WIRELESS VIDEO TRANSPORT USING PATH DIVERSITY: MULTIPLE DESCRIPTION VS. LAYERED CODING

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ABSTRACT

Typical video applications may need a higher bandwidth and/or higher reliability connection than that provided by a single link in current or emerging wireless networks. We propose to employ path diversity to provide higher bandwidth and more robust end-to-end connections than that affordable by a single path. Under this transport environment, two viable strategies for video coding are multiple description coding (MDC) and layered coding (LC). MDC is more effective when the underlying application has a very stringent delay constraint and the round trip time on each path is relatively long. LC can be a good alternative when limited retransmission of the base layer is acceptable and when it is feasible to apply unequal error protection over different paths. This paper describes the general issues involved in integrating MDC/LC with multiple path transport, and compares the performances of MDC and LC, under different path conditions.

1. INTRODUCTION

Video transport over wireless networks is challenging because the wireless links are unreliable and have limited bandwidth. Typical video applications may need higher bandwidth and/or higher reliability connections than that provided by a single link. In a network consisting of mobile nodes, the connection path between a source and a destination is constantly broken and has to be frequently updated. Although, following a path failure, one could switch over to an alternative route, this may take an unacceptably long period of time, causing a temporary disruption of a video session. To overcome both the limitation in link-level reliability and bandwidth and the time-varying nature of the network topology, we propose to establish multiple paths between the source and destination in a single virtual connection, so as to enhance the system robustness while increasing the usable bandwidth for an end-to-end connection beyond that of a physical link.

Multiple path transport (MPT) (also known as path or route diversity) schemes have been proposed in the past for wired networks for increased connection capacity and reliability for data transmission.We believe that MPT has more potential in wireless networks where individual physical links may not have adequate capacity to support a high bandwidth service, links are unreliable, and link connectivity changes constantly. In Section 2.2, we discuss possible ways to establish multiple paths in different wireless networks.

MPT has traditionally been considered for non-real-time data transmission, where the traffic is split on the bit level in a random manner. For transmission of compressed video streams obtained using temporal prediction and variable length coding, such random splitting can make the received information bits on one path useless, if any bits in the other path are lost. Therefore, one must jointly design source coding and traffic splitting (or allocation) strategies for MPT to be actually helpful. Two viable options for source coding are multiple description coding (MDC) and layered coding (LC). Both produces multiple sub-streams that can be carried on separate paths. With MDC, the sub-streams (each called a description) have equal importance in the sense that each received description alone can guarantee a basic level of reconstruction quality, and additional descriptions can further improve the quality. Because the loss of one description does not influence other descriptions, a lost packet in any path does not need to be retransmitted. On the other hand, with LC, the base-layer stream is more important and can provide a basic level of quality, whereas remaining enhancement-layer streams serve to refine the base-layer quality; the enhancement layers alone are not useful. The path carrying the base-layer packets should have a higher reliability, either naturally or through forward error correction (FEC), and any lost base-layer packets should ideally be retransmitted. Obviously, the choice of the coding strategy depends on the path conditions and the delay requirement of the underlying application.

We first considered using MPT in multihop wireless networks for non-real-time data in [1]. This study was later extended for image and video transmission, where we con-

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sidered the integration of MDC with MPT, and compared the performance obtainable with MDC using two symmetric paths and that with LC using asymmetric paths [2, 3]. Several other groups have also considered using MDC and path diversity, mainly for multimedia transport over the best-effort Internet [4, 5]. In this paper, we first briefly review the general system architecture proposed in [2, 3] and highlight the issues that call for cooperation between the source coder and the transport layer. In that work, each video frame is coded independently using a Lapped Orthogonal Transform (LOT)-based coder. The MD and layered coder differ in the LOT basis used and the way the LOT coefficients are split between the two streams. Retransmission is not simulated explicitly for the layered coder. In this paper, we report simulation results obtained with more efficient MD and layered coders, both incorporating motion-compensated temporal prediction. For the LC case, we consider two scenarios: one in which the underlying application does not permit retransmission, another where a selective retransmission scheme is applied to the base-layer.

2. SYSTEM OVERVIEW

2.1. The General System Architecture

The general architecture of the proposed system is shown in Figure 1. We assume that at any time instant, one can set up K paths between the source and destination, each with a set of quality of service (QoS) parameters in terms of bandwidth, delay, and loss probabilities. The transport layer continuously monitors path QoS parameters and feeds back such information to the sender. Based on path quality information, the coder at the sender generates M multiple bitstreams using either MDC or LC. Each substream is divided into smaller units so that each unit is carried in a single transport packet. The packets from different substreams are distributed by the traffic allocator among available paths. At the receiver, the packets arriving from all the paths are put into a resequencing buffer where they are reassembled into M substreams after a preset time-out period. All or some of the packets allocated to a path may be lost because of the errors on the path or because of path breakdown. Some packets may arrive late and will also be considered lost. Depending on the underlying application, limited retransmission of lost packets may or may not be invoked. The decoder will attempt to reconstruct a video sequence from the received packets in multiple substreams.

In practice, it is likely that the number of "good" paths that can be set up between two nodes in a wireless network varies in time (say between two to ten), and that the paths differ in terms of the bandwidth allocated for this connection and the packet loss of this path. On the other hand, for reduced complexity, it may be desirable for the sender to use a predesigned MD or LC coder that always produces a fixed

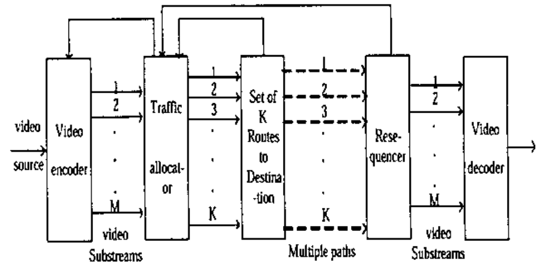

Fig. 1. General architecture of the proposed system using multistream coding and multipath transport.

number of substreams (say two to four) with fixed rate ratios. In fact, most MD and layered coders proposed thus far belong to this category. This is the reason that we consider, in Figure 1, the possibility of $M \neq K$, and suggest the use of a traffic allocator to distribute the packets from different substreams to available paths, based on the bandwidth and reliability of each path. In general, one substream may be split among several paths, or several substreams may be interleaved on a single path. Note that in the LC case, instead of producing M layers, one can also employ a scalable coder that produces a single embedded bit stream, and sequentially split this bit stream to multiple paths based on the allocated bandwidth on the paths.

A key to the success of the proposed system is the close interaction between the source coder and the transport layer. First, the transport layer must establish a number of "good" paths and allocate appropriate bandwidth on each path for the requested connection so that the total bandwidth falls in a desirable range (which depends on the statistics of the video source, the expected video quality, and the coder efficiency). The set of paths may need to be frequently updated during a single connection. Second, the source coder must choose appropriate coding mode (MDC vs. LC) based on the path conditions and application requirements. One objective of our study is to examine which coding method is more suitable for what path environments. The guidelines obtained from our simulation results are discussed in Sec. 3.3. Thirdly, the source coder has to adjust the rate allocation among substreams so that the total rate does not exceed the total allocated bandwidth. Finally, one must carefully allocate packets from different substreams among the available paths, to ensure, with high probability, a minimally acceptable reconstruction quality at the receiver. With LC streams, the base-layer packets should be delivered over more reliable paths, whereas with MD streams, the packets from different descriptions should be evenly distributed over the paths. Also, in the MD case, the allocation should be such that packets from different descriptions that carry information about nearby spatio-temporal segments of the video sequence be spread over different paths, so as to reduce the chance that these packets are simultaneously lost. By using dynamic path selection and bandwidth allocation on the network control side, adaptive coding (in terms of symmetry in importance and rate allocation) on the source coding side, and intelligent traffic allocation among multiple paths, the system can adapt quickly to changes in link-level connectivity and bandwidth.

Given that video transport can tolerate some amount of loss and may have real time delivery constraints, we use the Real-time Transport Protocol (RTP) as the transport layer protocol entity. To implement the transport layer functions in the proposed system, we introduce a layer called Meta-RTP, sitting on top of RTP in the application layer of the protocol stack. This layer is responsible for traffic allocation at the sender, resequencing at the receiver, and path quality monitoring and feedback. A more detailed description of Meta-RTP layer functions can be found in [3].

2.2. Set-up of Multiple Paths in Wireless Networks

To apply the proposed system for video transport over wireless networks, the first question to be answered is how to set up multiple paths between the sender and receiver. Here we outline several alternatives. In a multihop wireless network, such as an adhoc network, each station has router-like functionality, and can discover multiple routes to the destination. For example, most of the proposed adhoc routing protocols, *e.g.*, the Zone Routing Protocol [6], have the ability to discover multiple routes. In the CDMA system, a node can communicate with multiple neighbors simultaneously by having multiple transceivers in each mobile [7, 8], and using either receiver-oriented or link-oriented codes, or a code for each transmitter-receiver pair. Analogously, in a FDMA or a TDMA based system, a mobile could talk to its neighbors using multiple frequency channels or time-slots. Finally, in a single hop wireless network such as the cellular phone network or wireless local area network, a mobile node would need to establish channels to multiple base stations instead of one. This is already done in "soft" hand-off systems, during the hand-off phase. Alternatively, the mobile and base station can each be equipped with multiple transmit and receiver antennas, as in multiple input and multiple output (MIMO) systems, and one can consider each corresponding pair of transmit and receiver antennas as constituting a separate path.

3. SIMULATION RESULTS

Although the proposed system in general can deploy arbitrary numbers of paths, we have only simulated a system with two paths. We compare several different coding and transport control options, targeted for different scenarios. For situations in which retransmission is not acceptable, either because the average round trip time over a path is not sufficiently shorter than the acceptable delay by the underlying application, or that a feedback channel is not available, we compare MDC with LC, both without retransmission. We also consider LC with retransmission, for situations where limited retransmission is feasible.

3.1. Video Coding, Packetization, and Error Control

For MD video coding, we use a recently developed multiple description motion compensation (MDMC) coder [9]. The MDMC coder makes use of two previous frames for motion compensated prediction for a current frame. It includes even frame errors in one description, and odd frame errors in another. In anticipation that the receiver may receive only one description, it also codes the mismatch signal, which is the difference between the prediction obtained with the past two frames and that from the past even (or odd) frame only. The redundancy of the coder can be controlled by varying the predictor coefficient, the quantization parameter for the mismatch signal, and the intra-block rate. Simulation studies have shown that this coder can provide a wider trade-off range between coding efficiency and error-resilience than most competing MD video coding schemes, and it requires lower redundancy to achieve the same distortion from a single description. In the simulation results presented here, each group of blocks (GOB) is assembled into one packet, but packets from even and odd frames are sent on two separate paths. No retransmission is allowed in either path.

For LC, we use the SNR scalability option in the H.263+ coder [10]. Each group of blocks (GOB) produces two packets, a base-layer packet, containing a coarsely quantized version of the signal, and an enhancement-layer packet containing a refined version. The base-layer and enhancement layer packets are delivered over two paths separately, with the base-layer packets sent over the path with lower packet loss rates when available. For the LC+ARQ simulation, we use the selective ARQ scheme proposed in [11]. Specifically, a lost base-layer packet is retransmitted on the enhancement-layer path, and the enhancement-layer packet scheduled to be transmitted at that time instance is discarded. Only one retransmission attempt is permitted.

3.2. Channel Loss Simulation

We assume two paths can be set-up for the transmission of a video sequence, and that each path has sufficient bandwidth to accommodate an entire sub-stream, so that we do not need to consider traffic allocation. Furthermore, we assume each path may consist of multiple links, and FEC code is applied at the data link layer to correct bit errors. Thus packet losses in each path can be due to link failures or FEC failures. We did not simulate the resequencing buffer at the receiver, nor the actual propagation and queuing delay. Thus the losses due to excessive resequencing as well as other delays are not considered explicitly. The packet losses are generated randomly according to a specified average loss rate. One can consider some of the packet losses are due to excessive delays. A more explicit model taking into account of all these factors is being developed in our on-going research.

3.3. Results and Discussion

We compare the performance of three coding and transport schemes: MDC without ARQ, LC without ARQ, and LC with ARQ. The QCIF sequence "Foreman" (frame 1 to 300) is encoded at 10 fps using both the MDMC coder and the layered coder described in Section 3.1. The bit rate of each description or layer is limited to 57 bkps. TMN8.0 [10] rate control method is used in the layered scheme to obtain the desired rate. For MDMC, a fixed parameter setting ($a_1 = 0.9$ and QP2=15) is used, which introduces only small amount of redundancy. With both methods, 5% MB-level random intra refreshments are used.

To examine whether the existence of a more reliable path would help to improve the video quality in either scheme, we simulate both a symmetrical case where the two paths have identical packet loss rates, and an asymmetrical case where one path has a lower packet loss rate (but the average is equal to the symmetrical case). The latter case can be either due to the inherent asymmetry in the path quality or realized by deploying unequal FEC codes over the two paths. For each pair of specified loss rates, ten packet loss traces were generated. The average PSNRs of reconstructed video frames over ten simulations are summarized in Table 1.

From Table 1, we can see i) When no retransmission is allowed, MDC outperforms LC in the loss range considered here (1-10%), more so in the symmetric case than in the asymmetric case (Note that when the better path has a packet loss rate significantly lower than considered here, the LC method may perform better than MDC, unless the MD coder can work at extremely low redundancy); ii) When the extra delay caused by retransmission is acceptable and unequal error protection is feasible, LC with ARQ is better when the loss rate in the better path is medium to high; iii) Even when the extra delay caused by retransmission is acceptable, MDC can be as good as LC+ARQ, when the paths are symmetric and the loss rate is low to medium. This is because in such cases, the redundancy injected by the MD coder can effectively suppress error propagation. The actual break point between MDC and LC+ARQ in terms of the packet loss rate depends on the actual coder implementations. The MD coder should ideally vary the redundancy based on the channel loss rate.

Table 1. Average PSNRs (dB) of Decoded Video Frames

Packet loss rate	(3%,3%)	(1%,5%)	(10%,10%)	(5%,15%)
MDMC	31.2	31.2	27.2	27.9
LC	28.0	30.6	24.0	26.9
LC+ARQ	31.2	31.4	28.7	29.9

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