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Unequal error protection for real-time video in mobile ad hoc networks via multi-path transport ☆

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Abstract

In this paper, we propose a mechanism that combines retransmission-based error control with multi-path transport (MPT), to provide different levels of protection to real-time video in ad hoc networks. The mechanism factors in the importance of the retransmitted packets to the reconstructed video quality as well as the end-to-end latency constraints to minimize the overhead and maximize the reconstructed video quality at the receiver. Simulation results show that the proposed retransmission mechanism maintains the video quality under different loss rates and mobility speeds, with less overhead compared to error control methods that depend on controlling the intra-update rate. In addition, the mechanism is shown to be more robust to wireless losses and mobility than schemes that combine layered and multiple description coding with multi-path transport.

Keywords: Unequal error protection; Multi-path transport; Predictive coding; Video transport over ad hoc networks

1. Introduction

Video transmission in ad hoc networks is a challenging problem, due to the high bit error rates which can range from single bit errors to burst errors or even intermittent loss of the connection. The high error rates are due to multi-path fading, which characterizes radio channels, while the loss of the connection can be due to the mobility in such networks. In addition, designing the wireless communication system to mitigate these effects can be complicated by the rapidly changing quality of the radio channel.

All common video coding schemes, including standards such as MPEG [1] and H.263 [2], are designed to achieve high compression efficiency on expense of the error resilience. The coding efficiency in these schemes is achieved by using motion-compensated prediction (MCP) to exploit the redundancy between successive frames of a video sequence. Although MCP coding can achieve high compression efficiency, it is not designed for transmission over lossy channels [3]. A motion-compensated video sequence consists of two types of video frames: *intra-frames* (I-frames) and *inter-frames* (P- or B-frames). I-frame is encoded by only removing spatial redundancy present in the frame. P-frame is encoded through motion estimation using preceding I- or P-frame as a reference frame. B-frame is encoded bi-directionally using the preceding and succeeding reference frames. For each image block in an inter-frame, motion estimation finds a closely matching block within its reference frame, and generates the displacement between the two matching blocks as a motion vector. The pixel value differences between the original inter-frame and its motion-predicted frame are encoded along with the motion vectors. Removing the temporal redundancy within the stream poses a severe problem, namely error propagation (or error spread), where errors due to a packet loss in a reference frame propagate to all of the dependent frames leading to perceptible visual artifacts.

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Error propagation can be controlled by more frequently adding intra-frames (which are coded temporally independently). However, the ratio of the compression efficiency of an intra-frame over an inter-frame is as large as 3–6 times. Increasing the frequency of intra-frames could increase the bandwidth requirement too much for video transmission over a bandwidth-constraint network. Hence, by adjusting the amount of intra-coding, error robustness can be traded off against coding efficiency [1].

The widely varying error conditions in wireless channels limit the effectiveness of classic Forward Error Correction (FEC), since a worst-case design would lead to a prohibitive amount of redundancy. Although retransmission-based error recovery requires minimal network bandwidth and processing cost than FEC based error control methods, it is often considered inappropriate for real-time video, because of the limited playout delay at the receiver, which limits the number of admissible retransmissions [4].

The advantages of using multi-path transport (MPT) have been reported in many previous papers, for example, Refs. [19–26] and the references therein. An important advantage of using multi-path transport is the inherent path diversity (i.e., the independence of loss processes of the paths). As a result, the receiver can always receive some data during any period, except when all the paths are down simultaneously, which occurs much more rarely than single path failures. In addition, multi-path transport provides a larger aggregate rate for a video session, thus reducing the distortion caused by lossy video coders. Finally, multi-path transport distributes traffic load in the network more evenly, resulting in low congestion and delay in the network. These advantages come at the cost of higher coding redundancy, higher computation complexity, and higher control traffic overhead in the network [22].

In this paper, we propose an error control mechanism for real-time video applications in mobile ad hoc networks. The mechanism extends retransmission-based error control with redundant retransmissions on diverse paths provided by the ad hoc routing protocol between the sender and receiver. Our design goal is to maintain the video quality with low overhead. The mechanism factors in the importance of the retransmitted packets to the reconstructed video quality as well as the end-to-end latency constraints to minimize the overhead and maximize the quality at the receiver. As different paths can have independent loss characteristics, sending multiple copies of the retransmitted packet on different paths can increase the probability that the packet get received in less number of retransmissions. With a network loss rate l, the error rate can be reduced to

$$\text{Error rate} = l^{1+\sum_{i=1}^{L} M_i} \quad (1)$$

where L is the maximum number of retransmission trials, which is typically determined by the initial playout delay in the receiver as well as the round-trip delay. M_i is the number of retransmission copies during the ith retransmission, which depends on the importance of the retransmitted data to the reconstructed video quality. The maximum number of copies $\text{MAX}(M_i)$ is equal to the number of available paths between the sender and receiver. The priority for each data unit in the stream is determined by the application. Thus in the context of motion-compensated coding, the application can assign higher priority for I-frames data than P- or B-frames data. Also P-frames might be assigned varying priority levels, since P-frames that are closer to the preceding I-frame are more valuable for preserving picture quality than later P-frames in the group of pictures (GOP). This prioritization scheme can also be applied on the macro-block basis for encoding schemes with the flexibility to select the coding mode (i.e., intra- or inter-coding) on the macro-block level [2]. To ensure in-time delivery of retransmitted packets, and to prevent retransmitting expired packets, the retransmission is controlled by the packet lifetime, as well as estimate(s) of the path(s) delay. As will be shown in our performance study in Section 4, this can significantly limit the effect of error propagation and improves the quality of received video, under a limited playout delay.

As the interactivity of the video session can be hurt by the continuous failure of the transmission paths, which can be due to mobility, our mechanism monitors the paths one-way-delay (OWD) to the receiver, and accordingly it switches between them seamlessly.

We implemented the proposed mechanism as a sub-layer above real-time transport protocol (RTP) [5]. We refer to this sub-layer as multi-path-RTP (MP-RTP). MP-RTP is responsible for: (i) maintaining the priority level and the lifetime for each packet, as specified by the application, as well as implementing a delay constrained retransmission and (ii) monitoring the status of the available paths, and switching between them.

We compared the proposed mechanism to error control methods that depend on intra-frame updates, as well as mechanisms that combine layered coding (LC) and multiple description coding (MDC) with multi-path transport. Simulation results indicate that the proposed mechanism performs significantly better than reference frame update schemes in terms of perceived quality measured at the receiver as well as the transmission overhead. It is also more robust to wireless losses than LC and MDC schemes under comparable path conditions.

This paper is organized as follows. Section 2 provides a review for related works. The proposed mechanism is presented in detail in Section 3. We present experimental results and performance evaluation in Section 4. Finally, conclusions and future work are outlined in Section 5.

2. Related work

Boyce and Gaglianello [6] presented experimental results for the packet loss characteristics for MPEG video. Their results suggested the need for better error recovery and concealment techniques. Similarly, Feamster and Balakrishnan [7] analyzed the effects of packet loss on the qual-