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TRADER: A reliable transmission scheme to video conferencing applications over the internet

Jiyan Wu [a,b,*], Yanlei Shang [a], Chau Yuen [b], Bo Cheng [a], Junliang Chen [a]

[a] State Key Laboratory of Networking and Switching Technology, Beijing University of Posts and Telecommunications, Beijing 100876, PR China
[b] Singapore University of Technology and Design, 20 Dover Drive, Singapore 138682, Singapore

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ABSTRACT

In this paper, we investigate the problem of packet transmissions in real-time video conferencing from a single source to multiple destinations. To overcome the burst path losses over the Internet, we propose a transmission scheme dubbed *TRA*ding *DE*lay for *R*eliability (TRADER), which appropriately spreads out the FEC (Forward Error Correction) packets' departures with respect to the delay constraint. The network measurements from a video conferencing system and the PlanetLab demonstrate that the transmission delays to different destinations mainly differ by 100–250 ms. Motivated by this observation, we present a mathematical formulation and a systematic design to effectively leverage the delay differences for optimizing the receivers' video quality. The superiority of TRADER over existing transmission schemes is analyzed based on Gilbert model and continuous time Markov chain. We conduct performance evaluation via semi-physical emulations in Exata and experimental results show that TRADER outperforms the previous approaches in improving video quality in terms of PSNR (Peak Signal-to-Noise Ratio).

1. Introduction

The past few years have witnessed the exponential growth of video conferencing applications (e.g., Skype, http://www.skype.com, MSN Messenger, http://www.msn.com, and cloud gaming) over the Internet. In the conventional centralized service architecture (see Fig. 1), every conference peer sends its video session to a centralized server. Then, the server is responsible for replicating, transcoding and redistributing the sessions to all other participating peers. However, it is a crucial challenge to guarantee the Quality of Service (QoS) in such a scenario and the reasons can be summarized in the following:

- Video conferencing/telephony applications have stringent reliability requirements over two 9s (99%) (Li et al., 2007). Without error-resilience schemes, it is difficult even impossible to achieve the goal.
- Real-time video communications impose tight delay constraints of a few hundreds of milliseconds. Recent researches (Wu et al., 2013; Song and Zhuang, 2012) suggest that the worst case one-way delay should not exceed 400 ms to achieve acceptable video quality.
- Numerous measurement studies (Jiang and Schulzrinne, 2002; Yajnik et al., 1999) have reported that Internet packet losses often exhibit burstiness. Such behaviors significantly affect the effectiveness of data protection schemes.
- The heterogeneity in users' contexts (e.g., network conditions and end devices) makes it challenging to carry out effective QoS guarantee strategies.

To address the critical problems, there are four categories of research conducted in the past decades: (1) error resilient video coding techniques (e.g., the intra-macroblock updates, Cote and Kossentini, 1999 and redundant slices, Baccichet et al., 2007), (2) error concealment methods (e.g., spatial interpolation, Salama et al., 1998 and motion-compensated temporal prediction, Wu et al., 2000), (3) Forward Error Correction (FEC) mechanisms (e.g., Reed–Solomon codes, Frossard, 2001 and Luby Transform codes, Luby, 2002), and (4) packet transmission schemes (e.g., the Uneven Packet Transmission Rate (UPTR), Lu and Wah, 2010).

In this paper, we propose a novel transmission scheme dubbed TRAding DElay for Reliability (TRADER), which aims at leveraging the transmission delay differences among different destinations (receivers) to spread out the departures of FEC packets. Different from previous UPTR (Lu and Wah, 2010) schemes, the proposed TRADER is able to mitigate the burst packet loss without increasing the transmission

* Corresponding author. Tel.: +65 63036660.
E-mail addresses: wujiyan@126.com (J. Wu), shangyl@bupt.edu.cn (Y. Shang), yuenchau@sutd.edu.sg (C. Yuen), chengbo@bupt.edu.cn (B. Cheng), chjl@bupt.edu.cn (J. Chen).

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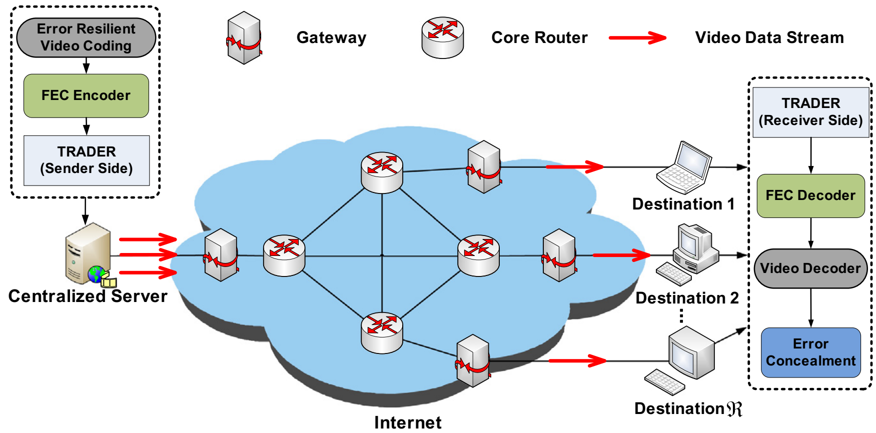

Fig. 1. Illustration of a video conferencing scenario from a single source to $\Re$ destinations over the lossy Internet. The proposed TRADER can be integrated with other error resilient solutions to improve the reliability.

rates. The main motivation for the proposal stems from the practically observed transmission delay differences, which we will discuss in Section 2.

The contributions of this paper are three-fold:

1. We conduct the network measurements of packet transmission delays to different destinations both from a real video conferencing system and the PlanetLab, finding that these delays mainly differ by 100–250 ms.
2. We present a mathematical formulation of transmission scheduling to minimize the end-to-end video distortion under capacity and delay constraints. Then, we introduce a systematic design to effectively leverage the transmission delay differences to evenly spread out the FEC packets' departures. The superiority of this scheme over the existing approaches is analyzed through a derivation of the transmission loss based on Gilbert model and continuous time Markov chain.
3. We evaluate the performance of TRADER through Exata emulations using real-time H.264 video streaming and the experimental results show that (1) TRADER improves the average video PSNR (Peak Signal-to-Noise Ratio) by up to 7.8, 10.2 and 11.3 dB compared to the OPTICS (Lu and Wah, 2010), TRICE (Lu and Wah, 2010) and FPTR (Fixed Packet Transmission Rate) schemes, respectively. (2) TRADER guarantees more than 98% video frames delivering within the delay constraint while the ratios for OPTICS, TRICE and FPTR are 88%, 81.3% and 78.3%, respectively. (3) TRADER achieves better PSNR and delay performance while picking the largest transmission delay of all conference peers' as reference than the delay constraint.

The remainder of this paper is organized as follows. In Section 2, we briefly review the related work and discuss the motivation for the proposed TRADER. Section 3 presents the system model and problem formulation. In Section 4, we describe the design of the proposed TRADER and compare it with the existing transmission schemes through theoretical proof. Performance evaluation is provided in Section 5 and concluding remarks are given in Section 6. The basic notations used throughout this paper are listed in Table 1.

2. Related work and research motivation

In this section, we first review the related work. Then, we discuss our research motivation.

Table 1
Basic notations.

Symbol	Definition
S	Centralized server (source node)
$\Re$	The number of destinations
D_r	The rth destination
P_r	The physical path from S to D_r
μ_r	The available bandwidth of P_r
τ_r	The path propagation delay of P_r
G/B	The Good/Bad state of P_r
π_B^r	The loss rate of P_r
μ_B^r	State transition probability of P_r from G to B
π_B^*	Transmission loss rate
Ψ	Packet transmission delay
$\tilde{\Psi}$	Transmission delay reference
Φ_{ir}	Departure time of the ith FEC packet to D_r
d_{FEC}	The delay constraint for FEC block
Δ	FEC packet size
n	Total number of data packets in a FEC block
n_r	Number of FEC packets allocated to D_r
k	Number of source data packets in a FEC block

2.1. Related work

Video conferencing applications have attracted considerable research attentions in the past decades. The existing researches can be divided into three branches: (1) application layer multicast, e.g., Akkus et al. (2011) and Zhang et al. (2012); (2) error-resilient solutions, e.g., Frossard (2001) and Yu et al. (2008); (3) packet transmission schemes, e.g., Lu and Wah (2010).

The research closest to ours is conducted by Lu and Wah (2010). The UPTR (Lu and Wah, 2010) scheme is motivated by the flexibility of packet transmission rates as the Internet is an interconnection of packet switching networks. The main innovation of UPTR is to provide unequal protection of the I frames by adjusting the transmission rates periodically. Although the transmission rate fluctuates around the average value, the delay constraint imposed by video applications may be easily violated, especially in the case of sever bandwidth fluctuations.

In Akkus et al. (2011), the authors propose a peer-to-peer (P2P) architecture for multi-point video conferencing using layered video coding at the end hosts. The architecture deals with end hosts with low-bandwidth network connections and enables them to create a multi-point video conference without any additional networking and computing resources beyond what is needed for a point-to-point conference. Zhang et al. (2012) propose an overlay multicast protocol, which organizes the multicast tree as a layered