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EFFICIENT JPEG2000 IMAGE TRANSMISSION USING RCPT CODES BASED ON CHANNEL OPERATING REGIONS

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The latest wavelet-based image compression standard, JPEG2000, provides five different progressive modes. Due to the embedded structure used in the codestream syntax, the received bits after transmitting over noisy channels exhibit severe error sensitivity and experience error propagation. In this paper, we propose an efficient JPEG2000 transmission system with separate design methods for image header and data packets. First, two hybrid FEC/ARQ schemes operating over an AWGN channel to realize near error-free transmission of the image header are studied. Second, compressed data packets for different quality layers are protected using an unequal error protection (UEP) method. A set of rate compatible punctured turbo codes (RCPT) is used for forward error correction. In addition to the rate-distortion based rate allocation models, a practical channel segmentation method, which avoids the complicated optimization computation, is presented for code rate selection based on channel operating regions. Using simulation methods, it has been found that the choice of code rates is best guided by segmentation of the SNR into operating regions of the AWGN channel. Experimental results show that the proposed system outperforms systems using equal error protection (EEP) up to 9 dB while significantly reducing the computation complexity compared to other UEP systems. One major contribution of the system developed in this paper is the separate treatments of the compressed data in the JPEG2000 codestream. Another contribution is that of a rate selection method for practical channel codes. The resulting system design approach yields improved overall image transmission quality with minimal bandwidth expansion.

Keywords: Joint source-channel coding, JPEG2000, Rate compatible punctured Turbo codes (RCPT), Unequal error protection (UEP)
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1 Introduction

Progressive image coding is a useful and efficient method in many practical applications such as web browsing and communications among wireless devices. The latest JPEG2000 image compression standard [1] includes five different progressive modes which enable increasing quality, resolution, spatial, and/or color component as more bits are decoded sequentially from the beginning of a compressed codestream. Embedded codestream structures, which permit flexibility and the capability to progressively reconstruct the image, always exhibit severe error sensitivity and experience error propagation after transmission over a noisy channel due to the inherent bit dependence within the codestream. Approaches using forward error correction (FEC) have been widely employed to combat the degradation of reconstructed image quality. In [2], an equal error protection (EEP) scheme was used with rate compatible punctured convolutional (RCPC) codes for images compressed by a set partitioning in hierarchical trees (SPIHT) coder [3]. This method was extended by using unequal error protection (UEP) in [4] which showed a great improvement compared to the EEP scheme. Corresponding methods on varying channels were studied in [5]. Later, the notion of using UEP had been introduced for JPEG2000 codestreams employing RCPC codes [6], [7]. Because of the channel capacity approaching performance of turbo codes, they have received much attention for use in image transmission applications. JPEG2000 image transmission systems protected by rate compatible punctured turbo (RCPT) codes have shown quality improvement in the reconstructed images both objectively and subjectively [8]-[10]. Most of these systems use joint source-channel coding methods, and among them, a powerful channel code rate selection algorithm is especially needed. Due to the special properties of the embedded progressive image encoder, one can easily, and directly, obtain the rate-distortion information at the encoder side. Thus, different types of rate allocation algorithms based on the rate-distortion method were established. Other general algorithms for rate allocation based on the rate-distortion method were discussed in [11]-[13]. Lagrange multiplier [14] and dynamic programming [15] approaches and their variations are the typical tools used for solving these optimal rate allocation problems.

In this paper, we propose an efficient JPEG2000 codestream transmission system for use on AWGN channels. Fig. 1 illustrates the block diagram of the proposed system. The output of the JPEG2000 source encoder is a quality progressive codestream that includes certain quality layers. The codestream is divided into two parts. One is the header part which contains the codestream header, tile headers, and packet headers while the other part includes all the data packets that belong to different quality layers. Transmission of these two parts are treated separately due to the extremely high level of protection needed for the header part. In order to get near error-free performance, hybrid FEC/ARQ schemes are designed. A set of punctured Turbo codes is used for FEC. In addition to FEC, two ARQ mechanisms, stop-and-wait (SW-ARQ) and incremental redundancy, are employed to further increase the error-free probability . All the data packets are protected by a set of RCPT codes which are derived from the punctured turbo code set. Channel code rates are selected by a central control unit. We establish both the theoretical optimization models and the practical code rate selection methods which avoid the need for a complicated code rate selection algorithm. Channel and source decoding are accomplished by inverting the encoding operations. Errors in the header after decoding will trigger a retransmission request. Decoding of subsequent data packets will start only after the header has been successfully decoded. Details of each component will be discussed in the remainder of the paper.

The organization of the paper is as follows. In this section, we shall describe the embedded codestream structure and the set of RCPT codes which will be used later. Section II presents two hybrid FEC/ARQ schemes designed for the image header transmission and provides a framework for establishing the rate-distortion method used for the code rate selection for all the image data packets. Section III discusses some practical considerations for code rate selection and introduces the channel segmentation method based on the notion of channel operating regions. Section IV illustrates the simulation results for both the header transmission and the data packets transmission, as well as demonstrates the effectiveness of the practical code rate selection method. Finally, Section V provides some concluding remarks.

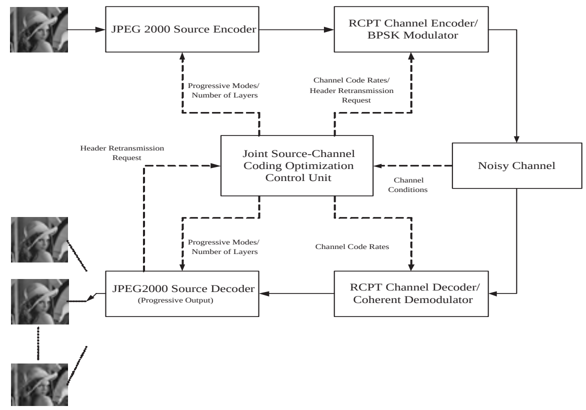


Fig. 1. Block Diagram of the Proposed Image Transmission System.

1.1 Codestream Structure

In our research, the JPEG2000 Part 1 standard is used as the source coding scheme. Details of the final codestream syntax are found in [16]. Fig. 2 illustrates the general structure of a JPEG2000 codestream. It has a hierarchical structure. Each codestream has a codestream header followed by all the tile-streams. Each tile has its own tile header followed by all the packets that belong to the tile. Different kinds of progressive modes can be accomplished by different packing orders for packets which appear in a tile-stream. Each packet has its own header and data section. A packet has information about one quality layer of one spatial location of one resolution of one component. For example, for an 8-bit grayscale image, if the entire image is taken as one tile, perform a DWT successively 5 times to get a total of 6 resolutions, and employ 3 quality layers during the compression process, we will have totally $3 \times 6 = 18$ packets in the final codestream. Within each packet, data are encapsulated in a deterministic order according to subband (HL,LH, then HH) within the corresponding resolution represented by the packet. Each subband is divided into several codeblocks which are the smallest units used in the entropy coding stage. In Fig. 2, we assume there are four codeblocks in one subband. Further, in each codeblock, there are three coding passes generated for each bit-plane after employing the context adaptive arithmetic coding. In Fig. 2, the third codeblock in LH subband of the second resolution contains of 15 coding passes which are distributed among the three quality layers. An error resilience mechanism referred to as 'packed packet header' provides the user with an optional packing scheme, that is to store all the packet headers within the codestream or tile header. Our results show that by providing strong protection for the packet headers included in the codestream or tile header, the misinterpretation of the packet header contents by the decoder is greatly reduced. In this paper we use one-tile compression and the 'packed packet header' mechanism. Hence, for convenience, we will refer to the overall header containing the codestream, tile, and packet headers as simply a header for the remainder of this paper.

1.2 Rate Compatible Punctured Turbo Codes

When transmitting a compressed image frame, use of an UEP method is employed since different parts of a frame have different levels of importance. RCPT codes are quite suitable

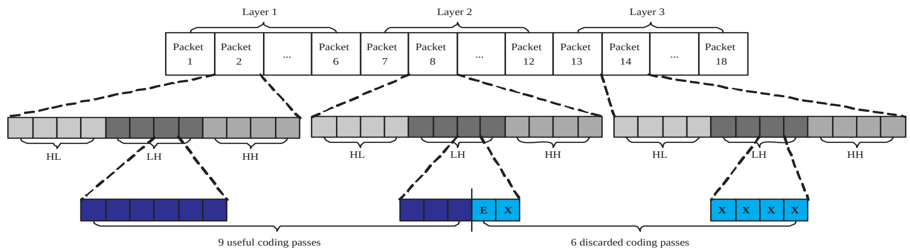


Fig. 2. General JPEG2000 Codestream Structure.

as UEP channel codes and have been quickly adopted for use in different image compression standards. We select the set of RCPT codes to be those introduced in [17] as the UEP codes to be used in transmitting both the header and data packets. The punctured rate set is defined by

$$r_k = \frac{k}{k+1} \qquad \text{for } k = 1, 2, 3, \ldots \tag{1}$$

together with another element, r_0, which is the rate of the mother code taken to be $1/3$. Thus, the code rate set is $\{1/3, 1/2, 2/3, 3/4, 4/5, 5/6, 6/7...\}$, and we denote it as code rate set I. Further, by letting $k = 2^a$, for $a = 0, 1, 2, \ldots$, a new subset of rates, code rate set II, is formed as $\{1/3, 1/2, 2/3, 4/5, 8/9, 16/17...\}$. It is quite easy to generate such rate $k/(k+1)$ codes with a puncturing pattern defined by $P(p, q)$ which saves the pth bit from every $2k$-bit parity block in the first constituent encoder, and the qth bit from every $2k$-bit parity block in the second constituent encoder. In this set, a higher rate code is embedded in a lower rate code, and thus meets the rate compatibility definition due to the special puncturing pattern, $P(p, q)$.

The asymptotic bit error probability for maximum-likelihood decoding of Turbo codes transmitted over an AWGN channel is given by

$$P_b \simeq \max_{w \geq 2} \left\{ \frac{w n_w}{K} Q \left(\sqrt{\frac{2 r d^{TC}_{min,w} E_b}{N_0}} \right) \right\} \tag{2}$$

where, $d^{TC}_{min,w}$ is the minimum weight of codewords for weight-w inputs, and n_w is the number of weight-w inputs resulting in a weight $d^{TC}_{min,w}$ codeword. Here E_b/N_0 is the bit energy to one-sided noise power spectral density ratio and r is the code rate. Finally, K is the block length of the information bits as well as the length of the corresponding interleaver. We have $K = rN$, where N is the length of the corresponding codeword. From (2), we can find that the error rate can be reduced simply by increasing the length K although this increases the bandwidth requirement. This is the basic trade-off when selecting a code rate. From (2), it is not easy to calculate the relationship between N and P_b. There are several simple models that have been used in RCPC codes, we borrow the typical log-affine model described in [13] for use in our scheme, that is

$$\log P = \alpha N + \beta \tag{3}$$

where, α and β are parameters acquired through off-line simulations and can be accessed by a codec using a lookup table in practice. The error probability P can be used to represent both the probability of bit error and frame error which will be shown later.

2 Separate Treatments of Image Header and Data Packets

Like other image/video compression standards, the JPEG2000 standard packs many important information items into the header section. Errors, even one bit error, in the header will cause the decoder to lose some basic information and/or synchronization, and subsequently fail to reconstruct the image. Thus, error-free transmission is necessarily required. In the existing systems, header transmission is either assumed error-free [2], [8], or treated the same as the data packets wherein it is easy to cause a reasonably large percentage of decoding failures [6]-[7], [9]. By requiring error-free performance two steps are proposed in this section. First, header should be treated as the most important section within the compressed codestream, thus it should receive highest protection compared to other sections. In order to further increase the error-free transmission probability, ARQ methods are necessary. It is quite nature to combine FEC and ARQ so as to maximize the throughput of the overall system. Two hybrid FEC/ARQ methods are introduced in our system. We attempt to transmit header information separately from data packets with a fixed constraint on the number of retransmissions and then maximize the throughput when selecting the code rates. During each transmission, a time consuming iterative Turbo decoding process is needed. Hence, placing a constraint on the number of retransmissions is an application-dependent consideration which realizes the near error-free performance. Two hybrid schemes are presented in the next two subsections. The rate-distortion methodology for the transmission of data packets is presented later.

2.1 Hybrid FEC/ARQ Scheme I

We assume the feedback channel is noise-free. An important point is that we should choose a "best" rate r_i from the previously defined code rate set I for a certain E_b/N_0. Here "best" means the rate that meets the transmission constraint while maximizing the throughput. The BCJR-based maximum *a posteriori* (MAP) iterative Turbo decoding process is the most time-consuming process. It was shown that the computational complexity is proportional to $2^{(N-K)}$ where $(N-K)$ is the number of parity bits in an N-bit codeword containing K information bits. Thus, by ignoring the transmission delay and using the SW-ARQ method for simplicity at both the receiver and the transmitter where no additional buffers are needed, the selection of code rate for a certain E_b/N_0 is given by

$$\max_i \eta_{SW} = \frac{K}{M(N_i + D\gamma 2^{(N_i-K)})} \tag{4}$$

with the constraint

$$M \leq M_c \tag{5}$$

where, i is the index into the code rate set I. The parameter M has the value that is either the expected number of transmissions which meets the first error-free header transmission, or the value M_c which is the pre-determined maximum number of transmissions. Also, D is the bit transmission rate in bits/second, γ is a constant that translates (together with D) the iterative decoding time, $2^{(N_i-K)}$, into the number of bits that could have been transmitted during the idle transmission time, and N_i is the length of the codeword associated with the code rate $r_i = K/N_i$. Let p_f denote the probability of the event 'bad header transmission' (BHT). If the channel decoded header contains errors and the source decoder cannot use it to reconstruct the image, we say that a BHT had occurred and the physical channel appears to be a packet erasure channel with either a fixed or varying packet loss rate. Usually, one bit error will cause a BHT, so the value of p_f is very close to, but lower than, the frame error rate (FER) of the associated channel codes. The relationship between p_f and N_i is set by the log-affine model defined in (3). Finally, the average number of retransmissions in terms of the probability of BHT is $M = 1/(1-p_f)$. Substituting M and (3) into the constraint problem in (4) and (5) establishes the relationship needed for selecting the code rate to be employed.

2.2 Hybrid FEC/ARQ Scheme II

When considering the class of rate compatible punctured codes, such as the code rate set II defined previously, it is natural to use an incremental redundancy method when combining the FEC and ARQ. The procedure employed is as follows. The encoder first selects a higher rate code, if the decoder sends out an NACK through the feedback channel which means a BHT had occurred, the encoder will only send additional parity bits (which were punctured in the previous codeword). This procedure of sending incremental parity bits can be used until either the lowest code rate, or the mother code rate itself is used. The throughput model for a certain E_b/N_0 in this scheme is given by

$$\max_{\Pi(i,M')} \eta_{IR} = \frac{K}{N_s} \ s.t.\ M \leq M_c \tag{6}$$

where, i is the index into the code rate set II and M is the number of compatible rates selected. Hence, $\Pi(i, M')$ is a subset of code rate set II which contains rates $\{r_i, r_{i+1}, \cdots, r_{i+M'-1}\}$. In practice, it is reasonable to set $M_c = M'$. The number N_s is the expected total length transmitted for one header with a certain number of retransmissions until we get the first useful header at the receiver, it is given by

$$N_s = (1-p_{f1})L(1) + \sum_{n=2}^{M} [p_{f1}p_{f2}\cdots p_{f(n-1)}(1-p_{fn})L(n)] \tag{7}$$

where, $L(n)$ is the total bit-length of n transmissions required for error-free transmission or near error-free transmission when $n = M_c$. Here $L(n)$ is defined as

$$L(n) = \frac{K}{r_n} + \sum_{i=1}^{n} \left(D\gamma 2^{(\frac{K}{R_i} - K)} \right) \tag{8}$$

and $p_{f1}, p_{f2}, \ldots, p_{f(M)}$ in (7) are the BHT probabilities for different rates, where index 1 corresponds to the highest rate and index M corresponds to the lowest rate. Note that the term corresponding to $n = 1$ is written separately in the right side of (7) so as to simplify the structure of the second term. Upon substituting (7) and (8) into (6), the rate selection model for this hybrid scheme is established. Practical considerations for both Schemes I and II will be discussed later.

2.3 UEP Method For Data Packets

In this section, we present a rate-distortion based quality progressive transmission scheme to be used for all of the data packets. The embedded packaging of information in a codestream causes strong dependence among the consecutive bits within the data packets. Error protection for these packets should be applied judiciously to improve the quality of the reconstructed image while incurring only minimal bandwidth expansion. The error resilience methods, ERTERM and RESTART, used in the JPEG2000 codec [16] provide a stop mechanism for the decoder. That means when an error has occurred in a coding pass, the decoder will discard this and all the future coding passes in the current codeblock. Fig. 2 shows that there is a bit error in the 10th coding pass of a codeblock. The JPEG2000 decoder will use the first 9 error-free coding passes and discard the last 6 coding passes. Since the arithmetic encoder is reset for each coding pass due to the selected error resilience method, such a stop mechanism can effectively prevent error propagation. In practice, the stop mechanism could be used for different granularities, such as a subband, a codeblock, or a coding pass. Let's consider any one of the granularities mentioned above and call it a block. The rate-distortion information obtained from the post-compression rate-distortion optimization procedure (PCRD-opt [16]) during the source coding process helps to establish the selection of channel rates for different blocks. In the proposed system, these rate-distortion data are included in the header section. Assume that the receiver successfully decodes m blocks and there exists at least one bit error in the $(m+1)$th block. The corresponding decoded image will have a remaining distortion denoted by d_m. For the ith block, let the probability that at least one bit error is contained in it be denoted as P_i. The log-affine model in (3) is used again for the relationship between P_i and N_i which is the length of the channel block. Suppose the fixed length of an information block is L, then we have $r_i = L/N_i$, which is the channel code rate for the ith block. The probability of error-free decoding of the first m blocks (first decoding error at block $m+1$) is

$$P(m) = \begin{cases} P_1 & \text{m=0} \\ P_{m+1}\prod_{i=1}^{m}(1-P_i) & \text{m=1,2,\ldots,M-1} \\ \prod_{i=1}^{M}(1-P_i) & \text{m=M} \end{cases} \tag{9}$$

where, M is the total number of blocks transmitted. The two special situations, when the first transmitted block contains errors ($m=0$), and when all the transmitted blocks have no error ($m=M$), are separately identified in (9) for special treatment. Having the rate-distortion information, our goal is to minimize the average distortion subject to a certain rate constraint. The problem can be expressed as

$$\min_{\Pi(i,M)} \overline{D_F} = d_0 P_1 + \sum_{m=1}^{M-1} d_m P_{m+1} \prod_{i=1}^{m}(1-P_i) + d_M \prod_{i=1}^{M}(1-P_i) \tag{10}$$

subject to

$$\sum_{m=1}^{M} N_m = N_T - N_H \tag{11}$$

where N_H is the length of channel protected image header, and N_T is the total length constraint for sending the entire image. Obviously, the image header has the highest priority for consuming the total length of the entire image. The remaining length $N_T - N_H$ is used for all the data packets which should be divided between the source and channel encoders. We choose to use code rate set II for assignment in this bit allocation problem. Two major parameters need to be considered in the policy, they are the rate set r_i and the total number of blocks M.

3 Practical Rate Selection Consideration

The optimum rate allocation problem described in (4), (6), (10), and their variations can be solved in different ways. The typical methods are Lagrange multiplier and dynamic programming. They are widely used in solving similar rate allocation problems found in [4]-[9] and [11]-[14]. Generally, the solution procedures are quite complicated. There are typically constraints on the selection of channel rates that are specific to real-world applications. For example, we only have a finite set of discrete rates and usually, for blocks $(L_1, L_2, \ldots, L_M)$, the rate set $\{r_0, r_1, \ldots, r_N\}$ with $M \geq N$ is always selected as $r_i \geq r_j$, when $i \geq j$ assuming that block L_j contains more important information than block L_i. So the analytical solutions must conform to meet these practical conditions. Given these constraints, we can only achieve near-optimum performance. It has been shown in [18] that for RCPC codes, a 3-rate set is enough for a BSC channel with a crossover error probability $\rho \leq 0.1$. When the number of available code rates and the number of quality layers are small, different algorithms generate almost the same rate assignment results and the overall performance in terms of quality of reconstructed image are very close to each other. The only difference is the computational complexity of the rate assignment processing. For many applications, due to the limited computational resource, computing such optimum problem is always prohibited. Thus, find a practical rate assignment is necessary. At the same time, its result should close to the result from the theoretical computation. In the case of data packets, we need two steps to achieve a practical rate assignment: the first step is to select an S-rate set for a certain channel region in terms of SNR boundaries; the next step is to assign these S rates to T blocks in one image, with $T \geq S$. Such a channel operating region segmentation method has been used in an integrated communications systems design methodology [19].

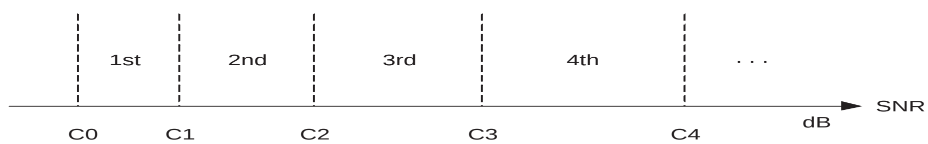


Fig. 3. AWGN Channel Operating Regions.

The selection procedure is described as follows. We have a rate set, $R = \{r_0, r_1, \cdots, 1\}$ where $r_i \geq r_j$, when $i \geq j$. Rate 1 means that there is no error protection employed. A segmentation of an AWGN channel is represented as $[C0, C1), [C1, C2), [C2, C3) \ldots$ where, Cn denotes the boundary of the channel operating region given in terms of E_b/N_0. Fig. 3 shows the segmentation of the channel. The segmentation is guided by the performance of the corresponding RCPT codes which means for a certain region, code rate with a performance (typically described as frame/bit error rate) lower than some value (e.g., 10^{-5}) is selected. The first S rates in set R are selected for channel range $[C0, C1)$. For the next channel range $[C1, C2)$, we again select S rates with the implementation as follows: S' rates after the previous S rates in set R are selected and the first S' rates within the previous S rates are deleted. Thus, a new set of S rates are found by shifting S' rates in the set each time assuming $S' \leq S$. Such a procedure is continued for different channel regions as E_b/N_0 increases. An example for $S = 3$ and $S' = 1$ is illustrated in Fig 4. At a certain point, when the E_b/N_0 is high enough, we could either fix a rate for all the blocks, or set the rate equal to 1 thereby providing no channel error protection. The rate set assignment by a sliding movement along the channel SNR axis matches well to the RCPT codes performance. That is because the performance curves for code rate set II are almost equidistance and parallel to each other over their "water fall" region. Within a certain channel range, we find a set of S rates as described above. These rates are assigned for T blocks. With $T \geq S$, each rate is used for at least one block. For a small number of blocks, setting $T = S$ is a simple approach that is used in some of the previous work mentioned above. For the image header, a similar method is employed with one code rate being sufficient for one region. The criteria are still based on the RCPT code performance. In a certain channel region, the code rates whose performance reduce the probability of BHT to be less than the reciprocal of the length of the coded header are candidates. This will guarantee that errors occur with a very small probability. Among them, the highest rate is no doubt the final selection because it requires the least bandwidth.

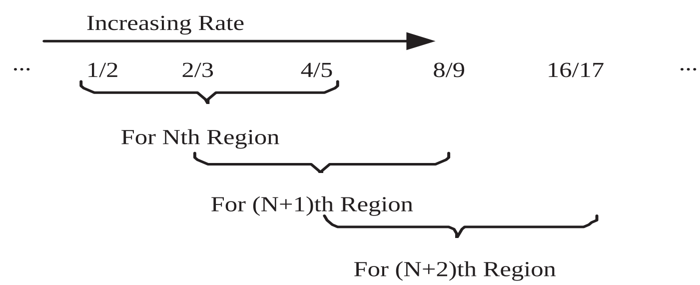


Fig. 4. Rate Selection with $S = 3$ and $S' = 1$.

4 Selected Simulation Results

Simulation results are presented in this section. The Lena image with 512×512 pixels each having 8-bit grayscale was used. In this study, one tile and 3 quality layers with an LRCP progressive mode are used. A typical 6-resolution encoding is obtained after applying a DWT five times. The source coding rates are set at 0.1, 0.5, and 1.0 *bpp*. The corresponding lengths of the headers and the layers are shown in Table 1 where the length is given in units of bytes. Also, the header length includes the 2 bytes for end of codestream (EOC). The Turbo code used here has constraint length 5, and the two constituent recursive systematic convolutional (RSC) encoders have the octal form generator matrix $(g1, g2) = (31, 27)$. The mother code rate is punctured with the puncturing pattern P(1,2) and forms two code rate sets, I and II, as mentioned previously. From a practical point of view, rate one, corresponding to no channel protection provided, is also included in these sets. In all the experiments described herein, we employ BPSK modulation and coherent demodulation over an AWGN channel with E_b/N_0 starting at 0 dB.

The two hybrid FEC/ARQ methods are simulated first. We select the source coding rate to be 0.5 *bpp*. The header block is padded with zero inputs in order to force the state of the first RSC encoder back to the zero state. The BHT rate is calculated as follows. First, we send the header through the channel, then decode it and add the noise-free layered data packets. If the source decoder can reconstruct the image successfully, it means that the transmitted header is useful, otherwise it is a bad header. Fig. 5 shows the BHT rate versus channel SNR for the image header by using the code rate set I. Table 2 shows the division of the channel operating regions and the corresponding rates selected for those regions that were used with Scheme I. The maximum retransmission number is set at $M_c = 2$. Using our selection method, which coincides with the solution of the theoretical model, in most channel situations we achieve error-free transmission with no retransmissions being required. Thus, the throughput values are very close to the selected rates. For lower SNR, which corresponds to a high BHT rate, ARQ method is need to increase the success transmission of header. Scheme II adapts to the channel situation and could be used for some selected SNR values by assigning two compatible rates. Scheme II was tested with $E_b/N_0 = 1.0, 2.0, 2.1, 2.9\ dB$ and rate sets $(1/2, 1/3), (2/3, 1/2), (2/3, 1/2), (4/5, 2/3)$ which were selected from the code rate set II. Table 3 shows the corresponding throughput values which are very close to the first rate in each set since the probability of requiring a second transmission is very small. When using Scheme II, success is achieved after only one transmission with a very high probability, and hence, the decoding delay is greatly reduced compared to Scheme I for some values of E_b/N_0. For the RCPT codes, the performance-guided code rate selection methods achieve near error-free transmission over the channel regions of practical interest with a high probability of needing only one or two transmissions.

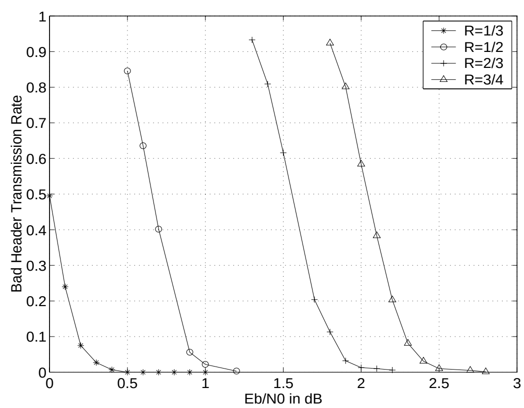


Fig. 5. Bad header transmission rate versus E_b/N_0. The first four rates in channel code rate set I were used.

Table 1. Lengths of image headers and layers.

source rate (bpp)	$r = 0.1$	$r = 0.5$	$r = 1.0$
header	482	637	741
layer1	1267	2883	15620
layer2	940	4639	8144
layer3	577	8098	8034

Simulation of data packet transmission is also performed over a binary input AWGN

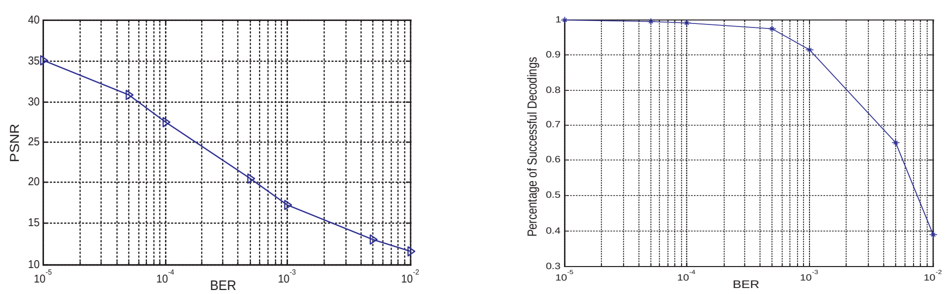
Table 2. Simulation results for hybrid Scheme I.

$E_b/N_0(dB)$	Rate	M_c	M	η_{SW}
$0 - 0.5$	$1/3$	2	2	$0.207 - 0.333$
$0.5 - 1.2$	$1/3$	2	1	0.333
$1.2 - 2.2$	$1/2$	2	1	0.500
$2.2 - 3.0$	$2/3$	2	1	0.667
above 3.0	$4/5$	2	1	0.750

Table 3. Simulation results for hybrid Scheme II.

$E_b/N_0(dB)$	Rate	η_{IR}
1.0	$1/2, 1/3$	0.500
2.0	$2/3, 1/2$	0.665
2.1	$2/3, 1/2$	0.666
2.9	$4/5, 2/3$	0.798

channel. We first show the results for $E_b/N_0 = 2.8\ dB$. All the headers are protected by code rate 2/3 according to Table 2. Even in cases where the header in a codestream is transmitted error-free, there still exists a non-zero probability of decoding failure due to the high error rate in the data packets for certain SNR conditions. Fig. 6 represents the percentage of successful decoding operations and the corresponding PSNR versus different bit error rates in the transmission of data packets. This general result shows that if the bit error rate is below 10^{-4}, we can achieve a high probability of successful decoding while the reconstructed image has a value of PSNR above 27 dB. This implies that we should choose a set of code rates that reduces the BER below 10^{-4} for a certain channel operating condition. Using this approach, we transmitted the test image using the parameters shown in Table 1 and the five different code rate sets presented in the first two columns of Table 4. Each quality layer is divided into fixed length frames (517 bytes) for channel coding. In order to simplify the parsing procedure, the last frame in each layer is padded with bits from the next layer to meet the length requirement. The first two code rate sets, $\{2/3, 2/3, 2/3\}$ and $\{4/5, 4/5, 4/5\}$, are EEP schemes. The first code rate set, $\{2/3, 2/3, 2/3\}$, is used as a rate constraint (assume the total length that can be consumed by other code rate set is shorter than the corresponding total length of this code rate set). When considering the EEP method only, the best choice should be the second code rate set, $\{4/5, 4/5, 4/5\}$, which contains the next available rate of 4/5. We have compared the performance of the other three UEP schemes with it. The aim is to find an UEP code set that has a corresponding normalized length (channel coded length normalized to the length of the corresponding codestream) curve which is below that of the all-2/3 code rate set while its corresponding PSNR curve is above that of the all-4/5 code rate set. Results shown in Table 4 are arranged in pairs: PSNR (in dB)/Normalized length. From Fig. 7 and Table 4, we can find that code rate sets 3, $\{2/3, 4/5, 8/9\}$, and code rate set 4, $\{2/3, 4/5, 4/5\}$, both satisfy the requirement for all three source rates $0.1, 0.5$, and 1.0 bpp. Since code rate set 4 provides stronger protection than code rate set 3, it outperforms code rate set 3 by 0.8 dB at 0.1 bpp, 1.4 dB at 0.5 bpp, and 1.7 dB at 1.0 bpp. But when comparing bandwidth efficiency, code rate set 3 requires less bandwidth expansion than code rate set 4. At source rate 0.5 bpp, its length after channel protection is even shorter than that of the all-4/5 rate set. For those applications where bandwidth and/or real-time image transmission are the key design factors, code rate set 3 is obviously the best choice. As mentioned before, with increasing channel SNR, the rate set should also be updated by sliding one rate in, and another rate out, to create the new code rate set for the next channel operating region. We tested this method for $E_b/N_0 = 3.5$ dB. By selecting $S = 3$ and $S' = 1$, the newly selected channel rate set is $\{4/5, 8/9, 16/17\}$. This time, the two associated EEP sets are $\{4/5, 4/5, 4/5\}$ and $\{8/9, 8/9, 8/9\}$ while the image header is protected using the code rate 4/5. Simulation results in Table 5 show that the third code rate set, $\{4/5, 8/9, 16/17\}$, has the same PSNR and bandwidth expansion behaviors as produced by the third code rate set in Table 4. Typical, yet similar, systems presented in [2], [6], [8]-[9], and [12] are studied on BSC channels with practical crossover error probabilities in the range $\rho = 10^{-1} - 10^{-3}$. These systems use different pre-selected channel code rate sets. That makes it difficult to compare the various results obtained from the different systems, including the system described in this paper. But all their UEP gain compared to EEP have the same behaviors as ours. The RCPT performance-guided selection idea presented here also provides a guideline when selecting code rates on other pre-selected rate sets. When the number of rates which could be selected under certain channel conditions is small, as is the case in most real-world applications, the sub-optimum solutions obtained from the theoretical models are always exactly as, or very close to, that of the intuitive segmentation scheme described.



(a) (b)

Fig. 6. PSNR and percentage of successful decodings versus BER when transmitting data packets: (a) PSNR behavior for different bit error rates, (b) the corresponding percentage of successful transmissions of image headers.

It is worth pointing out that applications and/or end users can control the source rate for each layer in one codestream to adjust the PSNR/normalized length difference between the channel code rate sets 3 and 4. Let's consider the source code rate to be 0.5 *bpp*. The lengths of the three quality layers shown in Table 1 correspond to the source code rates 0.1, 0.25, and 0.5 *bpp*. Based on this source code rate assignment, the corresponding PSNR and normalized length values for channel code rate set 3 and 4 when $E_b/N_0 = 2.8\ dB$ are shown in Table 4 as 28.91/1.242 and 30.33/1.304, respectively. The difference between them is $\Delta = 1.42/0.062$. Table 6 presents two other source code rate assignments for the three quality layers. Both assignments move more source bits to the third quality layer. Since the only different code rate between code rate sets 3 and 4 is the last one in their sets, moving bits from other layers to the third layer produces two distinct effects. First, in both of the code rate sets, the third code rate has the least error protection ability compared to the first two rates, so moving more bits to the third layer decreases the overall PSNR values. However, the code rate 4/5 in code rate set 4 has better error protection ability than the code rate 8/9 which is contained in code rate set 3, such a source bit adjustment increases the difference in PSNR between the two code rate sets. Second, the third code rate in both of the code rate sets provides the minimum bandwidth expansion compared to the first two code rates, so moving more bits to the third layer decreases the bandwidth consumption. The difference in the normalized lengths between the two code rate sets also increases. Table 6 shows the increase in PSNR and the decrease in normalized length for any one of the two channel code rate sets. The table also shows the increase in Δ between the two sets when more source bits are moved to the third layer. Increasing PSNR and reducing bandwidth expansion is a tradeoff when selecting between code rate sets 3 and 4 whenever the source bit allocation changes. As long as the PSNR performance is better than the second EEP code rate set, using code rate set 3 reduces the bandwidth expansion more than that of using code rate set 4. Such a selection avoids the need for recalculating the optimum dynamic programming problem which is very difficult in most real-world applications.

Table 4. PSNR and Normalized Length Data for different code rate sets. Channel $SNR = 2.8\ dB$. Source coding rates are 0.1, 0.5, and 1.0 *bpp*.

Set	Channel Code Rates	$r = 0.1\ bpp$	$r = 0.5\ bpp$	$r = 1.0\ bpp$
1	$\{2/3, 2/3, 2/3\}$	28.43/1.500	35.07/1.500	37.97/1.500
2	$\{4/5, 4/5, 4/5\}$	25.53/1.287	27.19/1.260	26.79/1.256
3	$\{2/3, 4/5, 8/9\}$	25.71/1.362	28.91/1.242	33.84/1.345
4	$\{2/3, 4/5, 4/5\}$	26.57/1.384	30.33/1.304	35.52/1.376
5	$\{2/3, 8/9, 8/9\}$	19.40/1.326	20.69/1.206	30.52/1.314

5 Conclusions

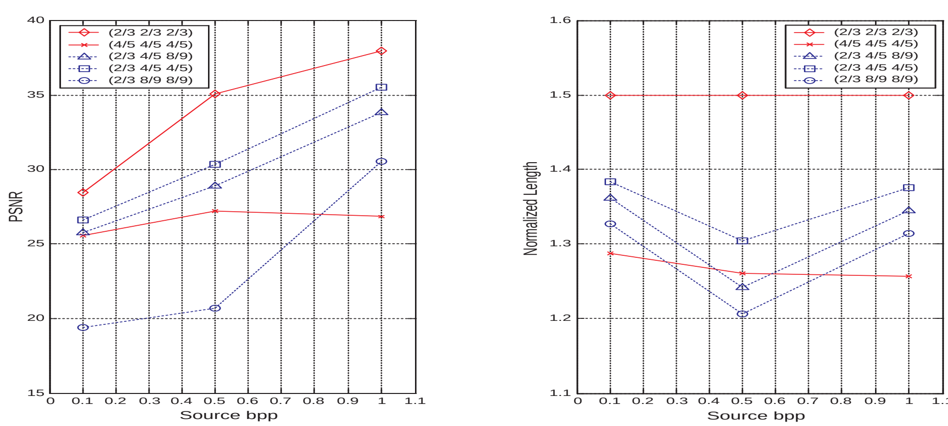
A new design for JPEG2000 codestream transmission over noisy channels has been proposed. It has been shown that header and data packets must be treated separately to achieve im-

Table 5. PSNR and Normalized Length Data for different code rate sets. Channel $SNR = 3.5\ dB$. Source coding rates are 0.1, 0.5, and 1.0 *bpp*.

Set	Channel Code Rates	$r = 0.1\ bpp$	$r = 0.5\ bpp$	$r = 1.0\ bpp$
1	$\{4/5, 4/5, 4/5\}$	27.88/1.250	33.46/1.250	36.34/1.250
2	$\{8/9, 8/9, 8/9\}$	25.24/1.143	27.02/1.130	27.23/1.128
3	$\{4/5, 8/9, 16/17\}$	25.77/1.181	28.38/1.121	33.23/1.172
4	$\{4/5, 8/9, 8/9\}$	27.15/1.192	30.26/1.152	35.50/1.188
5	$\{4/5, 16/17, 16/17\}$	19.02/1.163	19.87/1.103	31.23/1.157

Table 6. PSNR and Normalized Length Data for different source code rate assignments. Channel $SNR = 2.8\ dB$. Source coding rate is 0.5 *bpp*.

Rate Assignment for Layers	Code Rate Set 3	Code Rate Set 4	Δ
0.1/0.25/0.5 *bpp*	28.91/1.242	30.33/1.304	1.42/0.062
0.05/0.15/0.5 *bpp*	28.12/1.188	29.76/1.275	1.64/0.087
0.03/0.1/0.5 *bpp*	27.51/1.165	29.23/1.265	1.72/0.100



(a) (b)

Fig. 7. PSNR performance and normalized length for different sets of code rates: (a) PSNR performance for different source coding rates, (b) normalized length of the channel coded data packets for different source coding rates.

proved system performance. Two hybrid FEC/ARQ schemes for header and an UEP scheme for data packets are proposed. Rate-distortion channel rate allocation models are established. A practical rate selection method based on channel segmentation has been developed that is easier to implement and avoids solving the complicated theoretical optimum bit allocation problem. The qualities of the reconstructed images protected by the proposed UEP schemes are better than those obtained using EEP schemes for various source coding rates. For higher source coding rates, using the third code rate set, the PSNR improvement in the quality of reconstructed images increases by 6 to 7 dB while still satisfying the bandwidth expansion constraints.

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