknowledgement Messages (ACK). The feedback process and the redundancy rate calculation process takes a finite time. Hence there is no guarantee that this solution accurately reflects the channel condition and the load on the network. This is the reason for moving towards the Access Point based solutions where the AP takes care of rate calculation without the need for feedback information from the receiver system. In the series of implementation strategies in the AP based schemes; Random Early Detection (RED) algorithm is being embedded to improve the performance significantly over the existing scenario. This work can be preceded in two directions, as one according to the layered solutions and the other based on implementation at the sender or Access point. The video transmission is mainly concerned with the three communication layers of the network: APP layer, MAC layer and PHY layer. The features and working at each of these layers are:

A. APP Layer

Storage of pre-coded video or online encoding is done at this layer. This layer also provides convenient video streaming related information like priority structure for video frames, coding parameters of the video content and similar others.

B. MAC Layer

Error recovery parameters and channel access mechanisms are related to this layer and this has direct impact on the performance of the video streams being transmitted.

C. PHY Layer

The time varying channel information and the error information are dealt in this layer. This information can be better used to improve the overall efficiency of the transmission.

II. VIDEO APPLICATION SCENARIOS

There are two different scenarios in the video application: real time video transmission and video streaming. In the former scenario, several users of the same WLAN transmit their video data to each other or use the AP to do so for the remote users. This may include video conferencing and other interactive video services which need onboard video coding and perfect delay constraints. In the video streaming, the video server is responsible for the delivery of the video content to several users connected to the WLAN. This include video download, video-on-demand where at the receiver side the video frames are buffered before the play out. As a result, offline encoding is adopted and hence can tolerate relaxed delay constraints.

A wide variety of applications are growing up, to enable multimedia services delivery over WLANs like VoIP, video streaming, video conferencing and so on. Still, video delivery over wireless LANs is a challenging task, where a desired level of QoS needs to be met to its users. The first and foremost reason is, the channel is error prone. The most videos are compressed and this compressed video is error sensitive and time critical in nature. The algorithms used for video compression aims to reduce the bandwidth which results in complex dependencies among the transmitted video blocks. In such cases when a single packet is lost on transmission, it will affect both the current frame and also the upcoming frames. Also the transmission should be within the threshold level, which is decided by the decoding time at the receiver side. When the delay experienced during the transmission exceeds the threshold, the entire video becomes useless, though they are transmitted reliably. Hence reducing error and delay parameter are the conflicting demands in the video transmission over WLANs.

III. FORWARD ERROR CORRECTION

The basic idea behind the FEC mechanism is adding redundant packets to the source packets of the video stream in such a way that the original block of video packets can be reconstructed at the receiver end even if there is a loss in transmitted block. This avoids the need for retransmission of the video data to a major extent and hence the

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retransmission latency is less. Since there are possibilities to recover the packets which would otherwise be lost, the effective loss rate is lower than the actual loss rate. The coding theory techniques are used to derive the redundant packets and the widely used code is Reed-Solomon code [11]. This code proves to be ideal in protection against error caused by packet losses. Also, this scheme is able to recover the symbols of lost source data from relatively lesser number of received symbols when compared to other existing coding schemes.

IV. VARIANTS OF FORWARD ERROR CORRECTION SCHEME

A. Sender-Based FEC Mechanisms:

(i). Constant Error Rate FEC

This mechanism is based on the feedback information provided by the receiver end system using the packet error rate, which is periodically estimated at the receiver side. This scheme allows having a dynamic control over the QoS of real-time multimedia traffic over a heterogeneous environment. As the receiver sends the feedback, the sender system tries to restore the error rate to its original value by calculating the redundancy rate [5]. Hence maintaining a constant packet error rate is the main theme of this class of FEC scheme.

(ii). Cross-Layer FEC

The ultimate aim of this class of FEC mechanism is to maintain the quality of the video data above a pre-specified level for all the users of the multicasting wireless network. It is the task of every user to report the number of packets received periodically out of the source packets that were originally sent. Then the sender calculates the number of packets that were lost during the transmission and accordingly estimates the decodable level of all users on an average. This estimation is used to adjust the FEC redundancy rate [6].

(iii). Adaptive FEC

An adaptive FEC scheme is basically designed to facilitate end-to end transmission of real-time traffic where the timing constraints dominate the retransmission based congestion control. The idea behind the protocol is to adjust the redundancy rate according to the current delay in the network. When the network delay is more, the redundancy rate is decreased and if the delay on the network is less, the rate is increased. This is how the QoS is achieved using adaptive FEC schemes.

B. AP Based FEC Mechanisms:

In this class of mechanisms, the redundancy rate is calculated at the AP. This idea avoids the need for feedback information to be received from the receiver system as in the case of Sender based FEC schemes. Hence the retransmission delay is reduced and the video quality can be improved.

(i). Random Early Detection FEC

Previous strategies result in increased redundancy rate in order to cope up with increasing packet losses. This in turn degrades the network performance by increasing the network congestion. The RED FEC [9] is the scheme that was proposed to generate the FEC packets at the Access Point that considers the current traffic load into account. This traffic load is measured by the queue length at the AP. The idea is to increase the redundant packets as the queue length decreases and reduce in the other case. The most important point is that, no redundant packets introduced when the queue is almost full. This is to ensure that the network is not overloaded due to redundancy. This improves the quality parameters by avoiding excessive redundant packets in the network. Anyhow the packet loss rate is not considered in calculation of the redundancy rate.

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(ii). Adaptive Cross-Layer FEC

AC-FEC [10] uses the functionalities of different network layers. The packet loss information is taken from the MAC layer ARQ function. The redundancy rate is controlled using the UDP protocol at the application layer. The failure information of the MAC layer is used to assess the performance of the transmission continuously. When a complete block of video data is sent, the failure counter information is used to manipulate the FEC rate accordingly. This does not consider the network load into account which is a shortcoming in this scheme. This results in self-induced congestion in the network.

(iii). Enhanced Adaptive FEC

EAFEC [8] proposes the idea that any node wishing to send data to any other node in the network should send the data to the AP first. This is because AP is the place suitable for adding FEC mechanism. The AP considers the current network condition to estimate the FEC redundancy rate. The network traffic load is estimated using the AP queue length. Also the retransmission time taken by a packet indicates the channel status of the wireless network. But the limitation in this system is that the packet loss rate is ignored in calculation of the redundancy rate. This improved the additional packet loss problem encountered in the static FEC schemes. The optimum threshold values can be found out to improve the performance to certain extent.

(iii). Enhanced Random Early Detection FEC

This ERED-FEC [1] scheme is considered to be superior among the existing FEC schemes as of now. In this scheme the redundancy packets are calculated and introduced at the Access Point. This estimates both the Channel Condition and the network load in order to arrive at a suitable redundancy rate. As mentioned earlier, the network load is evaluated using the AP queue length as the parameter. The number of FEC packets is increased if the queue is nearly empty which indicates that the network is lightly loaded. If the queue length is almost full, then it indicates that there is a heavy load in the network and hence the redundancy rate has to be reduced accordingly. This eliminates the unnecessary overloading of the network thereby increasing the overall quality of the video data to be transmitted.

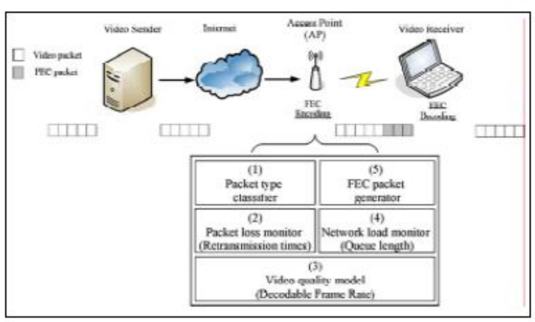

Figure 1 Architecture of ERED-FEC [1]

The steps in video streaming using the ERED-FEC scheme are as follows:
• The controller retrieves the information from the header of the video packet to identify the packet type by checking the RTP header value.
• The packet loss manager estimates the packet loss rate after a complete block of video data is received by calculating the number of packet retransmission for each block of data.
• Using a predefined video quality model, the FEC redundancy rate is calculated

.

When a block of packets arrives:
/* estimation of packet loss rate in wireless network */
$P_{estimation,i} = RT_{weight} \times P_i + (1 - RT_{weight}) \times P_{estimation,i-1}$
/* 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 */
$Q_{length,i} = Q_{weight} \times Q_i + (1 - Q_{weight}) \times Q_{length,i-1}$
if ($Q_{length} < Th_{low}$)
 Final_FEC_pkt = Max_FEC_pkt;
else if ($Q_{length} < Th_{high}$)
 $\text{Final\_FEC\_pkt} = \text{Max\_FEC\_pkt} \times \frac{Th_{high} - Q_{length}}{Th_{high} - Th_{low}}$;
else
 Final_FEC_pkt = 0;

Figure 2 ERED-FEC algorithm [1]

• The load monitor estimates the current traffic load by using the AP queue length as the parameter.
• Finally, the FEC packet generator adjusts the final FEC rate.
This scheme improves the video transmission quality using the wireless LANs over the previous mechanisms.

V. INTERLEAVING

Let d be the interleaving depth, and N be the block length. Then, prior to transmission, the blocks are written as rows of a dxN matrix, which is read by column to obtain the data stream which is actually sent on the channel. At the receiving end, the dual operation is performed and the original order of the data is restored. This operation has the effect of mitigating the effect of the channel memory and of breaking error bursts so that error correcting codes can be more effectively used, while requiring some memory and causing some delay.

As opposed to FEC, interleaving [8] does not add redundant data to the transmitted data but rearranges the sequence of the frames in order to reduce the impact of packet loss. Instead of recovering packet errors, this technique can only decrease or distribute the influence of packet loss. In addition, using this method may add some time to round-trip delay because packets must be placed in the senders' buffers to be transmitted and stored in receivers' buffers, to be reassembled and decoded. The main principle of the interleaving technique involves rearranging the information or packets to be transmitted and to reassemble the information at the receiver's buffer; therefore, two buffers are needed to complete the task: a transmitting buffer and a receiving buffer.

VI. SIMULATION TOPOLOGY AND PARAMETER SPECIFICATION

We set up a video server and simulated two background traffic flows. The video stream sent by the video server was delivered to the Internet next to the wireless AP in the receiver's area, and then to the receiver by the AP. The simulated background traffic flows include File Transfer Protocol (FTP) traffic, which was delivered in Transmission Control Protocol (TCP) packet format, and exponential traffic, which was delivered in User Datagram Protocol (UDP) packet format. Because the video streaming reached the receiver through the wireless AP, our proposed enhanced structure of layered FEC and interleaving implemented the wireless AP to enhance the video delivery over wireless networks.

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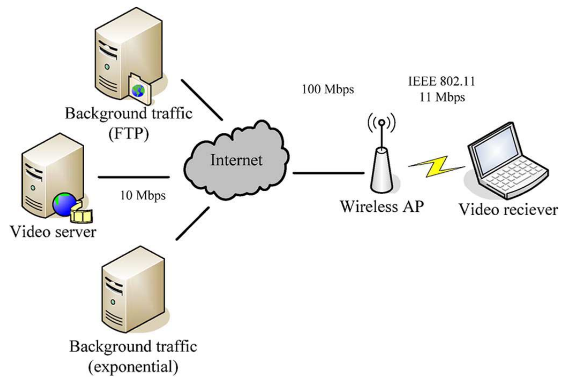

Figure 3 simulation topology

The adopted system in the network environment was an IEEE 802.11b 11 Mb/s wireless network and its error model was the Gilbert- Elliott model because the GE model can be used to simulate the error status of wireless networks, especially with respect to consecutive packet loss. The bandwidth for the two background traffic flows is 10 Mb/s. Therefore, our proposed structure was implemented to the AP to improve the video transmission over wireless networks.

The multimedia server stores the multimedia streaming for users to access and we used the extension of H.264/AVC-Scalable Video Coding as the coding format; this allows the server to offer video of different picture sizes, frame rates and video quality to users. Spatial (picture size) scalability, temporal (frame rate) scalability and SNR scalability can encode different adaptive video streams.

In this simulation, the importance levels of SVC video streaming were simplified into three groups according to the layered coding structure of SVC: the base layer as a group, Enhancement layer 1 as a group and Enhancement layer 2 as a group. The threshold values for the base layer were set to (10, 40, 5, 15) and for Enhancement layer 1 (10, 36, 6, 19), while Enhancement layer 2 was not protected. The former two values refer to the low and high threshold values in determining network congestion levels and the latter two values refer to the low and high threshold values in determining the channel status. Since the base layer is the most important, the highest FEC strength was assigned to the base layer and the lowest to Enhancement layer 1. Enhancement layer 2 was not protected by FEC because it is the least important, and served as the control group to verify the condition without FEC.

VII. CONCLUSION

This paper has presented the various technologies available for video transmission over a wireless LAN in an efficiently and effectively. One stream of classification is based on the implementation layer of the wireless network. The latter class of solutions is based on the error correction scheme and its variations. The advantages and disadvantages of each such type were discussed. We have focused mainly on the QoS related aspects for discussion which includes energy efficiency, latency, delay parameters and so on. Other important properties to be focused include the security and privacy aspects. As a further step, the optimization criteria and the cross-layer architecture can be further studied. The solutions are expected to improve and increase for the next generation video data over wireless LANs.

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