

# Relay-assisted Rateless Layered Multiple Description Video Delivery

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## Abstract

Multiple description coding has been proposed as a possible solution to real-time video delivery over relay-assisted wireless networks to leverage path diversity. In this paper, we study layered multiple description video over relay-assisted mobile networks, such as LTE-A, and develop a framework that sub-optimally routes packets to different relays based on source-channel-relay parameters. Furthermore, we employ application-layer forward error correction via Random linear codes (RLC) that are very suitable in the relaying scenarios, due to their rateless nature. We achieve unequal error protection (UEP), using the recently proposed expanding window technique, that relies on probabilistic scheduling of layered packets. We develop layered multiple descriptions using data slicing and data partitioning features of H.264/AVC and conduct simulations by modelling a two-relay LTE-A setup with the finite state Markov chain model. Simulation results show benefits of relaying with optimized routing and UEP.

## Keywords

Multipath video, wireless relaying, multiple description coding.

## I. INTRODUCTION

To enable seamless and uninterrupted real-time multimedia transmission to heterogeneous mobile clients, it is necessary to exploit different diversity techniques to rectify severe noise and fading effects of wireless channels. Multipath diversity together with multiple description coding (MDC) [1] is an efficient technique to combat noise, fading and shadowing in mobile channels.

Emerging wireless communications systems, such as Long Term Evolution-Advanced (LTE-A) Release 10 [2], [3] and mobile WiMax, offer a possibility of increasing the system capacity and extending coverage via relaying and cooperation [4]. In LTE-A, for example, the relays receive and retransmit the signals between the base station and mobiles, resulting in effective increase in the throughput and extending the coverage of cellular networks [5].

To autonomously and dynamically adapt to network bandwidth, delay, and channel noise fluctuations, a promising solution is to combine multipath relaying with MDC [1], [6]. MDC generates multiple descriptions of the same source data that can be independently decoded and is very well suited to emerging relay-assisted networks, where each description can be routed via a different relay exploiting network diversity.

However, since mobile channels are very prone to packet losses, forward error correction (FEC) is still necessary to ensure that enough descriptions are correctly received.

To provide adaptive resilience to packet drops, which are frequent in mobile channels, application layer FEC (AL-FEC) has become very popular. Indeed, AL-FEC via Raptor codes [7] has become standard for Digital Video Broadcasting-Handheld (DVB-H) and Multimedia Broadcast Multicast Service (MBMS) [8]. Another class of erasure codes recently developed, is Random Linear Codes (RLC) [9], [10], [11]. Like Raptor codes, RLC are rateless, capacity approaching, and of low encoding complexity. However, rooted in the network coding principles (see [9] and references therein), RLC enable simple and efficient relaying and network cooperation.

In this paper, we propose a system for real-time video streaming to mobile users, via LTE-A and similar relaying wireless systems, using H.264 Advanced Video Coding (AVC)-based layered MDC [12] together with RLC. The source data is encoded in multiple descriptions, which are sent to a receiver over non-cooperative relays. The receiver can reconstruct the encoded data from any subset of the descriptions received. To maximize benefits of relaying and MDC, we propose two algorithms for optimal and sub-optimal, but fast, relay selection and optimal resource allocation that minimizes reconstructed source distortion. We model a network using a finite-state Markov chain (FSMC) [13] path loss channel model using LTE-A Release 10 parameters suggested in [14]. Multiple descriptions are used in conjunction with expanding window RLC (EW-RLC), originally developed in [15], enabling flexible rate adaptation and competitive performance [16]. Although we use H.264/AVC to generate multiple descriptions, the proposed scheme is general enough to be applied to other MDC schemes.

The focus of the paper is maximizing benefits of relaying for mobile video streaming. The main contributions are: (1) an adaptive scheme for multiple-relay layered MDC communication; (2) optimal relay selection and optimal source-channel rate allocation via dynamic programming; (3) multipath EW RLC as a robust solution to unequally protecting the data of each description.

The rest of the paper is structured as follows. The related work is described in Section II. In Section III, the employed system model is shown. Section IV describes

our proposed resource allocation algorithms. Section V describes MDC and EW-RLC schemes used. Simulation results are given in Section VI. The conclusion and future research directions are highlighted in the last section.

## II. RELATED WORK

In this section we review related work. The first category of related work are contributions to resource allocation and scheduling in relay networks. Information-theoretical bounds of relaying, power allocation protocols, and error control code designs are proposed in [17], [18] and references therein. In [19], different relaying strategies for LTE-A are compared in terms of spectral efficiency. The relay selection problem using statistical analysis based on instantaneous channel characteristics at the physical layer is considered in [20]. In [21] a physical-layer relay selection method is proposed that minimizes the multiplexing loss with short feedback and decode-and-forward relaying.

Resource allocation in relay-assisted communication has also been studied extensively. In [22], for an orthogonal frequency-division multiple-access (OFDMA) based relaying system, the optimization of physical-layer transmission protocols is done using a set of pricing variables as weighting factors. In [23], an amplify-and-forward wireless relay system is considered, and novel power allocation strategies are proposed, based on geometric programming, that optimize the maximum transmit power and the network throughput. In [24] a dynamic resource allocation of transmission powers and sub-channels under power constraints based on instantaneous channel states that maximize the instantaneous total transmission rate is investigated and a greedy approach proposed. In [25], a globally optimal Pareto-based solution and sequential optimization algorithm using channel state information are proposed for the OFDMA-based half-duplex single-relay channel.

In contrast to the above papers which provide “lower layers” (physical, link, and/or network layer) parameters selection, our work focusses on the application-aware resource allocation, where upper-layer parameters (such as AL FEC) are considered as well, and the reconstructed source fidelity is used as performance measure.

The second category of related work covers the path selection and resource allocation for multipath video. In [26], optimal and polynomial-time suboptimal algorithms for se-

lecting the network path and scheduling packets of encoded scalable video are proposed that minimize video distortion and time delay at the receiver. In [27], an MDC relaying is considered, and an adaptive compression scheme is proposed at the source and relay based on the received feedback. In [28] a joint source-channel scheme is proposed that exploits information about packet losses to adaptively select reference frames in MDC. In [29] FEC and routing are jointly optimized to maximize reconstructed quality.

The main difference between our paper and the above contributions is that our resource allocation strategy considers both lower and upper layers parameters in a joint cross-layer design and proposes a dynamic programming and a suboptimal greedy solution. In contrast to [28], [29], we do not alter video encoding, making it compatible with the standard, but only adapt the coding rate and relay selection based on the channel characteristics. In the proposed scheme, descriptions are generated similarly to [30], however, we create independent descriptions with least possible duplication using slicing as well as data partitioning [12].

### III. SYSTEM DESCRIPTION

In this section we describe the system model used in the paper. A source node generates source blocks that need to be streamed in real time to a mobile destination. The source node compresses independently each data block of length  $B$  symbols (e.g., bytes) into  $D$  descriptions,  $D \ll B$ , using an MDC scheme, and each description consists of  $l$  layers (e.g., quality, temporal, etc).

After source coding, the descriptions are fed into a rateless AL-FEC encoder, which for each description, generates potentially infinite stream of coded packets of  $L$  symbols each. Each encoded packet is a random linear combination of the source packets over sufficiently large field. Source can send the encoded packets to the destination via a direct link, or via  $N-1$  relays available. The  $j^{\text{th}}$  channel ( $j = 1, \dots, N$ ) is characterized by the total number of packets that can be transmitted until the deadline, which is denoted by  $R_j$  and depends on the channel bandwidth, and the packet loss probability,  $p_j$ . Note that  $R_j$  and  $p_j$  can in general be functions of time, and  $p_j$  also depends on the relay/source transmit power.

Note that though the source exploits rateless AL-FEC coding, due to real-time play-

out constraints, only a finite number of encoded packets  $R_j$  can be sent through the channel  $j$  for each encoded source block. The destination collects rateless packets from all available links, until it either receives enough to decode all descriptions or until playout deadline is reached. Note that each description can be decoded independently, the descriptions are of unequal importance to the final recovery, and each encoded description can only improve the reconstruction.

In particular, the above system can be used to model video streaming over LTE-A networks, where a base-station (source) in the pico-cell streams video to a mobile client via direct link and also via femto-cell relays as shown in Fig. 1 for the case of two femto-cell relays.

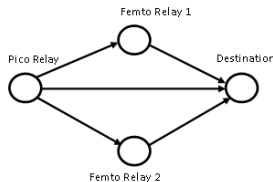


Fig. 1. The block diagram of the system with two relays.

Due to strict time constraints of real-time streaming, we assume that the employed relays do not decode the received packets but operate in the amplify-and-forward mode. As in LTE, the pico-relay or the base station (BS) uses separate orthogonal links to send packets to the relays and destination. The relays also forward the packets to the client using orthogonal channels. Since the BS and the relays are static (and these connections are often optical), in our analysis, we assume that the source-relay channels are error-free.

#### IV. PROPOSED RESOURCE ALLOCATION OPTIMIZATION

Given statistical knowledge of the source content and channel characteristics, the task is to find a source-channel-relay resource allocation that maximizes the reconstructed video quality.

Let  $\delta \subseteq \{1, \dots, D\}$  be a subset of descriptions. Let  $\pi = \{r_{ij}\}$  be a  $D \times N$  matrix of the rate allocation strategy over  $N$  channels, where  $r_{ij}$ ,  $i = 1, \dots, D$  and  $j = 1, \dots, N$ ,

is the rate (in terms of the number of encoded packets) of description  $i$  that will be sent over channel  $j$ . Note that the total number of encoded packets of description  $i$  is  $\sum_j r_{ij} = \lceil \frac{q_{s_i} q_{c_i} B}{L} \rceil$ , where  $q_{s_i}$  and  $q_{c_i}$  are source and channel coding rate for description  $i$ , respectively.

Matrix  $\pi$  is thus used at the source to select how many encoded packets from each description should be routed over each relay. However, since each description consists of  $l$  quality layers, encoded packets are not of equal importance. Thus, furthermore, for each description, the source probabilistically schedules encoded packets for transmission using channel coding probability matrix  $Q_{EW}$  (which will be explained later, see also [15]). Thus,  $Q_{EW}$  is used by the source to decide which particular packets for each description to send. This decision is based on the importance of each packet to recovery of the description or on the time playout information and channel characteristics.

Let  $\rho = [\rho_1, \dots, \rho_N]$  be the set of transmitter powers, where  $\rho_1$  is the transmitter power at the source  $\rho_i$ ,  $i = 2, \dots, N$  is the transmitter power at the  $i$ -th relay. Then, we have the following resource allocation problem that maximizes the total reward  $w$ :

$$\max_{\pi, \rho, q_{s_i}, Q_{EW}} w = \max_{\pi, \rho, q_{s_i}, Q_{EW}} \sum_{\delta} P(\delta | \pi, \rho, q_{s_i}, Q_{EW}) w(\delta), \quad (1)$$

where  $P(\delta | \pi, \rho, q_{s_i}, Q_{EW})$  and  $w(\delta)$  are the probability of receiving  $\delta$  and the reward in that case, respectively, given that strategy  $(\pi, \rho, q_{s_i}, Q_{EW})$  is used. The above optimization is done under  $N$  channel constraints, namely  $\sum_i r_{ij} \leq R_j$ ,  $D$  source coding constraints:  $\sum_j r_{ij} \leq B/L$ , and  $N$  power constraints  $\rho_k \leq P_k$ , where  $P_k$  is the power constraint of the  $k$ -th transmitter.

Note that the optimization in (1) includes source optimization ( $q_{s_i}$ ), relay power allocation ( $\rho$ ), relay selection ( $\pi$ ), and scheduling/channel coding ( $Q_{EW}$ ). The above problem is a hard combinatorial problem that cannot be solved in real time. Thus, we introduce the following simplifications. First, assuming that the source block is pre-encoded, we fix all  $q_{s_i}$ 's, and consider  $\rho$ ,  $\pi$ ,  $Q_{EW}$  one at the time.

Power allocation  $\rho$  is done for each relay by looking only at the single relay-destination channel and selecting the lowest power level that will provide the average packet loss rate below a certain threshold. If that is not possible, then we set  $\rho_k = P_k$ . This

strategy is justified by the fact that the transmitter power is handled at the physical layer, in order to minimize the bit error rate for each link, and without any knowledge of the source.

Next, we proceed with finding optimal  $\pi$ , assuming that no scheduling is done, i.e., all encoded packets of a description are treated equally. Then description  $i$  is decoded if at least  $s_i \geq q_{s_i} B/L$  packets from this description are received. Note that for maximum separable codes,  $s_i = \lceil q_{s_i} B/L \rceil$ , with high probability, while for RLC codes  $s_i$  very slightly exceeds  $\lceil q_{s_i} B/L \rceil$  (the number of source packets to be sent). In the analysis we further assume that  $R_j$ 's and  $p_j$ 's are constant during the transmission of one encoded source block for all  $j = 1, \dots, N$ .

Even with fixed  $q_{s_i}$  and  $\rho$ , optimizing  $\pi$  is a difficult optimization problem. We simplify it by assuming that if the destination decodes description  $i$ , it gets reward  $w_i$ , where  $w_i$  is estimated based on the source recovery using only description  $i$ . The main motivation for this simplification is that it will reward descriptions that contribute highly to the reconstruction while significantly reducing optimization complexity. The total reward is the sum of the rewards for all decoded descriptions. Thus we optimize (1) restricting  $\delta$  to be of cardinality 1. With a slight change of notation, for clarity we rewrite (1) as:

$$\max_{\pi} w(\pi) = \max_{\pi} \sum_{i=1}^D P_i(\pi) w_i, \quad (2)$$

where  $P_i(\pi)$  and  $w_i$  are the probability that we recover description  $i$  if  $\pi$  is used and the reward in that case, respectively. Next we propose an algorithm that efficiently solves optimization problem in (2).

#### A. Global Optimum: Algorithm 1

Without loss of generality, we assume that all available resources are allocated. Therefore,  $\sum_{i=1}^D r_{ij} = R_j$ . Then, we have a resource allocation problem

$$\text{RA}(R_1, \dots, R_N, p_1, \dots, p_N; s_1, \dots, s_D, w_1, \dots, w_D)$$

whose optimal solution  $\pi^*$  is the one that maximizes (2).

*Brute force complexity.* For a channel with  $R_j$ , there are  $(R_j + 1)^D/D!$  ways of allocating the  $D$  descriptions. Therefore, the brute force complexity is  $\sim (\prod_{j=1}^N (R_j + 1)^D/D!)$ .



$1))^D \sim O(R^{ND})$ , where  $R$  is the geometric mean of  $R_j$ ,  $j = 1, \dots, N$ .

*Proposed dynamic programming solution.* Note that if  $\pi^*$  is the optimal solution of

$$\text{RA}(R_1, \dots, R_N, p_1, \dots, p_N; s_1, \dots, s_D, w_1, \dots, w_D),$$

then the sub-solution defined by sub-matrix  $\pi^*(1 : D - 1, :)$  has to be the optimal solution of  $\text{RA}(R_1 - r_{D,1}, \dots, R_N - r_{D,N}, p_1, \dots, p_N; s_1, \dots, s_{D-1}, w_1, \dots, w_{D-1})$ . We can take advantage of this fact to decrease the optimization complexity by dynamic programming.

In a nutshell, the algorithm will first seek the optimal solutions for all combinations of  $1 \leq R'_j \leq R_j$  and all  $1 \leq d \leq D$ . At  $d = 1$ , the problem is trivial as the optimal allocations for  $\text{RA}(R'_1, \dots, R'_N, p_1, \dots, p_N; s_1, w_1)$  is simply  $\pi^* = [R'_1, R'_2, \dots, R'_N]$  (i.e., allocate all resources to the first description).

Assume now that we have solutions for  $d'$ . For  $d = d' + 1$ , computing optimal solutions of  $\text{RA}(R'_1, \dots, R'_N, p_1, \dots, p_N; s_1, \dots, s_{d'+1}, w_1, \dots, w_{d'+1})$  will only involve scanning through the  $\prod_i R'_i$  solutions of the previous level. Therefore finding all optimal solutions for this level ( $d' + 1$ ), involves approximately  $(\prod_i (R'_i + 1))^2/2$  computations.

Note that, to compute the optimal solution of

$$\text{RA}(R_1, \dots, R_N, p_1, \dots, p_N; s_1, \dots, s_D, w_1, \dots, w_D)$$

will require going through all  $D$  levels and thus have complexity  $(D - 1)(\prod_j (R_j + 1))^2/2$  or approximately  $D(\prod_{j=1}^N (R_j + 1))^2 \sim O(DR^{2N})$  instead of  $O(R^{ND})$ , needed for the brute force method.

### B. Fast Solution: Algorithm 2

Algorithm 1 finds the optimal solution to the allocation problem in (2). However, its complexity and memory requirements might still be too large for real-time applications, especially when  $N$  or  $R_j$ 's are large. That is why, we also propose a simple and fast ad hoc strategy. It will be shown in Result Section that this strategy works well for the video MDC, where the descriptions do not vary largely in importance.

First, we arrange the channels in the decreasing order of quality, i.e., such that  $p_1 \leq p_2 \leq \dots \leq p_N$ . That is, Channel 1 (Ch1) is the best channel and so forth. Next,

we arrange the descriptions such that  $w_1 \geq w_2 \geq \dots \geq w_N$ . That is,  $D_1$  is the most important description, and so forth. Without loss of generality we assume that  $D \leq N$ . If  $D > N$  we could merge descriptions to end up with total of  $N$ .

Algorithm 2 proceeds as follows: Put  $D_i$  into  $\text{Ch}_i$  either until (1)  $\text{Ch}_i$  is filled; or (2)  $\frac{(1+\epsilon)s_i}{1-p_i}$  packets have been put, where  $\epsilon$  is heuristically set to 10%.<sup>1</sup> In the next step, channels  $1, \dots, D$  that have not been filled are filled equally with the descriptions for which condition (1) occurred, or if there are no such descriptions,  $\text{Ch}_i$  is filled with description  $i$ . Channels  $D + 1, \dots, N$  are filled equally with all descriptions.

The algorithm can be made blind of the source, by arranging descriptions in an arbitrary order. Note that, Algorithm 2 is applicable to any channel model by properly selecting  $\epsilon$ .

### C. AL-FEC and packet scheduling

AL-FEC is a flexible software-based solution that can effectively combat fading and easily be tied to source coding via source-channel coding algorithms (see [16] and references therein). Digital Fountain Raptor codes [7], as most efficient rateless packet loss codes, have been adopted for AL-FEC in DVB-H and MBMS [8]. The main problem of Raptor codes is the fact that they equally protect the entire stream and have a sharp avalanche effect. Thus, they are not very suitable to applications where importance of the sent packets significantly varies, which is the case in video coding.

Expanding window fountain (EWF) codes are a recent solution to this problem proposed in [31]. EWF is a class of unequal error protection (UEP) fountain codes based on the idea of creating a set of “nested windows” over the source block. The rateless encoding process is then adapted to use this windowing information while producing encoded packets.

The EWF concept is used in [15] to create EW-RLC by replacing LT/Raptor fountain codes by RLC. RLC applied over a source message produce encoded symbols as random linear combinations of source symbols with coefficients randomly selected from a given

<sup>1</sup>Note that  $\frac{s_i}{1-p_i}$  is the number of packets needed to be sent over a random loss channel with packet loss rate  $p_i$  so that  $s_i$  packets are in average received.  $\epsilon$  can be seen as a percentage of additional packets to be sent to ensure that at least  $s_i$  packets are received with high probability.

finite field. As a packet level AL-FEC solution, RLC are simple to implement and perform as near-optimal erasure codes for sufficiently large finite field used for creating linear combinations of source symbols (one-byte field  $GF(256)$  is usually sufficiently good [9]).

RLC codes provide similar performance as Raptor codes for point-to-point communications [16] at the expense of the increased decoding complexity. However, it is shown in [16] that if the used source block is small, RLC perform similarly to Raptor codes, and have acceptable decoding complexity via progressive Gaussian Elimination [10] [11]. The main advantage of RLC compared to Raptor codes comes in the multipath scenarios, as RLC effectively apply network coding concepts in the multi-node settings [9].

The general layout of a window structure of EW-RLC with three importance layers is shown in Fig. 2. The window with the most important subset of encoded data is the first window (W1) and the importance of data additionally included in windows progressively decreases as we proceed to the third window (W3). The subset data of W1 is contained in all the subsequent windows and is hence the best protected. Apart from W1, each window in addition to some of its own data also encloses the data of the higher importance windows. The size and structure of a window depends upon the elements meeting particular set criteria from a specific subset window. The number of windows is governed by the aggregation scheme employed to group encoded elements. The decoding of a window is the same as RLC decoding, in that, a window is recoverable if the receiver collects at least the same amount of linearly independent encoded symbols obtained from the window (or the windows contained in it) as there were in the window [15].

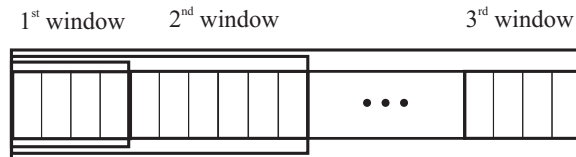


Fig. 2. EW with three windows.

The encoding process for EW-RLC starts by selecting a window from which the

RLC encoded symbol is to be generated. These probabilities of window selections are organized into a vector  $Q_{EW}$  and are equivalent to channel coding rates for each source layer/window. After a window is selected, the encoding is the standard RLC encoding performed over the source packets contained in that particular window only [15]. The authors have compared in [16] the performance of expanding window structure proposed here and a similar scheme termed non-overlapping window (NOW). In NOW scheme, the windows are non-overlapping and each window is protected with a separate AL-FEC code. The performance of EW-RLC schemes is concluded to be better than the corresponding NOW-RLC schemes, and this gain comes from the increased coding flexibility introduced by overlapping the windows.

In this paper we employ EW-RLC over  $l$  windows from consecutive source blocks that are of unequal importance, i.e., each description is organized into  $l$  layers of unequal importance, which form  $l$  windows of the EW-RLC scheme.

To provide optimal allocation of redundancy to different source priority classes, for each description  $i$  one can maximize the expected reward using analytically computed probabilities of decoding error performance. That is,

$$\max_{Q_{EW}^i} w_i(Q_{EW}^i) = \sum_{k=0}^l P_i(k|Q_{EW}^i) w_i(k), \quad (3)$$

where  $Q_{EW}^i$  is an  $l$ -length vector of window selection probabilities that determines the UEP allocation scheme,  $P_i(k|Q_{EW}^i)$  is the probability that layer  $k$  of description  $i$  will be the highest layer recovered if  $Q_{EW}^i$  is used,  $P_i(0)$  is the probability that nothing is recovered,  $w_i(k)$  is the reward if all layers up to and including layer  $k$  are recovered,  $k = 1, \dots, l$ . Analytical expressions for probabilities that a layer is recovered assuming a random channel loss model can be found in [15].

## V. SIMULATION SETUP

In this section, we describe the specific source-channel-network scheme with the parameters used.

### A. Video Coding and AL-FEC

H.264/AVC [12] has emerged as a video coding standard in wireless mobile systems, due to excellent rate-distortion performance at low-complexity decoding. H.264/AVC also provides many error-resilience features to mitigate the effect of lost packets during transmission. Error resilience via slicing splits the frame into multiple slices that are independently encoded and separated by resynchronization points.

Since slices can be made independently decodable, the partitioning of a frame into slices can be used to create multiple descriptions with fine granularity. For example,  $D = 2$  descriptions, namely, MDC1 and MDC2, can be created by assigning three slices from each frame to each description. In this way each description has a significant representation of every frame which helps the decoding process. The way the slices are chosen for inclusion in a description is shown in Fig. 3. Firstly, both descriptions have the intra-coded Instantaneous decoder refresh (IDR) [12] as the first entry. Thereafter, starting from the first following frame, alternate slices are copied to each description. This way, the overlap in the source content between two descriptions is minimal.

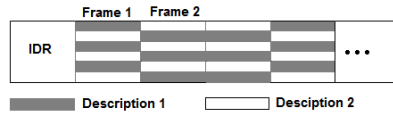


Fig. 3. Generation of two descriptions.

To achieve scalability, each description is encoded using data partitioning (DP), which enables partitioning of each slice into up to three partitions (denoted as A, B, and C) based on the importance of the encoded video syntax elements to video reconstruction. Data partition A (DP A) contains the most important data comprising slice header, quantization parameters, and motion vectors. DP B contains the intra-coded macroblocks (MB) residual data, and DP C contains inter-coded MB residual data. The decoding of DP B is made independent of DP C, using Constrained Intra Prediction (CIP) parameter in the H.264/AVC encoder. The effect of error propagation is limited by inserting periodic macroblock intra updates (MBIU) [12]. This way, each description can be coded using up to three quality layers.

TABLE I  
 COMPRESSION RESULTS (SIZE IN BYTES AND PSNR IN dB) FOR THE FIRST GOP OF THE  
 FOREMAN SEQUENCE.

Layer	MDC1			MDC2		
	Size	PSNR	Pkts	Size (bytes)	PSNR	Pkts
BL	27,960	25.76	39	31,871	25.58	44
EL	23,980	26.79	33	23,539	26.2	33
Total	51,940	26.79	72	55,410	26.2	72

After video encoding, two independently decodable descriptions of possibly different sizes are generated. Each description contains  $l = 2$  quality layers obtained by grouping IDR + DP A + DP B into one layer (i.e., the base layer (BL)) and DP C into the second quality enhancement layer (EL). For this case, Table I shows the peak signal-to-noise ratio (PSNR) and the rate (in bytes) for the first Group of Pictures (GOP) of size 16 frames for the standard CIF Foreman video sequence. The video is encoded at 25 frames per sec and the total compression rate is 1.2 Mbps.

The best video quality of 41.07dB is achieved when both MDCs are received completely. However, it can be seen from Table I that due to the way the descriptions have been designed, the video can be decoded with the BL of any one description only. This layering of video data within a description makes this scheme adaptive and practical for varying channel conditions. In the case where both descriptions are lost entirely and nothing is decodable for the GOP then the decoder applies a simple error concealment technique by replacing the lost GOP by the last frame of the previously successfully decoded GOP. Each description is independently protected against packet losses by an EW-RLC. The BL forms the first window – W1, and the (BL+EL) forms the second window – W2.

### B. Simulation Parameters

The BS compresses the video, performs RLC protection, relay selection based on the feedback from the relay regarding the quality of the relay-destination link and the relay transmit power, and schedules encoded packets for transmission.

We consider the setup with one, two, and three relays and a direct link from the BS to the destination, thus  $N = 1, 2$ , or  $3$ . To model the fading channel with multiple femto-relays, we use the FSMC channel model that reflects the change in the bit error rate depending on Markov chain state-space resulting in different received data rates. In the model each wireless Rayleigh block-fading channel is represented by a packet-level FSMC channel model [13] that adapts with the mobility of the user and transmit power.

We use the symbol size of 734 bytes for RLC. We assume a header overhead of 60 bytes per packet to cater for the headers added at the various protocol layers, e.g., RTP/UDP/IP (note however that the robust header compression can significantly reduce the size of the header). We always put one symbol in the packet, thus transmission packets are of size  $734+60=794$  bytes. Gaussian Elimination decoding is performed on source block sizes of 70-80. These low block sizes enable fast decoding on a smart-phone without sacrificing performance [11]. We assess reconstructed quality using PSNR and perform optimization (Algorithms 1 and 2) with the reconstructed frame-average PSNR as reward  $w$ .

### *C. Simulation Results and Discussion*

In this section we present our simulation results for the CIF Foreman sequence. Similar results are obtained for the CIF Container.

First, we look at the system with  $N = 2$  relays shown in Fig. 1 with pico-relay being a BS. The distances between the destination and BS, femto relay 1 and femto relay 2 are 70m, 25m, and 20m, respectively [14]. The data transmission rate is 1000 kbps, 600 kbps and 550 kbps for the BS, Femto 1 and Femto 2 relays, respectively. This corresponds to the maximum number of encoded packets that these paths can take during one source block (one encoded GOP) of 100, 60, and 55, respectively.

The simulations have been done for two different power constraints of the two femto relays: in Configuration 1 (C1) we set the required average packet loss rates in the femto relay 1 and 2 links to 0.25 and 0.3, respectively. This results in physical transmitter powers of 30dBm and 35dBm, respectively, which translates into the averaged received signal-to-noise ratio (SNR) of 23dB and 33dB, respectively. In Configuration 2 (C2)

we set the required average packet loss rates in the femto relay 1 and 2 links to be 0.05 and 0.1, respectively. This results in the physical transmitter power of 47dBm, for both relays, which translates into the average received SNR of 40dB and 44dB, respectively.

For the above femto-relay transmit power constraints, we vary the BS power level in the range of 40 to 70dBm, which results in the average received SNR between 12dB and 43dB. Then, the average packet loss rate in the direct link varies from 0.05 to 0.3.

We select several UEP EW-RLC schemes, obtained by changing  $Q_{EW}$  to test their robustness to varying channel conditions. Note that we always use the same  $Q_{EW}$  for the two descriptions.

The results for the mobile speed of 3km/hr, comparing Algorithms 1 and 2 are shown in Figure 4. The term  $EW(x, y, z)$  refers to a set of selection probabilities for windows W1 (BL), such that the selection probability of the BL for both descriptions for the direct link, link via Relay 1 and link via Relay 2 are  $x$ ,  $y$ , and  $z$ , respectively. It can be seen from the figure that the two algorithms show similar performance. Indeed, Algorithm 1 is for only 0.5-1dB better. Note that, in the  $EW(1.0,1.0,1.0)$  scheme only window W1 is selected, hence the PSNR cannot go beyond that achievable with the two BLs.

Setting the probability of selection of the BL to 0.8 ( $Q_{EW} = (0.8, 0.2)$ ) provides the best results for all source power levels. Note that in this case, 80% of the rate is allocated to the BL, but still even at the loss rate as high as 30%, ELs are often recovered showing a PSNR gain of 4dB compared to the case when only BL is sent alone. For Configuration 2, setting the probability of selection of the BL to 0.5 or 0.8 provides the best results. Algorithm 1 provides a negligible performance gain over Algorithm 2 only at the highest packet loss rates.

The results for the random loss channel model are shown in Figure 5. Note that, we use the same average channel loss rates as those given by the FSMC model. As expected, the performance of Algorithm 1 for Configuration 1 is consistently better than that of Algorithm 2. However, the difference is very small (under 1dB). The  $EW(1.0,1.0,1.0)$  scheme again performs poorly, since it does not exploit the available bandwidth to the fullest. Indeed, a gain of 7-10dB can be observed by sending ELs.



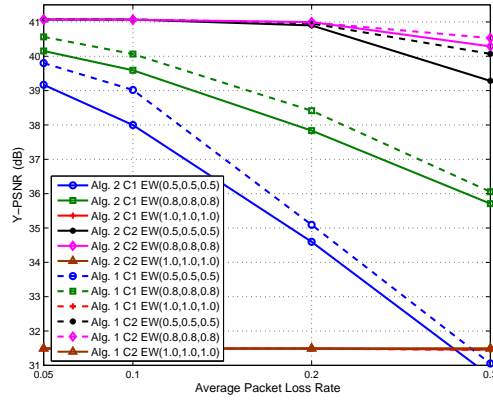


Fig. 4. Frame average Y-PSNR vs. average packet loss rate for Configurations (1) and (2) for the FSMC channel.

The results for the random channel are for up to 0.5dB better than those for the FSMC channel, which demonstrates that the proposed schemes are robust to the bursty nature of the FSMC channel. For Configuration 2, where the loss rates over the relay channels are much lower, both algorithms perform equally well.

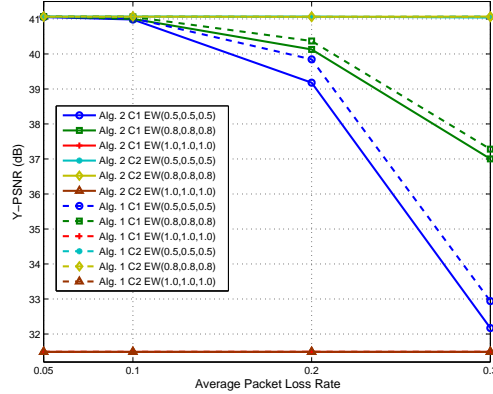


Fig. 5. Frame average Y-PSNR vs. average packet loss rate for Configurations (1) and (2) for the random loss channel model.

In the next two figures we compare results of Algorithm 2 to the benchmark schemes – an adaptive source-optimized single description no relay (SDNR) schemes that transmit a single-layer AVC (without MDC) without relays. For each testing channel condition we find the optimal source rate (via exhaustive search over all video source rates

between 500kbps and 1.5Mbps where the lower and upper video source rates were found to provide too much and too little protection, respectively, and were never selected); thus this scheme adaptively changes the source rate to match the channel conditions. It is assumed that the perfect knowledge of the channel is available prior to source coding, which is not the case for the proposed schemes.

Figure 6 shows results for the FSMC channel and both configurations and four different scheduling strategies. It can be seen that Algorithm 2 outperforms the benchmark schemes if the packet scheduling is properly done. Note that the benchmark schemes do not use MDC, and hence provide better performance in the error-free scenarios. Using relaying, power is effectively distributed among three nodes providing increased performance compared to the SDNR schemes. It can also be observed that by increasing the power levels of the two relays for 9dB in total, a gain of roughly 5dB at the highest channel loss rate and 0.5dB for the lowest channel loss rate is obtained.

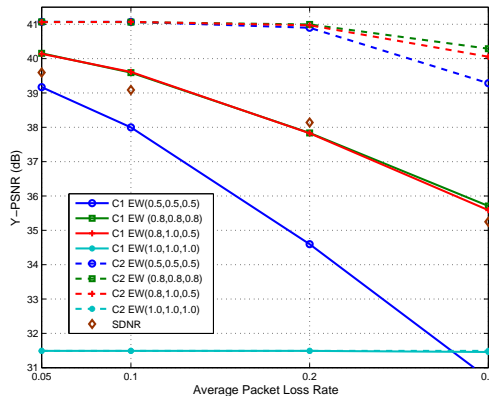


Fig. 6. Frame average Y-PSNR vs. average packet loss rate for both configurations for the proposed Algorithm 2 and the single description no relay schemes for the FSMC channel.

The results for the random loss channel model are shown in Figure 7. The optimized relay-assisted scheme performs for almost 2dB better than the no-relay scheme at the lowest packet loss rate and almost 5dB better for the highest packet loss rate. Results for Configuration 2 are for up to 5dB better than those for Configuration 1. Note that for both types of channels, Configuration 1 scheme at 0.3 packet loss rate shows roughly the same performance as the source-optimized benchmark scheme, because in

this case, there is only a little use of relaying due to high loss rates at the two relay links.

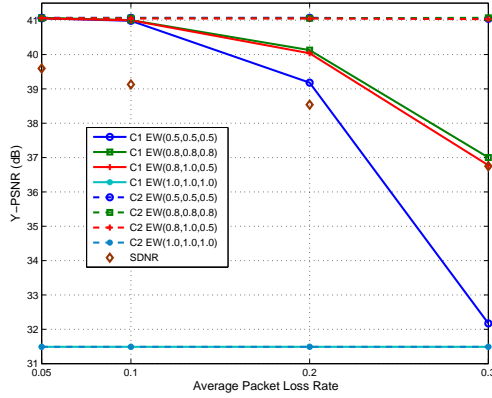


Fig. 7. Frame average Y-PSNR vs. average packet loss rate for both configurations for the proposed Algorithm 2 and the single description no relay schemes for the random loss channel model.

From these results, it is obvious that the relay-assisted schemes are very robust to the change in the channel conditions of the direct and relay links. Suboptimal and fast Algorithm 2 shows close performance to Algorithm 1. It is important to send both BL and EL for both descriptions to obtain high performance. Scheduling packets at the source (varying  $Q_{EW}$ ) is important at the high packet loss rates and can provide a gain of over 10dB.

Figure 8 shows results for the FSMC channel and the mobile speed of 30km/hr. Comparing Figures 6 and 8, one can see that the proposed solutions are very robust to the increase in mobile speed.

Next, we test the proposed scheme using two different network topologies. We use two benchmark schemes: the SDNR, as previously used, and the second one is the single-description relay (SDR) scheme, that transmits a single-layer coded stream over the relays. Note that the two benchmark schemes have a performance gain compared to the proposed scheme in the error-free environment.

Figure 9 shows the results for the network with three relays and a direct link between the source and the destination. The data transmission rates are 600kbps over the direct link and 300kbps, 300kbps, and 200kbps over the three relays. Thus, the direct link

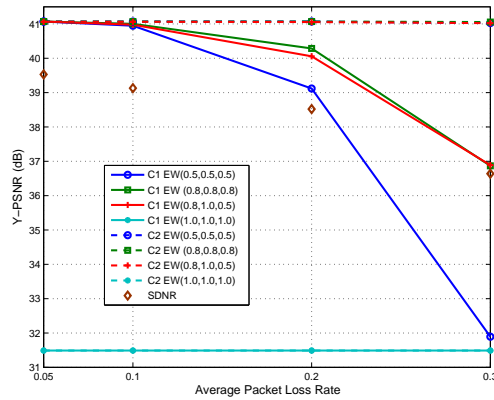


Fig. 8. Frame average Y-PSNR vs. average packet loss rate for both configurations for the proposed Algorithm 2 and the single layer schemes for the FSMC channel for the mobile speed of 30 km/hr.

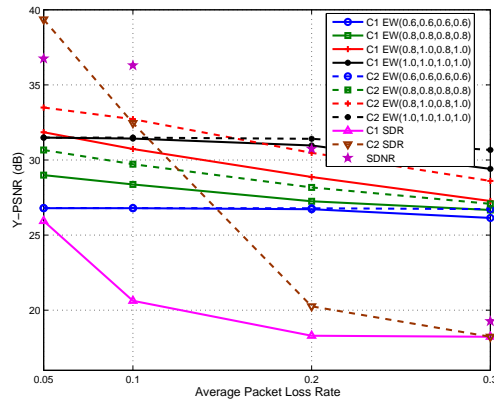


Fig. 9. Frame average Y-PSNR vs. average packet loss rate for both configurations for the proposed Algorithm 2 and the benchmark schemes for the three-relay network.

has a significantly higher bandwidth than the relay links. This corresponds to the maximum number of encoded packets that these paths can take during the duration of one GOP of 70, 30, 30, and 20, respectively. Configurations 1 and 2 are the same as before, with the packet loss rate over the added relay path of 0.25 and 0.15 for Configuration 1 and 2, respectively. We vary the packet loss rate on the direct link and plot the average received PSNR.

One can see from Figure 9 that at a very low packet loss rate (0.05), the benchmark schemes outperform the proposed scheme due to better error-free performance of these schemes. However, as the packet loss rate increases the proposed schemes become better

achieving a maximum gain of up to 12-14 dB compared to the benchmark schemes. For C1, SDR always performs poorly because the relay links are severely corrupted, and hence rarely the entire bitstream can be decoded correctly. This does not affect significantly the proposed scheme since one description is always decoded successfully. The SDNR scheme is powerful at low packet loss rates since it uses only the direct link, but very quickly its performance decreases as the packet loss rate at the direct link worsens. One can see that the proposed schemes are robust to the variation in packet loss rate characteristics of the direct link as well as the relay links (since the difference between C1 and C2 curves is small).

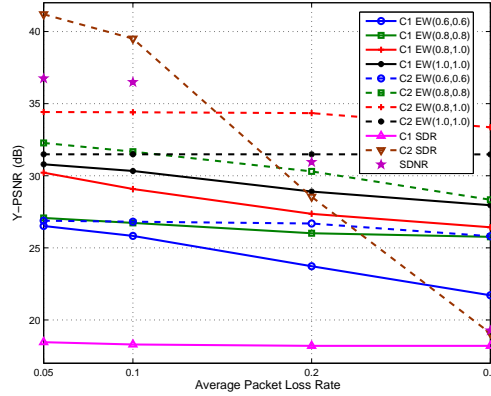


Fig. 10. Frame average Y-PSNR vs. average packet loss rate for both configurations for the proposed Algorithm 2 and the benchmark schemes for the one-relay network.

In the next example, we use a single-relay network, with data transmission rates of 700kbps and 800kbps, for the direct and relay link, respectively. Thus in this case, the relay link has a higher bandwidth, and we set a high packet loss rate over the relay link of 0.3 for Configuration 1 and a lower loss rate of 0.05 for Configuration 2. Figure 10 shows the obtained results. One can see that for C1, the SDR scheme performs poorly since it cannot transmit the entire bitstream successfully due to a high loss rate at the relay. The SDNR scheme, on the other hand, has the best performance at very low packet loss rates (it does not use the relay, so it is not affected by the bad relay link), but very quickly loses its advantage with the increase of the packet loss rate at the direct link. For C2, the SDR scheme is the best at the start due to the low packet

loss rates at the relay and direct link, but at the highest packet loss rate the difference between the proposed MDC scheme and SDR is over 15 dB.

From the conducted simulations, it is obvious that the proposed MDC-based scheme provides significant gains compared to the non-MDC schemes in the case of severely corrupted links. On the other hand, when at least one relay link is clean or almost clean and with enough bandwidth, the non-MDC schemes perform better, as expected. The reasons for this are: (1) The MDC schemes introduce a small rate penalty compared to the optimal single-description coding scheme. (2) Single-description schemes are “all-or-nothing” schemes, which, if all source packets are recovered, provide the highest fidelity, and otherwise nothing can be recovered. On the other hand, the MDC scheme provides the basic reconstruction if only one description is received, which often can be done even if the channel conditions are bad. (3) The RLC scheme is very suitable for relaying as it enables the receiver to simply wait for enough packets to arrive regardless of which links the packets are sent over. This way, even if one relay (or direct) link fails, the packets can be routed via the other links without any alternation in source or channel coding. All schemes, (MDC and non-MDC) take advantage of this property.

## VI. CONCLUSION

We considered real-time transmission of layered MDC video over relay-assisted paths. We designed an MDC scheme using slicing and data partitioning features of H.264/AVC and fed the resulting packets into the EW-RLC encoder for erasure protection. The encoded packets were streamed over direct link and over two relay-assisted channels. A resource optimization framework was developed, using dynamic programming and the proposed fast algorithm, that optimally and sub-optimally select relays and schedule packets for transmission.

Future work could explore a quality-of-service and quality-of-experience driven solutions, where it is necessary to guarantee certain level of quality at the user side.

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