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Effective quality analysis for video streaming over wireless ad hoc network

Zhihai He
Chang Wen Chen

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Zhihai He[1] and Chang Wen Chen[2]
[1]Department of Electrical and Computer Engineering
University of Missouri, Columbia, MO 65211
Email: HeZhi@missouri.edu
[2]Department of Electrical and Computer Engineering
Florida Institute of Technology, Melbourne, FL 32901
Cchen@fit.edu

Abstract

Video encoding and streaming over wireless ad hoc network operates under severe conditions, such as time-varying channel characteristics with bursty errors, limit power for data transmission, and dynamic topology of the self-organized network, stringent time delay for packet delivery, etc. Due to the dynamic topology, complex mechanism, and time-varying nature of the wireless ad hoc network, the network system and the streaming service often exhibit unpredictable behaviors. The ultimate goal in the wireless video streaming service is to provide the end user with the best possible video presentation quality. The video streaming quality, often measured by the end-to-end picture distortion, is affected by the scene coding characteristics $\mathcal{S}$ of the input video, and the configuration of the network parameters $\mathcal{N}$ which includes bandwidth, transmission power, bit error ratio, delay, etc. This brings up the following important and challenging issue: given an input video with scene characteristics $\mathcal{S}$ and a network configuration $\mathcal{N}$, what is the average video streaming quality the receiver could expect the system to provide. To address this issue, in this work, we examine the behaviors and constraints of the major components in the streaming system, and propose an across-layer framework to model, control, and optimize the end-to-end video streaming quality.

I Introduction

A wireless ad hoc network is a system of mobile devices communicating with each other over a dynamic and self-organizing wireless network. Wireless ad hoc networks have the potential to provide instantaneous person-to-person, person-to-machine or machine-to-person communications, without the need for any centralized network administration or fixed infrastructure, such as base stations or access points. Information processing and communication over wireless ad hoc network has found many potential applications in battlefield communication, video surveillance, security monitoring, and environmental tracking. With the recent development of efficient compression systems, high-speed wireless local area networks (WLAN) technologies, and the low-cost off-the-shelf capture and communication devices, video communication over the wireless ad hoc network is becoming a reality.

Video encoding and streaming over wireless ad hoc network operates under severe conditions and unique constraints, which impose a set of new challenges in research and development. First, in wireless communication, the signals may travel through multiple paths between the transmitter and the receiver. The receiver may observe variations of signal amplitude, phase, and angle of arrival signals. The multi-path fading makes the wireless channel a time-varying channel with relatively large

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fluctuation in channel characteristics, such as bandwidth, bit error ratio (BER), etc [6, 7]. Second, the mobile device, often powered by batteries, has limited power supply for data transmission. Third, in the self-organized dynamic wireless ad hoc network, the data packets could be lost due to medium access contention and link disconnection. Finally, video streaming, as a real-time communication service, has a stringent delay requirement. The delay issue becomes even more critical in the ad hoc network where the wireless link and network topology are arbitrary and dynamic.

The ultimate goal in the wireless video streaming service is to provide the end user with the best possible video presentation quality [3]. The video streaming quality, often measured by the end-to-end picture distortion, is affected by the scene coding characteristics $\mathcal{S}$ of the input video, and the configuration of the network parameters $\mathcal{N}$ which includes bandwidth, transmission power, bit error ratio, delay, etc. This brings up the following important and challenging issue: given an input video with scene characteristics $\mathcal{S}$ and a network configuration $\mathcal{N}$ what is the average video streaming quality one could expect the system to provide? To address this issue, in this work, we examine the behaviors and constraints of the major components in the streaming system, and propose an across-layer framework to model, control, and optimize the end-to-end video streaming quality.

The rest of the paper is organized as follows. In Section II, we examine the basic process of video encoding and streaming over the wireless ad hoc network, and define the end-to-end video quality. In Section III, we introduce the traffic profile function to describe the scene characteristics of the input video for traffic management in the wireless network. Section IV presents the analysis on effective transmission power. Section V introduces the end-to-end distortion model which integrates the traffic profile function and the effective transmission power to predict the expect video streaming quality at the end user. Some concluding remarks are given in Section VI.

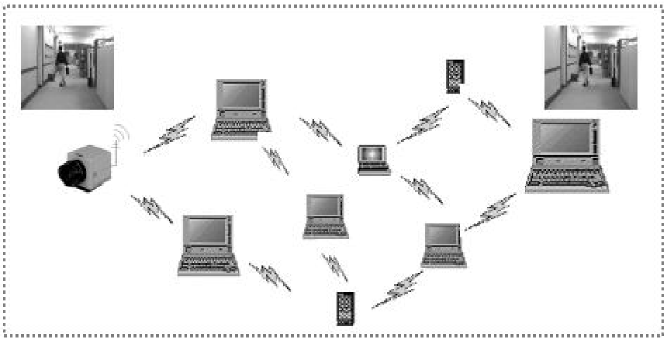

Figure 1: Video encoding and streaming over wireless ad hoc network.

II Video Streaming Over Wireless ad hoc Network

II-A System Design

In video encoding and streaming over wireless ad hoc network as illustrated in Fig. 1, the compressed video data, either precoded and stored on the streaming server, or live captured and compressed by the video encoder, is transmitted from the source communication node to its neighboring nodes. With cooperative efforts from other nodes for packet relaying and efficient network-level packet routing, the video packets arrive the receiver node after multi-hop wireless transmission. The basic framework we propose to implement such type of video streaming over wireless ad hoc network is illustrated in Fig. 2. The ultimate goal of the system design is to make the video presentation quality at the end user as best as possible. A good video presentation quality implies a high spatial quality of each compressed video frame, as well as a consistent and smooth temporal quality across frames. The video is first captured from a camera on the portable device, and then compressed by a video encoder. Without compression, the amount of data in the raw video is huge, which requires tremendous energy for data transmission. To achieve consistent temporal quality, variable bit rate (VBR), or buffer constrained VBR encoding is assumed [14]. The video quality, measured by the source coding distortion D_s which is the mean square error (MSE) between the encoded picture and the original one, is a function of the encoding bit rate R. Let the video frames are indexed by n. The bit rate profile, $\{R(n)\}$, determines the required channel bandwidth for video streaming, and provides the necessary information for traffic management, admission control, resource allocation, and quality of service (QoS) provisioning [15]. In the networking literature, the estimation of the statistics of $\{R(n)\}$ is also called *traffic modeling*. In Section III, we will develop a scheme to predict the traffic profile function $\{R(n)\}$ for a given video streaming quality level D_s.

One major challenge in wireless communication is that the data transmission channel is severely error corrupted, especially in the wireless ad hoc network where the communication nodes are mobile, the transmission power on the mobile device is limited, and the network topology is dynamic. Therefore two types of transmission errors in wireless ad hoc network: bit errors and packet loss caused by link contention. The link contention problem will be considered in detail in Sections IV-A. To protect the video data from the channel bit errors, forward error correction (FEC) is applied to the compressed video data to increase the error resilience and to allow the receiver nodes to correct some of the error bits using FEC decoding. The corrupted data bits and the lost packets will cause decoding errors in the received video. Most importantly, the decoding errors in the current video frame will propagate to its following frames along the motion prediction path, which significantly degrades the video presentation quality [2]. As illustrated in Fig. 2, this type of picture distortion, measured by the MSE between the encoded picture and the received one, is called channel transmission distortion, denoted by D_c. Note that D_c depends on the scene characteristics $\mathcal{S}$ of the encoded video and the probability of transmission errors p_e. Note that the source coding distortion and the transmission errors are two independent error sources. Therefore, we have

$$D = D_s + D_c = D_s(R) + D_c(\mathcal{S}, p_e). \tag{1}$$

The end-to-end video quality, indicated by the end-to-end distortion D, is the ultimate performance measure for the wireless video streaming service. It should be noted that the proposed framework is a simplified version of the practical system for video streaming over wireless ad hoc network. In practice, there are many other issues in each layer of the networking architecture, such as physical layer channel modeling, specific network routing and flow control, etc. However, the objective of the proposed framework is to capture the major components in the network streaming system which have significant impact to the end-to-end video presentation quality. In additional, the framework-level simplification is also necessary to enable us to analyze, model, and optimize the behavior of a practical end-to-end system.

II-B Effective Quality Analysis

The objective of this work is to analyze the basic components in the practical wireless ad hoc network system which have significant impact on the end-to-end video streaming quality, and develop a basic framework to predict the quality of the video streaming service under various constraints, such as bandwidth, transmission power, channel BER, transmission delay, etc. Specifically, we will develop a traffic model for the input video using a compression-based approach. We will also discuss the mutual relationship among the transmission power, channel BER, FEC and media access control

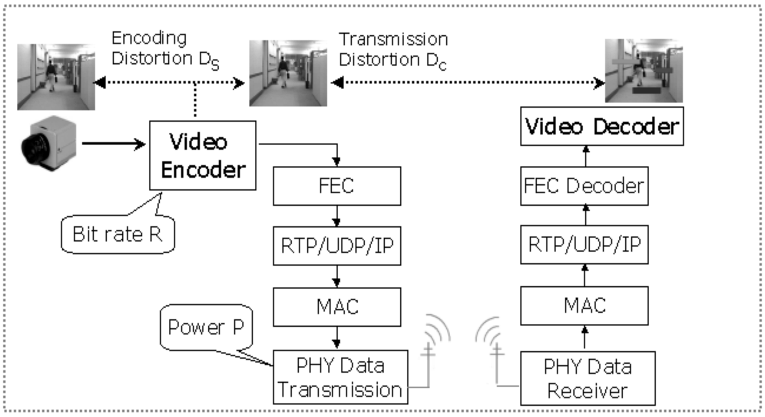

Figure 2: Illustration of the proposed system for video streaming over wireless ad hoc network.

(MAC) in the wireless ad hoc network. Using a transmission model developed in our previous work, all of these components will be integrated together to form an end-to-end video quality analysis framework.

III Traffic Profile Function

III-A Traffic Modeling

For successful streaming of the video data over the wireless network, the traffic of the input video needs to be analyzed and modeled for efficient resource allocation and network management. It also plays a very important role in quality of service (QoS) provisioning [15]. A good traffic model provides a tool to accurately characterize the video sources and enables a theoretical analysis of the network performance as well. Due to the scene activity in the video sequence and inherent encoding mechanism of the compression system, the video traffic often exhibits a burty nature, which has imposes tremendous pressure on bandwidth resources and networks congestion control techniques as well as traffic management. Without a traffic model or characterization, the efficiency and accuracy of the measurement-based admission and congestion control will be significantly reduced. Many traffic models reported in the literature are based on statistical analysis, using some statistical models, such as Markov models [9], self-similar or fractal models [13], and auto-regressive processes [12], to analyze the output bit rate of the encoder and model the time-varying behavior of the compressed video data. In this work, we adopt a different approach for traffic modeling. Specifically, this approach is based on *a prori* prediction of the video traffic before compression instead of statistical analysis of encoder bit rate after the video compression.

It is well known that the coding statistics of a video frame is described by its rate-distortion (R-D) function $R(D)$, where R and D represents the coding bit rate and picture distortion. Let the video frames indexed by n. The function $R(D, n)$ describes the coding characteristics of the video sequence. In video encoding and streaming, variable bit rate (VBR) coding is often used to achieve consistent encoded video quality. For a given quality level D_0, $\{R(D_0, n)\}$ is the video traffic data which the traffic models are trying to analyze and model. Therefore, $R(D, n)$ describes the video traffic for any given quality level D, and we refer to it as the *traffic profile function*. If we can estimate this traffic profile function, then we can easily compute all the traffic and admission control parameters, such as average data rate, peak data rate, effective bandwidth, buffer queuing length, etc.

III-B Estimating the Traffic Profile Function

Note that the standard video compression systems are based on temporal motion prediction, in which the reconstructed frame is used for motion estimation and compensation of its next frame. In other words, the R-D distortion function of frame n depends on the quality level of the reconstructed frame $n-1$. Before the encoding process reaches to the frame, its R-D distortion function is hard to estimate, because even the motion compensated difference picture is not available. The recursive nature of this inter-frame R-D dependency make it very challenging to estimate the traffic profile function $R(D, n)$ before compression [10]. In this work, by making a reasonable assumption which simplify the estimation process, we propose a simple and efficient scheme to estimate the traffic profile function $R(D, n)$. In addition, from our simulation results, it turns out this assumption does not affect the performance of the traffic profile function in traffic and admission control. According to the classical R-D formula, the R-D functions is given by

$$R(D) = \frac{1}{2}\gamma \log_2 \frac{\sigma^2}{D}, \tag{2}$$

where σ^2 is the variance of the motion compensated difference picture if intercoded or the variance of the original picture if intracoded. γ is a model constant. To compute the R-D function for frame n, we need to estimate variance of the frame, denoted by $\sigma^2(n)$. As mentioned in the above, the value of $\sigma^2(n)$ depends on the history of the encoding process, especially the quality level of the previous reconstructed frame. The proposed estimation scheme has two major steps. In the first step, we perform look-ahead processing of the input video. Specifically, the original frame $n-1$ is used for motion estimation and compensation of frame n to obtain the difference picture. The variance of this difference picture is called the original frame difference, denoted by $F_d(n)$. Let $F(n)$ be the original frame n. We have

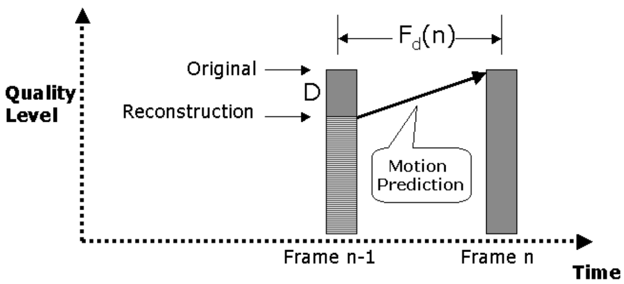

Figure 3: Estimation of the frame variance during the look-ahead processing.

$$F_d(n) = \mathrm{Var}[\, F(n) \overset{\mathrm{mc}}{—} F(n-1)\,], \tag{3}$$

where $\overset{\mathrm{mc}}{—}$ represents the motion compensation based difference operation. Let $\hat{F}(n-1)$ be the encoder reconstruction of frame $n-1$. We have

$$\sigma^2(n) = \mathrm{Var}[\, F(n) \overset{\mathrm{mc}}{—} \hat{F}(n-1)\,]. \tag{4}$$

Let D be the average distortion level, corresponding to the average video quality, of the VBR encoding and streaming session. By the definition of picture distortion,

$$D = \mathrm{Var}[F(n) - F(n-1)]. \tag{5}$$

Note that the picture distortion is originated from the quantization errors. Since the quantization errors and the original frame difference are two independent statistics, from Eqs. (3)-(5), we can see that

$$\sigma^2(n) = F_d(n) + D, \tag{6}$$

as illustrated in Fig. 3. Therefore,

$$R(D,n) = \frac{1}{2}\gamma \log_2 \frac{F_d(n) + D}{D} = \frac{1}{2}\gamma \log_2[1 + \frac{F_d(n)}{D}]. \tag{7}$$

This is the simple and efficient model we are going to use to predict the traffic profile function. It should be noted that here we assume that the motion vectors in the actual video encoding are the same as those obtained from the look-ahead process. This assumption is reasonable, especially when the quality level of the streaming session is relatively high. Since the purpose of traffic modeling is to extract sufficient high-level statistics from the video data for traffic and admission control, the minor deviation of the motion vectors in the actual encoding will not affect the overall performance much. Fig. 4 shows the actual frame bits statistics and the estimation for the "Foreman" CIF sequence coded at 30 fps with an average quality of 33.5 dB. We can see that the estimated rate profile function is statistically close to the actual data.

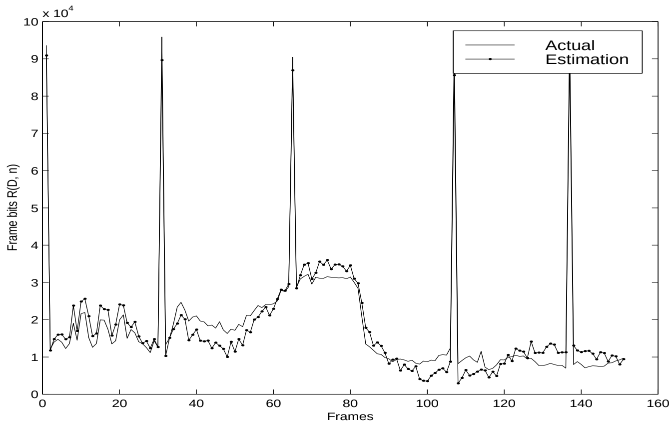

Figure 4: Estimation of the traffic profile function $\{R(D,n)\}$ for "Foreman" CIF coded at 30 fps with an average quality of 33.5 dB.

III-C Effective Bandwidth Analysis

In video streaming over the networks, the video traffic exhibits a bursty nature. The bursts from different video sources which have already been admitted into the network traffic may coincide and temporarily require a data rate higher than the network is able to provide. One popular and successful approach to handle this problem is to use the concept of *effective bandwidth*. Let $S(t)$, $t \geq 0\}$, be the amount of source data (in bits) over the time interval $[0,t)$ arriving at the network. Clearly, we can see that

$$S(t) = \sum_{n=0}^{\lfloor t/\Delta \rfloor} R(D,n), \tag{8}$$

where Δ is the time interval between two consecutive encoded frames. For $t_2 \geq t_1 \geq 0$, let $A(t_1,t_2) = A(t_2) - A(t_1)$, which represents the accumulative arrived data within the time period of $[t_1,t_2)$. For

$\alpha \geq 0$ and $t \geq 0$, we define

$$\mathcal{B}(\alpha, t) = \frac{1}{t} \sup_{\theta \geq 0} \log E[e^{\alpha S(\theta, \theta+t)}], \tag{9}$$

and

$$\mathcal{B}(\alpha) = \lim_{t \to \infty} \sup \mathcal{B}(\alpha, t). \tag{10}$$

If $\mathcal{B}(\alpha) < \infty$ for all α, the corresponding traffic process $S(t)$ is called exponential process. In this case, the effective bandwidth is defined as

$$W(\alpha) = \frac{\mathcal{B}(\alpha)}{\alpha}, \tag{11}$$

for all $\alpha > 0$. The physical mean of this effective bandwidth function is that: $W(0)$ represents the long term average rate of the data arrival process $S(t)$; $W(\infty)$ represents the long term peak rate of $S(t)$; and $W(\alpha)$ is an increasing function from the long term average rate to the long term peak rate. Once the effective bandwidth function is obtained, it can be used for admission control and network QoS provisioning. Using the traffic profile function, from Eqs. (8)-(11), we can see that the effective bandwidth function can be computed from the video scene characteristics $\{F_d(n)\}$. Fig. 5 shows the effective bandwidth derived from the actual frame bits statistics and the one derived from the estimated traffic profile depicted in Fig. 4. We can see that the effective bandwidth estimation is quite accurate, especially when α is relatively small. When α is large, the estimation performance depends on how accurate the peak rate estimation is, especially for those Intra or high motion frames.

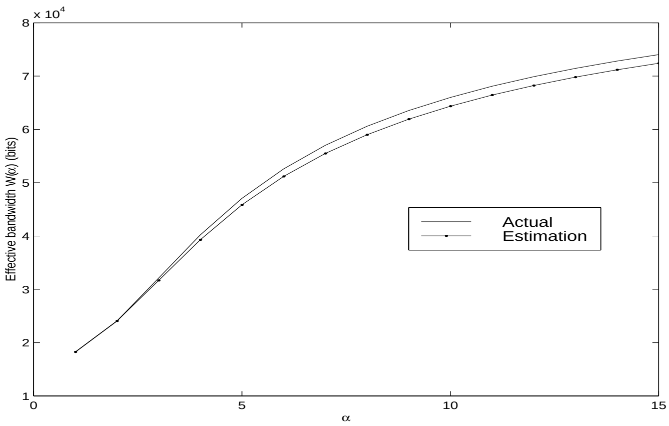

Figure 5: Effective bandwidth estimation using the traffic profile function for "Foreman" CIF coded at 30 fps with an average quality of 33.5 dB.

IV Effective Transmission Power and Hybrid FEC-ARQ

As mentioned in Section I, there are two types of errors in data transmission over the wireless ad hoc network: bit errors and packet loss caused by link contention. The link contention problem is related to the media access control (MAC) layer protocol of the ad hoc network. The relationship between the BER and the transmission power can be analyzed using a wireless channel model.

IV-A 802.11 MAC Layer

Compared to the point-point single-hop wireless communication as the cellular network, the wireless ad hoc network has unique features such as multi-hop data transmission, self-organization, dynamic network topology, and arbitrary link connection, which impose significant challenges in network routing, flow control, and traffic management. The 802.11 protocol has been developed to address these issues. In the standard, if node A is ready to send a data packet to node B, it first transmit a short RTS (Request To Send) which includes the source, destination, and duration of the following data transmission and acknowledgement from B. If B receives the RTS, it responds with a short CTS (Clear To Send) packet to A. A then sends out the data packet and B responds with an ACK packet if the packet is received correctly. If the packet is lost due to collision in transmission or received in errors due to the channel bit errors, no ACK is sent and the packet is retransmitted. The retransmission is performed either until the data packet is received correctly or the maximum number of retransmission is reached. The maximum number of retransmission is often determined by the end-to-end transmission delay. Obviously, packet retransmission requires additional transmission energy, and hence reduces the energy efficiency.

IV-B Effective Transmission Power

To protect the information data against the channel bit errors and reduce the packet retransmission caused by packet corruption, forward error correction (FEC) has been added into the 802.11 specification. In this hybrid FEC-ARQ scenario, increasing the transmission power will lower down the channel BER, which results in less data corruption and fewer packet retransmission. On the other hand, reducing the transmission power will increase the channel BER, which requires more packet retransmission. Therefore there is a tradeoff between the transmission power and the MAC layer power consumption for packet retransmission. It has been observed that the energy consumption of the wireless data transmission is proportional to the number of transmitted bits. Let's define the effective transmission power as as

$$P_{eff} = \frac{P \cdot B_{all}}{B_{data}}, \tag{12}$$

where B_{data} and B_{all} represent the size (in bits) of the video data and the number of actual transmitted bits including the retransmitted packets. We can see than P_{eff} is larger than the actual transmission power due to the overhead power consumption for packet retransmissions. To study the behavior of the hybrid FEC-ARQ at the MAC layer and the effective transmission power, we use the Gilbert-Elliot channel model [8]. For a given a specific modulation scheme and transmission power, the BER of each channel state can be computed. Using the discrete event network simulator, we are able to analyze the effective transmission power and estimate the packet loss ratio for a given configuration of network. Fig. 6 shows a typical plot of the effective transmission power as a function of the actual transmission power. In this example, there are 4 nodes in the ad hoc network about 50m apart from each other. The transmission frequency is 2.4 GHz, and the packet size is 512 bytes.

V End-to-End Distortion Prediction

In our previous work [2], we have developed a model to predict the picture distortion caused by transmission errors. At the decoder, the transmission errors cause decoding failure of the packet, which translates the transmission errors into packet loss. Let p be the packet loss ratio. Let Let β be the fraction of intra macroblocks (MB's) in the video frame. The overall channel distortion can

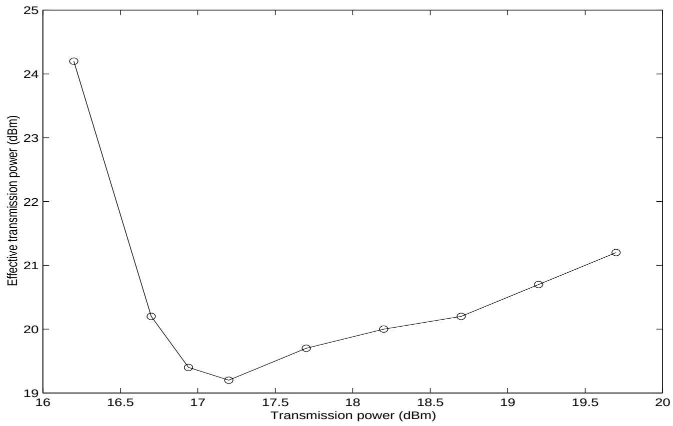

Figure 6: A typical plot of the effective transmission power as a function of the transmission power.

be approximated by

$$D_c(n) = \Gamma_1 \cdot D_c(n-1) + \Gamma_2 \cdot F_d(n), \tag{13}$$

where $\Gamma_1 = (1-p)(1-\beta)b + p$, and $\Gamma_2 = pa$. a and b are model parameters whose detailed definition and estimation scheme are given in [2]. Solving the recursive equation (13) yields,

$$D_c(n) = \Gamma_1^n \cdot D_c(0) + b \sum_{i=1}^{n} \Gamma_1^i \cdot F_d(i). \tag{14}$$

Note that the original frame different $F_d(i)$ has also been used in the estimation of the traffic profile function. Let T be the total number of coded video frames. The average channel distortion, denoted by $\bar{\mathcal{D}}_c(T)$, is given by

$$\begin{aligned} \bar{\mathcal{D}}_c(T) &= \frac{1}{T} \sum_{n=1}^{T} D_c(n) = \frac{1}{T} \sum_{n=1}^{T} \Gamma_1^n \cdot D_c(0) + b \frac{1}{T} \sum_{n=1}^{T} \sum_{i=1}^{n} \Gamma_1^i \cdot F_d(i) \\ &= \frac{1}{T} D_c(0) \frac{1}{1-\Gamma_1} + \frac{\Gamma_2}{1-\Gamma_1} \frac{1}{T} \sum_{i=1}^{T} F_d(i)(1-\Gamma_1^{T-i}). \end{aligned}$$

Note that $F_d(n)$ has a upper bound $C = 255 \times 255$, and

$$\sum_{i=1}^{T} F_d(i)(1-\Gamma_1^{T-i}) \leq C \frac{1}{1-\Gamma_1}. \tag{15}$$

Therefore, we have

$$\begin{aligned} \bar{\mathcal{D}}_c &= \lim_{T \to \infty} \bar{\mathcal{D}}_c(T) = \frac{\Gamma_2}{1-\Gamma_1} E[F_d(n)] &(16) \\ &= \frac{a}{(1-b+b\beta)} \frac{p}{1-p} E[F_d(n)], &(17) \end{aligned}$$

where $E[F_d(n)]$ is the mean value of the frame difference $F_d(n)$. The experimental results reported in [3, 6, 11] confirms the analysis given by (17). From (17), we observe that, asymptotically, the average channel distortion caused by packet loss is proportional to the mean frame difference.

From Eq. (7) we know for a given streaming quality level $\bar{D}_s$, the traffic profile of video source is approximated by

$$R(\bar{D}_s, n) = \frac{1}{2}\gamma \log_2[1 + \frac{F_d(n)}{\bar{D}_s}]. \tag{18}$$

For the whole video clip, the average transmission bit rate is given by

$$\bar{R}(\bar{D}_s) = \frac{1}{T}\sum_{i=1}^{T} R(\bar{D}_s, n) = \frac{1}{2T}\gamma \sum_{i=1}^{T} \log_2[\frac{\bar{D}_s + F_d(n)}{\bar{D}_s}] = \frac{1}{2}\gamma \log_2 \frac{H[F_d(n)]}{\bar{D}_s}, \tag{19}$$

where

$$H[F_d(n)] = \{\prod_{i=1}^{T}[\bar{D}_s + F_d(n)]\}^{\frac{1}{T}} \simeq \bar{D}_s + [\prod_{i=1}^{T} F_d(n)]^{\frac{1}{T}}. \tag{20}$$

Let

$$E_g[F_d(n)] = [\prod_{i=1}^{T} F_d(n)]^{\frac{1}{T}}, \tag{21}$$

which is the geometric average of the $\{F_d(n)\}$. The end-to-end distortion can be written as

$$\bar{D} = \bar{D}_s + \bar{D}_c \tag{22}$$

$$= (2^{\frac{1}{\gamma}\bar{R}} - 1)^{-1} E_g[F_d(n)] + \frac{a}{(1 - b + b\beta)} \frac{p}{1-p} E[F_d(n)]. \tag{23}$$

This model describes the basic relationship between the end-to-end video streaming quality $\bar{D}$ and the average bandwidth $\bar{R}$, scene characteristics $E[F_d(n)]$ and $E_g[F_d(n)]$, and the network transmission condition p_e. Although the model needs further improvement, it provides a theoretical basis and practical guideline for resource allocation, QoS provisioning, and performance optimization for video encoding and streaming over the wireless network.

VI Concluding Remarks

In video encoding and streaming over the wireless ad hoc network, the system performance is measured by the end-to-end video streaming quality. In this work, we have analyzed the major components which have significant impact to the end-to-end video quality. Specifically, at the application layer, we have developed a scheme to estimate the traffic profile function, which is used for traffic control, resource allocation, and QoS provisioning. At the MAC layer and the physical layer, we have studied the behavior of the hybrid FEC-ARQ, and analyze the effective transmission power. From our analysis, we can see that the original frame difference, which describes the scene characteristics, plays an important role in both traffic analysis and transmission distortion modeling.

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