Modeling and Optimization of HFR Video Transmission over WN

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Abstract—High frame rate (HFR) video is emerging as a new paradigm in popular multimedia applications (*e.g.*, cloud gaming) to achieve smooth viewing experience perceived by end-users. In the context of HFR streaming video, *end-to-end distortion* and *sending frame rate* are equally important to the perceptual quali-ty. This study presents a modeling-based approach to optimize the HFR video transmission over wireless networks. First, we develop an analytical model dubbed FRIED (Frame Rate versus vIdEo Distortion) to characterize the tradeoff between sending frame rate and end-to-end video distortion. Second, we propose a Joint frAme Selection and FEC (Forward Error Correction) cOding (JASCO) approach based on the FRIED model to optimize the transmission performance. The efficacy of the proposed JASCO is evaluated through extensive semi-physical emulations in Exata involving H.264 video streaming. Experimental results show that JASCO outperforms the reference approaches in improving video peak signal-to-noise ratio (PSNR) at the same frame rate. Or conversely, JASCO is able to achieve higher received frame rate while guaranteeing the same video PSNR.

Index Terms—high frame rate video; wireless video commu-nication; model-driven optimization; forward error correction

I. INTRODUCTION

High frame rate (HFR) video is becoming widely deployed in popular multimedia applications (*e.g.*, cloud gaming [1] and Youtube) to provide fluent viewing experience. The perceptual quality improvement resulted from the increased frame rate is acknowledged in both industry and academic communities [1-7]. With the prevalence of powerful media servers, HFR video is expected to dominant emerging multimedia applications for providing high-quality video services [*e.g.*, the ClouUnion and StreamMyGame gaming systems [8] already provide stream-ing video up to 60 frames per second (fps)].

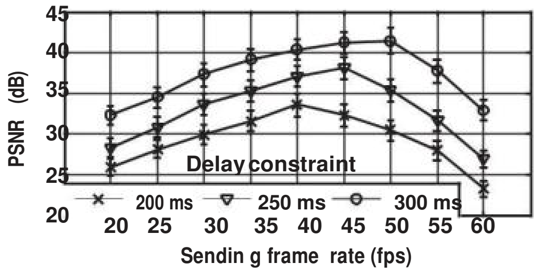
In this paper, we focus on the mobile HFR video applica-tions with stringent delay constraint (*e.g.*, live sports program, video call, multi-player cloud gaming, *etc.*). It is a challenging issue to guarantee the delivery quality of HFR streaming video over wireless networks and the main reasons are three-fold:

1) ***Throughput demand***. HFR video traffic is featured by high transmission rate and the throughput demand is mainly between 2 – 6 Mbps [1][3].
2) ***Bandwidth limitation***. Wireless network resources are scarce and time-varying. Recent studies [9][10] reveal that the available bandwidth for individual users in 4G LTE generally ranges from 1.5 to 3 Mbps.
3) ***Delay constraint***. Real-time HFR video applications are imposed with tight delay constraint as the round trip time (RTT) should be less than 130 ms to guarantee the playback fluency [1][11][12].

impacted by the *end-to-end distortion* [13][14]. Of equal importance

To address these critical problems, it is necessary to study the special features of HFR video streaming so as to maxi-mize the perceived quality with the scarce wireless resources. Conventionally, the user-perceived quality in video streaming environment is for the perceptual quality of HFR video, the playback fluency is dependent on the *sending frame rate*[1] controlled at the transmitter side. However, there is an inherent tradeoff between the end-to-end distortion and sending frame rate with regard to the bandwidth and delay constraints as shown in Figure 1. The emulations are performed with a HFR video streaming (2.5 Mbps, 60 fps) over a cellular network with the bandwidth limitation of 2.6 Mbps. Section introduces the detailed specifications of network environ-ment and video coding parameters. As shown in the plots of Figure 1, the sending frame rate to achieve minimal distortion will decrease under tighter delay constraints. On one side, a higher sending frame rate leads to a larger video traffic load. While the received video frames out of the deadline cannot contribute to the decoding process, but also results in the expired arrivals of subsequent frames and increases the end-to-end distortion; On the other side, decreasing the sending frame rate helps to reduce the probability of overdue frames and mitigate end-to-end distortion, but at the sacrifice of playback smoothness. Therefore, the key research issue in optimizing the HFR wireless video delivery is to understand and leverage the tradeoff between sending frame rate and end-to-end distortion. Towards this end, we present in this paper a modeling-based approach to optimize the real-time HFR video transmission over wireless networks. The proposed scheme is an application-layer protocol and can be implemented in existing multimedia communication systems without losing compatibility with the TCP/IP suite. [4] proposed a system in which the cross-diamond search algorithm employs two diamond search patterns (a large and small) and a halfway-stop technique. It finds small motion vectors with fewer search points than the DS algorithm while maintaining similar or even better search quality. The efficient Three Step Search (E3SS) algorithm requires less computation and performs better in terms of PSNR.

[1]The number of video frames delivered to the destination per second and this parameter is less than or equal to the encoding frame rate.



Profiling the tradeoffbetween sending frame rateand peak signal-to-noise ratio (PSNR) (*i.e.*, $PSNR = 20 \times \log_{10} \sqrt{\frac{255}{Distortion}}$) with regard to different delay constraints.

In particular, the contributions of this study can be summa-rized in the following.

- Develop an analytical model (FRIED, Frame Rate versus vIdEo Distortion) to capture the tradeoff between send-ing frame rate and end-to-end distortion of HFR video transmission over wireless networks.
- Propose a model-driven optimization approach dubbed JASCO [Joint frAme Selection and FEC (Forward Error Correction) cOding] that effectively integrates the follow-ing components.
  1) an online video frame selection algorithm to mini-mize the end-to-end distortion while achieving tar-get frame rate.
  2) a FEC coding scheme that dynamically adjusts the redundancy value and packet size to achieve imposed quality requirement based on utility maxi-mization theory.
- Perform extensive semi-physical emulations in Exata in-volving HFR video streaming encoded with H.264 codec. Experimental results show that:
  1) JASCO improves the video PSNR by up to 6.5, 9.3, and 12.2 dB compared to the Adaptive, HARQ [15], and JSS [16] schemes, respectively.
  2) JASCO reduces the end-to-end delay by up to 22.3, 30.3, and 38.6 ms compared to the Adaptive, HARQ, and JSS schemes, respectively.
  3) JASCO increases the received frame rate by up to 11.43, 15.62, and 19.3 fps compared to the Adaptive, HARQ, and JSS schemes, respectively.

The remainder of this paper is structured as follows. Section II briefly reviews and discusses the related work to this study. In Section III, we introduce the proposed FRIED model in detail. Section IV presents the solution procedure of the modeling based optimization approach. Section V provides the performance evaluation and concluding remarks are given in Section VI.

II. EXISTING SYSTEM

The related studies to this paper can be generally classified into two categories: 1) high frame rate video; 2) real-time wireless video transmission. In the rest part of this section, we will provide discussions on each topic, respectively.

A. High Frame Rate Video

HFR video refers to the video streaming with frame rates higher than the typical prior practice of 24 fps. Popular HFR video applications include the sports program, HDTV broadcasts, cloud gaming [1] and movies (*e.g.*, Hobbit and Avartar 2). The recent studies on high frame rate video technology are mainly focused on the video coding algorithms. Suo *et al.* address the problem of combinational super-resolving and denoising. The authors propose to conduct noise separation and super resolution under a unified optimization framework. Miao *et al.* [2] propose a two-layer coding scheme for HFR screen video compression. The non-skip screen frames are directly compressed with the conventional codec in the base layer. The screen contents sensitive to the video quality degradation are selected for improved coding in the enhancement layer. In reference [50], the authors model the impact of frame rate on perceptual quality of video. Wu *et al.* [7] develop an adaptive transmission framework that performs frame-level distortion estimation and scheduling to enable HFR video streaming in mobile cloud gaming applications. However, the tradeoff between frame rate and video distortion is not considered in reference [7]. This research explores this relationship to optimize real-time HFR video transmission over wireless networks.

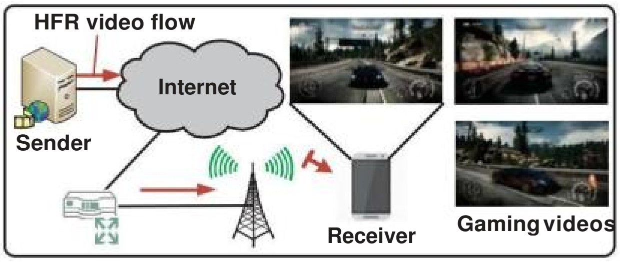
HFR video technology is expected to dominate popular multimedia applications since it is able to provide higher quality, *e.g.*, in smoothness and clarity [1-6].

B. Real-Time Wireless Video Transmission

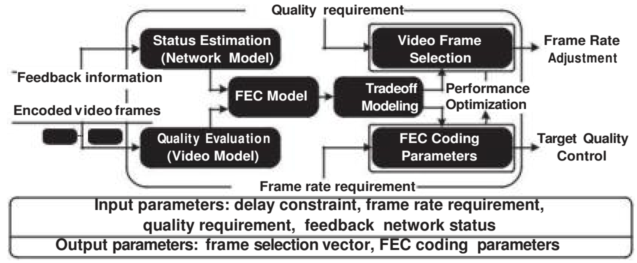
Real-time video transmission over wireless networks has been extensively investigated in recent studies [17-25] due to the popularity of mobile video.

Gong *et al.* [17] propose an adaptive transmission frame-work that jointly exploits intra-refreshment and modulation code selection for low-delay encoded video. Soltani *et al.* [18] propose a delay constraint packet embedded error control (DC-PEEC) protocol, which ensures reliable and rapid delivery of video packets by employing various channel codes to minimize fluctuations in throughput and provide timely arrival of video packets. Karande *et al.* [19] analyze the utility of hybrid erasure error protocols (HEEPs) for efficient video transmis-sion. The authors make a comparative analysis to identify the conditions under which the HEEPs can provide improved performance over conventional protocols. Song *et al.* [41] present a Probabilistic Multipath Transmission (PMT) scheme, which sends video traffic bursts over multiple available paths based on a probability generation function of packet delay. Ismail *et al.* [42] propose a content and energy aware packet scheduling approach for video upload service in heterogeneous networks.

The frame-level FEC coding is proposed in [20] to reduce coding latency but the source packet number is often not large enough to achieve efficient coding in standard-quality video. Huang *et al.* [12] propose a hybrid FEC/ARQ protocol built on a packet streaming code. Xiao *et al.* [21] propose a Dynamic Sub-Gop FEC coding scheme to optimize real-time



(a) Transmission scenario



(b) Model overview

video quality. However, the network transmission delay and bandwidth fluctuations are not addressed in this approach.

In [22][55], the authors propose a sub-frame level schedul-ing approach to minimize the end-to-end delay of high-definition (HD) video streaming over multiple wireless access networks. The video frame selection and scheduling algorithm-s are proposed in [16][23] but the time-varying channel status is not addressed in these solutions. Wu *et al.* [24] propose a differentiated packet scheduling approach to trade delay for distortion in multiple-destination real-time video transmission.

To the best of our knowledge, this is the first work to model and exploit the tradeoff between sending frame rate and end-to-end distortion for optimizing HFR wireless video delivery under stringent delay constraint.

TABLE I

MATHEMATICAL NOTATIONS.

Symbol	Definition
P, E	the probability value, expectation value.
$\boldsymbol{T}$	the imposed delay constraint.
$RT\,T$, μ	the round trip time, available bandwidth.
$P_{i,j}(\theta)$	the transition probability from i to j in time θ.
G/B	the Good/Bad state of the communication link.
ξ_G	the state transition probability of p from B to G.
π_B	the average packet loss rate.
n, k	the number of data, source packets in a FEC block.
L	the number of lost source packets in a FEC block.
$\boldsymbol{\Pi}$	the effective data loss rate.
π_t, π_o	the transmission, overdue loss rate.
$\boldsymbol{Y}$	the FEC packet size.
S, F	the sending, encoding frame rate.
$\underline{S}$	the sending frame rate requirement.
λ	the video source (encoding) rate.
D,D	the end-to-end distortion, quality requirement.
D_{src}, D_{chl}	the source, channel distortion.
M	the number of video frames in a group of pictures.
φ_m	the current status of video frame m.
d	the end-to-end delay.
U, U	the system utility, utility variation.

III. PROPOSED SYSTEM

A. Model Overview

As shown in Figure 2a, we consider the problem of end-to-end HFR video transmission from a transmitter (sender) to a mobile receiver (client) over wireless packet switching networks. The typical transmission scenarios include mobile video call, cloud gaming, sports program, *etc*. Figure 2b presents the analytical framework of the proposed FRIED model. The proposed framework includes the mathematical models of wireless network, video distortion, and systemat-ic FEC coding. Specifically, the encoded video frames and feedback channel status are the input items for the proposed model in each decision epoch. The video quality and send-ing frame rate requirements are imposed by the HFR video applications. We consider FEC-coding based wireless video communication because it is pervasively used in real-time multimedia transmission systems as the error-resilient scheme [26]. The tradeoff between sending frame rate and end-to-end distortion can be analytically described based on the above mathematical models. To exploit this important tradeoff to optimize transmission performance, we propose to selectively schedule the video frames and adapt FEC coding parameters. We will describe the mathematical models and analyze the tradeoff in the rest of this section. The basic notations used throughout this paper are listed in Table I.

B. Model Description

Wireless Network Model

The wireless network represents the end-to-end communication link between transmission terminals and includes

ratio of data packets failed to arrive at the destina-tion while travelling across the communication network. These packet drops can be caused by network congestion, channel fading, external inference, *etc*.

In addition to the above metrics, we model the burst loss behavior using the Gilbert loss model [28], which can be expressed as a two-state stationary continuous time Markov chain. The state $X(t)$ at time t assumes one of two values: G (Good) or B (Bad). In this work, we assume the packet loss rate to be zero when the communication path

Video Distortion Model

To optimize the real-time streaming video quality, we employ a generic end-to-end video distortion model [13]. The user-perceived quality in video streaming environment is impacted by the the end- to-end distortion (D). Specifically, D is the sum of two major categories of distortion: source distortion (D_{src}) and channel distortion (D_{chl}), *i.e.*,

$$D = D_{src} + D_{chl}. \quad (1)$$

This analytic model indicates the streaming video quality depends on both the distortion caused by the data compres-sion of the media information and the distortion due to the transmission impairments in the communication network. The source distortion is mostly driven by the video encoding rate (λ) and the video sequence content. These parameters largely impact the efficiency of video codec (*e.g.*, the larger distortion will be induced for a more complex video sequence under the same video encoding rate). The channel distortion is mainly impacted by the *effective data loss rate* (Π) defined as follows.

Definition 1: The effective data loss rate (Π) is the ratio of lost video data after the FEC recovery process. This loss probability includes both the channel errors/losses and expired packet arrivals.

D_{chl} is generally proportional to the number of lost video frames. Specifically, the end-to-end distortion can be expressed [in units of Mean Squared Error (MSE)] as [13][14]:

α

bytes are often padded to keep the same size with other packets. The FEC packet size (Y) also affects the tradeoff between loss recoverability and end-to-end delay in the presence of burst losses. With a smaller FEC packet size, the value of n (FEC block size) will become larger and there is higher possibility for the receiver to recover the lost packets. However, a larger FEC block size also delays the FEC decoding because the recovery process can only start after the client has received k FEC packets.

Effective Data Loss Rate

As explained in Definition 1, the effective data loss rate is the

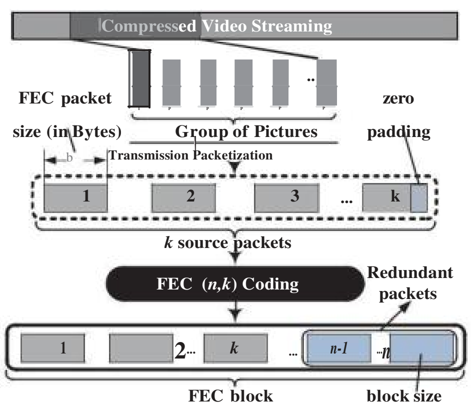
$$D = \overline{\lambda - \lambda_0} + \beta \cdot \Pi, \qquad (2)$$

Dchl

in which α, λ_0 and β are parameters depending on specific video codec and sequence. These parameters can be online estimated by using trial encodings at the sender side [10][14]. To enable fast adaptation of the transmission scheduling to abrupt changes in the video content, these parameters can be updated for each group of pictures (GoP).

FEC Model

We employ the systematic Reed-Solomon (RS) [26] block erasure code for video data protection against wireless channel losses. As shown in Figure 3, a FEC block of n data packets contains k source packets and $n - k$ redundant packets. The receiver is able to reconstruct all the k source packets if any k packets of the FEC block are correctly received. In the FEC(n, k) coding, ($n - k$) redundant data packets are introduced for every k source packets to make up a codeword. If the destination receives at least k out of the n data packets, it can recover all the source packets. If the number of received packet is less than k, these received packets can still be used for the decoding process due to the systematic nature of the FEC code.



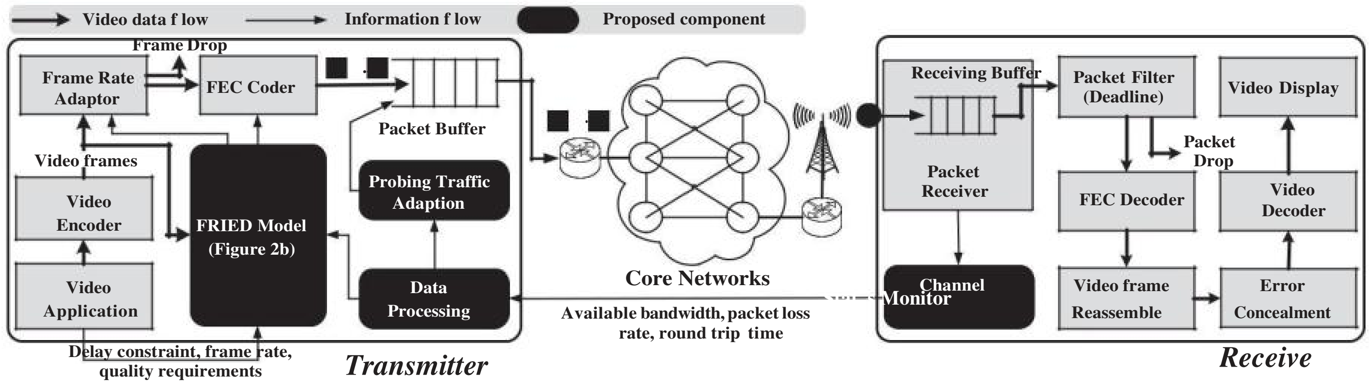
combined probability of the transmission and overdue loss rates defined as follows.

Definition 2: Transmission loss rate (π_t) is the ratio of FEC packets that encounter channel losses or buffer overflows in packet switching networks.

Definition 3: Overdue loss rate (π_o) is the probability of FEC packets violating the delay constraint imposed by the video applications (i.e., expired arrival at the destination).

To facilitate the analysis, we assume the overdue loss rate is imposed on the received video data that experience the channel losses and FEC recovery process.

IV. SYSTEM ARCHITECTURE



case of optimal scheduling and such problems prove to be NP-complete. First, it is computationally prohibitive to obtain the expectation value of π_t by considering all the possible failure configurations (c) and interleaving levels (θ). Second, it is also important and necessary to take into account the frame priority and decoding dependency. To solve this optimization problem, we introduce a heuristic algorithms for the video frame selection.

Proposed Algorithm

The online video frame selection is motivated by the fact that *an appropriate level of dropping lower-weight video frames helps to conserve network resources for protecting more important frames, and thus improve the overall streaming quality* [16][23]. The first step is to estimate the effective data loss rate Π according to the available information. For each transmission scheduling cycle, the feedback channel status is collected from the communication network and the raw data is smoothed using the Exponential Weighted Moving Average (EWMA) with a parameter of 0.5 [51].

In order to estimate the effective data loss rate Π, the transmission loss rate is obtained through Monte Carlo simulations that stochastically generates pattern of packet losses with the state transition probabilities. We assume the even packet interleaving (*i.e.*, $\theta_i = \theta_{i+1}$, $1 \leq i \leq n-1$) is employed in order to facilitate the analysis. Given the channel status information, the Monte Carlo approaches that use the repeated random sampling are able to obtain reliable values for the pattern and number of losses in video bitstream if the number of simulations is adequate. After estimating the overdue loss rate with Equation (8), we can obtain the effective loss rate with Equation (3) to generate the combined packet loss pattern. The impaired video data can be obtained with the encoded video stream and the analytical results of effective data loss rate.

We propose to drop the frames based on the frame priority in an iterative manner. In each iteration, the FEC coding al-gorithm is invoked to estimate the redundancy level. The loop will terminate if the end-to-end distortion cannot be further reduced or the lower bound of sending frame rate is reached. The online video frame selection process is summarized in Algorithm 1 and the following propositions summarize the

Line 1: Smoothing the feedback channel status with moving average.

Line 2: Set the initial values of $\{\varphi_m\}_{1\leq m\leq M}$ as 1 for all the video frames.

Line 3: Reorder the video frames based on priority in ascending order.

Line 4-17: The search loop to determine the scheduling action for all the video frames.

Line 5: Drop the video frame with lowest priority.

Line 6: Estimate the FEC coding parameters (n, k) with Algorithm 2.

Line 7-9: Calculate the transmission loss rate, overdue loss rate and end-to-end
distortion.

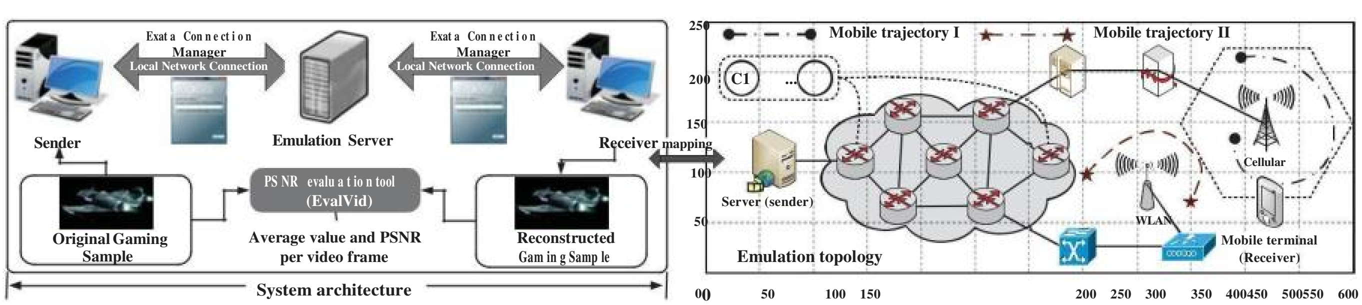
Line 11-13: Continue to drop video frames if the total traffic rate is larger than the
available bandwidth.

Line 14-16: Break the loop if the sending frame rate is lower than the requirement
or the end-to-end distortion cannot be further reduced.

V. PERFORMANCE EVALUATION

In this section, the efficacy of the proposed JASCO frame-work isperformance metrics, and ref-erence schemes. Then, we present and evaluated via semi-physical emulations over the Exata platform. First, wediscuss the evaluation results in detail.
describe the evaluation methodology that includes the emulation setup,

A. Evaluation Methodology



System architectureand emulation topology for performance evaluation.

VI. CONCLUSION AND DISCUSSION

Motivated by the increasing popularity of HFR video ser-vices, we explore two key quality metrics associated with such streaming video, *i.e.*, end-to-end distortion and sending frame rate. Towards understanding the tradeoff between the two key metrics, this research develops an analytical model dubbed FRIED (Frame Rate versus vIdEo Distortion). Then, we present a joint frame selection and FEC coding (JASCO) approach based on this analytical model to optimize the HFR video transmission over wireless networks. Extensive evaluation results demonstrate that JASCO outperforms exist-ing transmission schemes in improving the video PSNR and received frame rate.

As future work, we will consider the following aspects:

- The transmission of multiple HFR video streams with different quality and frame rate requirements. In this scenario, it is important to leverage the tradeoff to satisfy the differential service requirements in video distortion and frame rate.
- Recent studies reveal the limitations of PSNR in correlat-ing with video quality [43][44]. It is a promising solution to model and leverage the relationship between sending frame rate and perceptual visual quality metrics [45][46] to optimize the experience of HFR video.
- Transmission Control Protocol (TCP) has been pervasive-ly adopted as the transport protocol in real-time video applications (*e.g.*, mobile Skype and Web RTC). It is im-portant to study the real-time HFR video communication using TCP by leveraging the special features (*e.g.*, the delay-friendliness [53]).

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