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Rate Efficient Wireless Image Transmission using MIMO-OFDM

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Abstract—In this paper, we propose a rate efficient JPEG 2000 image transmission system over hybrid wireless networks using MIMO-OFDM. The objective is to minimize the expected end-to-end distortion given the rate constraint, which is achieved by jointly adjusting source coding schemes and channel coding rates. In the proposed system, MIMO-OFDM is used to increase the channel capacity and mitigate the inter-symbol interference, variable rate space frequency codes and Reed Solomon codes to combat the channel errors, and error resilient source coding schemes to restrict the error propagation. In case network congestion may happen, packet erasure codes are used to alleviate the packet dropping. The advantages of the proposed system lie in three aspects: adaptivity, optimality, and low complexity. Based on the characteristics of the image content, the estimated channel conditions, and the distortion constraint, the proposed low-complexity joint source channel coding and rate control algorithm adjusts the coding and transmission strategies adaptively, which can approximate the optimal solution with a tight bound.

Index Terms—Wireless image transmission, JSCC, variable rate space frequency codes, MIMO-OFDM, JPEG 2000, rate efficiency, packet erasure code.

I. INTRODUCTION

Multimedia communication over wireless channels has become more and more popular with the development of wireless cellular systems and wireless local area networks. However, to design an efficient multimedia transmission system over hybrid wireless networks, there still exist many challenges, of which some are caused by severe wireless channel conditions, some due to the special characteristics of compressed multimedia data, and some from resource limitation, such as bandwidth and power constraints.

Due to severe wireless channel conditions, such as path loss, fading, co-channel interference, and noise disturbances, the capacity of the wireless channels is much lower than the wired channels, and the bit error rate (BER) is much higher [1]. Meanwhile, the network throughput may fluctuate due to time-varying characteristics of wireless channels. The severe channel conditions have placed a major obstruction for designing efficient multimedia transmission systems over wireless environments.

Besides high BER's due to severe wireless channel conditions, as a consequence of resource limitation, multimedia transmission may also suffer from network congestions. Congestion occurs on a communication link whenever the amount of traffic injected on that link exceeds its capacity. This excess traffic causes queuing delays to increase rapidly

as buffers fill up, and in extreme cases can cause the buffers to overflow, losing packets [2]. Since multimedia data usually has delay constraint, packet dropping can degrade the performance drastically.

Since multimedia data contains a lot of redundancy, to efficiently utilize limited resources, source compression is always necessary. Compared with general data, the compressed multimedia data has special characteristics, such as *unequal importance*, *error tolerance*, and *constrained error propagation*. *Unequal importance* denotes that different part of the compressed bitstream exhibits different perceptual and structural importance. Unlike reliable data communication¹, where any error can blow up the whole transmitted data, *error tolerance* means that even if errors are introduced, the original information may still be reconstructed with tolerable distortion. To improve the compression efficiency, prediction or variable length coding (VLC), which is very sensitive to unpredictable errors, has been used by most of existing multimedia compression systems. *Constrained error propagation* denotes the phenomena that if bits are corrupted, the neighboring bits are likely to become useless as well, especially in the case when variable-length coding or prediction is applied. Meanwhile, the affected bits can be restricted inside a certain range by applying error resilient coding schemes. These characteristics differentiate multimedia transmission from general reliable data communication.

Unlike wired networks, the resources in wireless networks are always limited. For instance, wireless networks are always low-bandwidth, and sometimes, power supply is limited. In general, maintaining a good quality of service is contradictory to minimizing resource usage. In wireless multimedia communications, the contradiction becomes more salient due to the adverse channel conditions and special characteristics of compressed image content. To address this contradiction, besides improving channel code error correction capability, source coding efficiency and error resilience, schemes based on joint source channel coding (JSCC) principle can be applied to utilize the limited resources in a more efficient way.

In this paper, targeting on the point-to-point image transmission over hybrid wireless networks, we propose a rate efficient system to minimize expected end-to-end distortion given the rate constraint, or to minimize overall rate consumption given the expected end-to-end distortion constraint. In the proposed system, an image is first encoded as a scalable bitstream

¹In this paper "reliable" is used to denote that no error can be tolerated.

with multiple quality layers that is optimal in the R-D sense. Given the estimated channel condition, the characteristics of the image content, and the end-to-end distortion constraint, the proposed system can adaptively determine the number of layers to be transmitted and adjust the source coding rate, the source level error resilient scheme, the channel coding schemes and rates for each layer through the proposed low-complexity joint source channel coding and rate control (JSC-CRC) algorithm.

In the proposed system, JPEG 2000 [3] is adopted as the source coding standard; it has the following desirable properties: state-of-the-art compression performance, quality scalability, and strong error resilience [4], [5]. These three properties are of importance for image transmission over error-prone channels, where high compression performance can be used to save limited resources, quality scalability to facilitate the unequal error protection (UEP), and error resilient coding to restrict the error propagation range. In our work, the variable rate space frequency codes proposed in [6], [7] are used as channel codes, since it can work well in frequency-selective fading channels, and can provide variable degree of protection using the same codec with low coding complexity, which is very suitable for hardware implementation. Since errors usually exhibit bursty pattern in fading channels, Reed Solomon (RS) [8] codes can be used as the inner codes to further improve the system performance. In case network congestion may happen, packet erasure codes [9] are applied to combat packet dropping. To increase the channel capacity, exploit the spatial diversity, and mitigate the inter-symbol interference, MIMO-OFDM is used to transmit and receive the protected bitstream [10]–[13].

The superiority of the proposed system to the existing systems, as well as the contribution of the paper, lies in the following properties: *adaptivity*, *optimality*, and *low complexity*. These are the key properties for efficient multimedia transmission systems. Adaptivity indicates that the proposed system is adaptive to the channel condition, the characteristics of the image content, and the QoS constraint. Optimality means that it can approximate the optimal solution with a tight bound. Low complexity lies in that the JSCCRC algorithms can be executed with negligible time, which is very suitable for online processing. Another advantage of the proposed system is that it exploits the error resilient coding schemes at the source coding stage. Currently, no existing image or video transmission system has incorporated all these properties at the same time. Most existing systems focus on reducing the channel BER, this paper shows that restricting the error propagation range by applying source error resilient coding schemes can improve the system performance dramatically in many situations.

The rest of the paper is organized as follows. Section II describes the underlying channel model, discusses the available techniques to tackle those design issues and presents the related work. Section III describes the proposed system, and formulates the constrained optimization problem. The channel codes to be used are introduced in Section IV. The JPEG 2000 source coding procedure is introduced in Section V, followed by the description of the applicable error resilient source

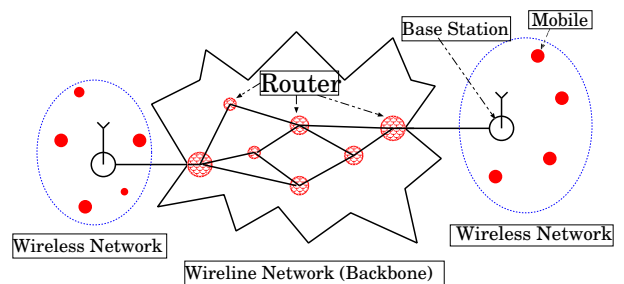


Fig. 1. A Hybrid Network Architecture

coding schemes as well as their performance analysis. The proposed low-complexity JSCCRC algorithm is presented in Section VI. Section VII presents the experimental results and the performance comparison. Finally, Section VIII concludes this paper and suggests the future work.

II. AVAILABLE TECHNIQUES AND RELATED WORK

In this section, we first presents the channel models to be based on, then we discuss the available techniques to tackle those design issue introduced in Section I, as well as how to efficiently combine these techniques to achieve better performance. At the end of this section, we shows the related work, and address their pros and cons.

A. Channel Model

In this paper, we consider a *hybrid wireless network* model, which is composed of several small wireless networks connected by a wired network (backbone). The mobiles in the same wireless network can communicate directly with each other, and mobiles within different wireless networks can communicate with the help of their base stations and the backbone, as illustrated in Fig. 1.

We assume the hybrid wireless network is based on packet-based network protocols, where data is transmitted packet by packet. Packet dropping is only caused by network congestion, which is different from the reliable data communication system, where any intermediate errors introduced during transmission will cause the packet to be dropped. We assume that the transmitter can know the estimated channel conditions between the transmitter and the receiver, such as average signal-to-noise (SNR) and network packet dropping rate (PDR), but does not need to get the instantaneous feedback from the receiver. We assume that no intermediate transcoding is applied during transmission.

The wireless part of the transmission path can be modeled as a fading channel, and the wired part can be modeled as a packet erasure channel. Based on the assumptions that no extra transcoding is applied and intermediate errors will not cause packet dropping, the hybrid wireless network model can be viewed as the concatenation of a wireless fading channel with a packet erasure channel, where the two channels are independent. The average BER of the wireless fading channel can be calculated as the ratio of the number of error bits introduced to the total number of transmitted bits, and the

average packet dropping rate (PDR) calculated as the ratio of the number of received packets to the total number of transmitted packets, where BER is independent of PDR.

B. Discussion of Existing Techniques

As discussed in Section I, to design an efficient image transmission system, we need to combat with limited resources, severe wireless channel conditions, network congestion, and error propagation. In this paper, we focus on the bit rate constraint, which is a function of transmission time and channel bandwidth. To combat the error propagation, error resilient source coding schemes can be applied, which are addressed in Section V.

Multimedia communication is bandwidth-consuming, while wireless channel is bandwidth-limited. To tackle this conflict, besides performing source compression, increasing the channel capacity is also a good solution. One approach to increase the channel capacity is to employ multiple input multiple output (MIMO) systems, which have been playing a significant role in the development of high data rate wireless communication systems. By employing multiple transmit and receive antennas, the information capacity of wireless communications increases linearly with the number of transmit antennas, given the number of receive antennas is no less than the number of transmit antennas [10], [11]. Another advantage of MIMO system lies in that it provides spatial diversity to combat the severe wireless channel conditions.

Next we discuss how to combat the high BER of the wireless channels. According to [14], the average BER is a function of the channel code characteristics, the types of waveforms used to transmit the information over channel, the transmitter power, the characteristics of the channel, and the method of demodulation and decoding. To maximally reduce the BER, all the above should be considered jointly. In this paper, the wireless channels are fading channels, and we assume the transmitter power is fixed during transmission. Next, we consider the modulation and channel coding schemes.

For wireless fading channels, multipath is a major obstruction to reduce the BER. In many situations, especially in high bit rate communication systems, the delay spread of the multipath may be higher than the symbol rate, which causes severe inter-symbol interference (ISI), and degrades the system performance drastically. To mitigate the ISI, orthogonal frequency division multiplexing (OFDM) has been shown to be one of the most promising techniques, as it eliminates the need for high complexity equalization and offers high spectral efficiency [12], [13].

Besides efficient modulation, channel codes have also been widely used to combat the high BER by exploiting the temporal correlation and spatial diversity, and adding structural redundancy. Based on the dependence on the instantaneous feedback information, the channel coding schemes can be partitioned into two categories: feedback-based schemes, such as Automatic Repeat reQuest (ARQ), and non-feedback-based schemes, such as forward error protection (FEC) [8], [15]. In this paper, since we assume the transmitter can not get the instantaneous feedback information from the receiver, only

FEC are considered. There exist many FEC channel codes, such as the conventional linear block codes, convolutional codes, and the state-of-the-art space time and space frequency codes for MIMO systems [7], [16]–[18]. When designing a system, which codes should be adopted depends on many issues, such as their error correction capability, coding complexity, feasibility, and availability.

Since network congestion may happen, to combat the packet dropping, erasure codes can be applied, which is a special class of FEC codes. In [9] they have presented a class of efficient packet erasure codes and shown their feasibility in real-time applications. Another class of useful FEC codes is rate compatible codes [19], which are suitable in multicast application with different users having different channel conditions.

Due to the fading characteristic of the wireless channel, the errors usually exhibits bursty pattern. However, most existing channel codes do not work well with bursty errors. To solve this problem, one simple but efficient approach is to use interleaver, which can translate the bursty errors to non-bursty errors. The drawback is that it introduces extra decoding delay and computation overhead. Another approach is to use Reed Solomon (RS) codes, since they have excellent bursty error correction capability.

Now we can see that build a multimedia transmission system over error-prone channels, two components are necessary: source encoder and channel encoder. According to the source channel coding separation theorem [20], the multimedia data can be first compressed using source encoder, then transmitted error-freely by applying channel coding protection. However, the theorem does not hold in practice, since we require the channel codes having short block length, small delay, and low coding complexity, that is, no practical channel codes can guarantee the error-free transmission. Since error can propagate inside the compressed bitstream, special action should be taken to combat the effects, which can be achieved by applying error resilience source coding schemes to constrain the error propagation.

Based on the fact that source channel coding separation theorem does not hold in practice and the special characteristics of the compressed multimedia data, joint source channel coding (JSCC) has been proposed. Now the design of multimedia transmission system over error-prone channels under rate constraint can be summarized as follows. First, we determine the source coding techniques, channel coding techniques, modulation scheme, and wireless transmitter system to be used. Then, we need to design an efficient rate allocation scheme to assign the source coding rate and channel coding rate such that the system performance is optimized given the total rate constraint and current channel conditions.

C. Related Work

In the last decades, JSCC has been widely studied and been shown to be an efficient way for multimedia transmission over error-prone channels. Roughly, the existing JSCC schemes can be divided into two categories: joint source channel matching (JSCM) and *true* JSCC. JSCM, as in [21]–[28], denotes that the source data is optimized for a noiseless channel, and is

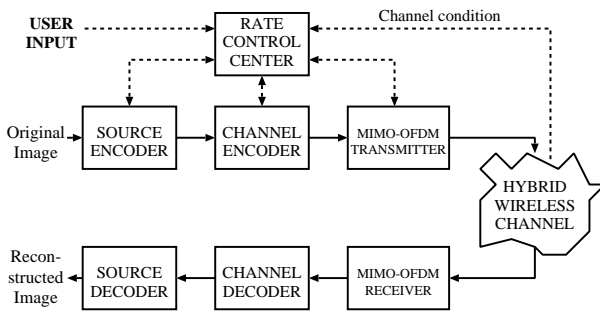


Fig. 2. Architecture of the proposed image transmission system

then made robust toward channel errors by optimally allocating channel protection rates or power level such that the objective is achieved. True JSCC, as in [29]–[32], means that the source coding and channel coding is combined together to minimize the distortion, given rate constraint.

Although the existing JSCC or JSCM schemes have achieved a lot of success, there still exist some drawbacks and further improvement are needed. First, most of the existing *true* JSCC schemes have high complexity, and the scalability and feasibility are still a major issue. Another drawback of existing JSCC schemes is that no source error resilience schemes have been included, where they mainly play with quantization and FEC. Second, most JSCM-based systems are based on the assumption that once errors are introduced, all the following data can not be used, which is not true in general. Third, due to the high computation complexity of the rate allocation algorithm, seldom can be used for on-line processing. Fourth, the existing systems only target on certain simplified channel models, such as BSC, AWGN channel, and seldom have based on realistic channel models.

III. SYSTEM DESCRIPTION AND PROBLEM FORMULATION

In this section, we propose an efficient JSCC-based JPEG 2000 image transmission system targeting on the hybrid wireless network model. The objective of the system is to minimize the expected end-to-end distortion given the rate constraint, and the system should be adaptive to channel condition. This is achieved by jointly allocating the source rate, source error resilience coding scheme, and channel code rates, which is a *true* JSCC scheme.

A. System Description

The block diagram of the proposed system is illustrated in Fig. 2. The transmitter is composed of four components: source encoder, channel encoder, MIMO-OFDM transmitter, and JSCC rate controller (JSCCRC). The description of each component, as well as the interaction among them, are illustrated by Fig 3. The receiver end includes three components: MIMO-OFDM receiver, channel decoders, and source decoder.

The JPEG 2000 source encoder is adopted to generate quality scalable bitstream with multiple layers under certain rate constraint, either through compressing the raw image, or through transcoding a compressed image. Meanwhile, to

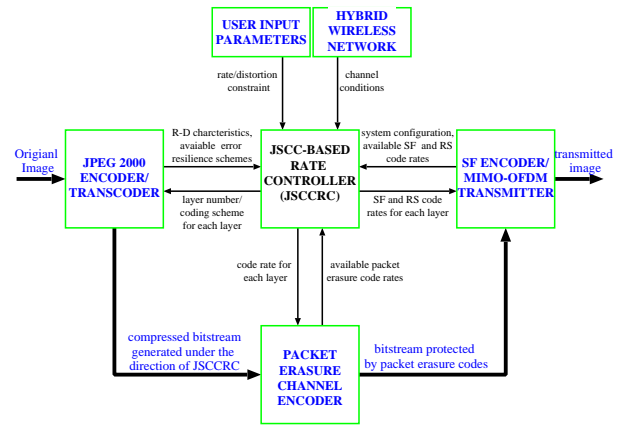


Fig. 3. The Interaction description

restrict the error propagation, some extra structural redundancy can be inserted by applying resilient coding schemes. The detailed description of JPEG 2000 encoder and the available error resilient coding schemes are presented in Section V.

To combat the channel errors introduced by the fading channel, the variable rate space frequency (SF) codes [6], [7] and the Reed Solomon (RS) codes are adopted as channel encoders. To alleviate the effects of network congestion, the packet erasure (PE) codes presented in [9] are used. Since the errors introduced in fading channels exhibit bursty pattern, interleaving can be applied to translate these bursty errors to non-bursty errors. To improve the error correction capability of, interleaving is used to translate the bursty errors to non-bursty errors. At the transmitter, we first use RS codes to protect the compression bitstream, then we perform byte-level interleaving inside a quality layer, finally, the interleaved data protected further protected using SF codes. In the proposed system, RS codes are used to protect the header information in the compressed bitstream. The description of variable rate SF codes, RS codes, and PE codes are presented in Section IV.

As the core of the proposed system, the JSCCRC performs optimal rate control between the source coding stage and the channel coding stage. For each image to be transmitted, the JSCCRC gets the rate constraint from the system, and the current estimated channel conditions from the wireless environments. Based on the available rate and the image size, JSCCRC determines how many quality layers need to be generated, where we assume the length of each quality layer is fixed. The source encoder/transcoder then generates the desired number of quality layers in a rate distortion (R-D) optimal sense. As the side effect, the R-D characteristics of each layer are also generated. Based on the user input information and the R-D characteristics of each generated layers, as well as the available source error resilience coding schemes and the channel coding rates, the JSCCRC then tries to find how many layers need to be transmitted, and the optimal coding and transmission strategy for each layer, which is a combination of the source error resilience coding scheme, the SF code rate, the RS code rate, and the PE code rate.

After the optimal coding and transmission strategy for each layer has been determined, the source encoder then generates

the desired bitstream, which is further protected by the channel encoders using desired rates, and transmitted by the MIMO-OFDM system. At the receiver end, the bitstream is then received and decoded by the channel decoder and source decoder.

B. Problem Formulation

In this paper, we assume the channel conditions keep unchanged during the transmission of each image, and assume the system configuration keeps fixed. For layer l , let S_l^s , S_l^f , S_l^r , and S_l^p denote the set of available source error resilience coding schemes, the set of available SF code rates, the set of available RS code rates, and the set of available PE code rates, respectively, and let $s_l = \langle s_l^s, s_l^f, s_l^r, s_l^p \rangle$ denote the coding and transmission strategy. Let D_{total} denote the overall distortion if no layers are received. Let $G_l(s_l)$ denote the expected distortion reduction when layer l is coded and transmitted using strategy s_l . Let s_l^0 denote the layer l will not be transmitted.

Assume the distortion metric is additive, then the overall expected distortion reduction $G(\bar{s})$ can be calculated as

$$G(\bar{s}) = \sum_{l=1}^L G_l(s_l) \quad (1)$$

where $\bar{s} = \langle s_1, s_2, \dots, s_L \rangle$, and the total expected distortion $D(\bar{s})$ becomes

$$D(\bar{s}) = D_{total} - G(\bar{s}) \quad (2)$$

Let $R_l(s_l)$ denote the number of bits needed by applying strategy s_l on layer l , then the overall rate $R(\bar{s})$ can be calculated as

$$R(\bar{s}) = \sum_{l=1}^L R_l(s_l) \quad (3)$$

Let R^M be the available rate, then the rate allocation problem can be formulated as a constrained optimization problem:

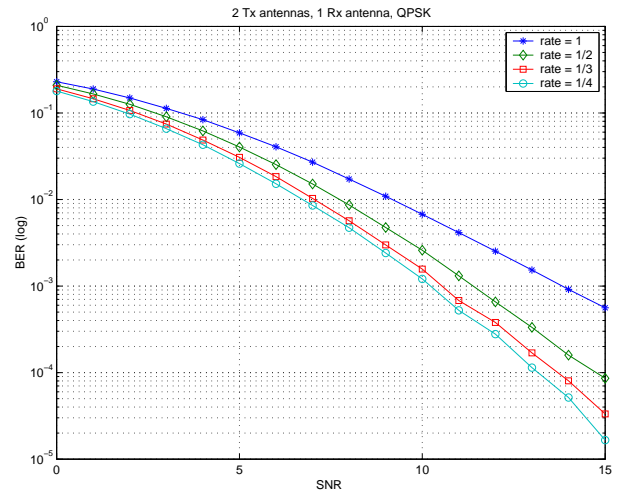
$$\arg \max_{\bar{s}} G(\bar{s}) \quad \text{s.t.} \quad R(\bar{s}) \leq R^M \quad (4)$$

IV. CHANNEL CODING TECHNIQUES

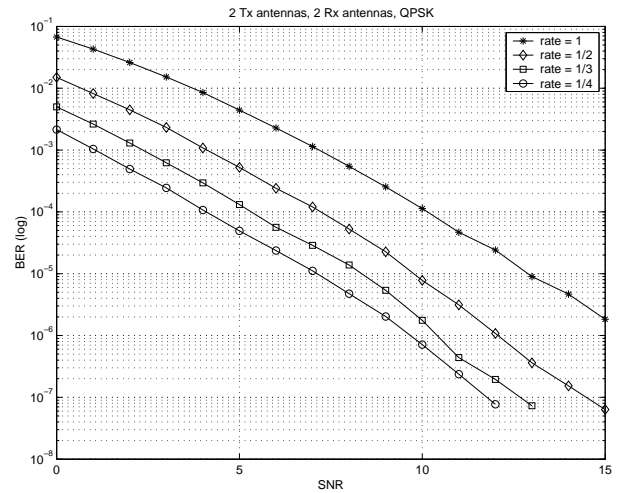
This section introduces the channel codes to be used in the system. The variable rate SF codes for MIMO-OFDM system are described in Section IV-A, as well as its advantages. Section IV-B introduces the Reed-Solomon codes. The packet erasure codes are introduced in Section IV-C.

A. Variable Rate Space Frequency Codes

The variable space frequency codes used in this paper is an extension of the variable rate space time trellis codes proposed in [6]. The codes can provide variable protection against the adverse effects of wireless channel. The method is based on the multiple trellis coded modulation (MTCM) [17] construction: the variable rate data protection is realized by changing the number of output symbols per state transition.



(a) two transmit antennas, one receive antenna



(b) two transmit antennas, two receive antennas

Fig. 4. Performance of SF codes under different rates

Given the average SNR, the average BER of the variable rate SF codes can be approximated by

$$P_e(r) = \frac{C_{MN}}{SNR^{1/r}} \quad (5)$$

where C_{MN} is a constant determined by the underlying trellis code and the MIMO-OFDM system configuration, i.e., the number of transmit antennas and receive antennas, and r is the rate of the code. Fig. 4 presents the simulation results of the SNR vs. BER curve for the SF codes used in the proposed system. Fig. 4(a) presents the code performance for 2 transmit antennas and 1 receive antenna system, and Fig. 4(b) presents the code performance for 2 transmit antennas and 2 receive antennas system. The simulation results are obtained based on 2-ray channel model.

Since the diversity and extra protection is obtained using simple repetition, the codes have very low complexity, which is very suitable for real-time applications. Another advantage is that these codes can work well at low SNR's. The disadvantage lies in that the available rates are not dense enough, and they can be outperformed by some codes with high complexity.

B. Reed Solomon Codes

Reed Solomon codes are a subset of Bose-Chaudhuri-Hocquenghem (BCH) codes and are linear block codes [8], [15]. A RS code over Galois field $GF(2^s)$ is specified as $RS(N, K)$ with s -bit per data symbol, where N is the total number of symbols in a codeword, K is the number of information symbols, and $N - K$ is the number of parity check symbols. A RS decoder can correct up to $(N - K)/2$ symbols that contain errors in a codeword. In this paper, the $RS(255, 223)$ code over $GF(2^8)$ is used. By letting a number of information symbols be zero during the encoding stage, not transmitting them, and then re-inserting them at the decoder end, we can generate new RS codes with different number information symbols (rates) using the same generator polynomial.

Assume the interleaving has been performed to translate the bursty errors to non-bursty errors, and assume the average BER is P_e , then for the $RS(N, K)$ codes derived from $RS(N, K)$ over $GF(2^s)$, the probability that the decoder can not correct all the introduced errors can be calculated by

$$P_{rs}(N, K) = 1 - \sum_{i=0}^{(N-K)/2} (P_s)^i (1 - P_s)^{N-i} \quad (6)$$

where P_s is the s -bit symbols error rate, and

$$P_s = 1 - (1 - P_e)^s \quad (7)$$

In the proposed system, the RS codes are used in two ways. First, they are used to protect the header information of the compressed source bitstream as an error resilient coding scheme. Second, they are used to combat the severe wireless channel condition combined with the SF codes, in this situation, the RS codes act as the inner codes, and SF codes act as the outer codes.

C. Packet Erasure Codes

In [9], Rizzo has proposed a simple but very flexible erasure code which can be used in network protocols. The code is based on Vandermonde matrices computed over $GF(p^r)$. For a (N, K) RS style packet erasure code, a minimum number of K source/parity packets suffices to recover the K source symbols. Given the packet dropping rate P_d , and the packet erasure code $PE(N, K)$, the probability that the packet can not be correctly recovered by the receiver can be calculated as

$$P_{pe}(N, K) = 1 - \sum_{i=0}^{N-K} (P_d)^i (1 - P_d)^{N-i} \quad (8)$$

V. JPEG 2000 AND ERROR RESILIENT CODING SCHEMES

In this section, we first introduce the JPEG 2000, a new still image coding standard [3], then we discuss the applicable error resilience coding schemes for JPEG 2000 compressed bitstream. Finally, we present the analytical performance analysis of these error resilient coding schemes are also presented.

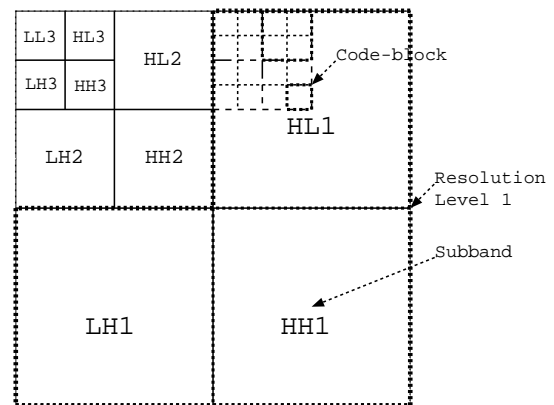


Fig. 6. Partitioning a wavelet-transformed component into resolution levels, subbands, and code blocks

A. Introduction to JPEG 2000

JPEG 2000 is a wavelet-based still image compression standard [33] using the embedded block coding with optimized truncation algorithm [4]. The general encoding structure is illustrated in Fig. 5. After performing inter-component transform, the wavelet transform decomposes each component into several *resolution levels*, each containing a series of *subbands*. After quantization, the coefficients in each wavelet subband are partitioned into regular arrays of *code blocks* for entropy coding. The hierarchical partition structure is illustrated in Fig. 6.

Each code block is entropy-coded independently using the MQ coder, which is essentially a bit-plane coder, relying on the use of classical context adaptive arithmetic coding [4]. Each bit-plane is coded by three passes, and each pass generates an embedded bitstream, called *coding pass*, to provide a variable quality contribution to the reconstructed image. Following entropy coding, a post compression rate allocation procedure selects coding passes from each code block in such a way that an optimal rate allocation is achieved, which minimizes the reconstructed image distortion under certain desired bit-rate constraint. The selected coding passes are packetized into several data packets, which are then assembled into the final coding stream. Each packet includes two parts: packet header and packet body. The header indicates which coding passes are included in this packet, and the body contains the actual data of the selected coding passes.

The packetization procedure imposes a particular organization on the selected coding passes to facilitate many of the desired coding stream features, such as *quality scalability*, which denotes that the bitstream contains embedded subsets, each of which represents an efficient compression of the original image at increased distortion [5]. Let n_l denote the number of bits in layer l , and w_l denote the average distortion reduction per bit for layer l , which is defined as the total distortion reduction over the total number of bits in this layer. Since the bitstream is generated in a R-D optimal sense, we always have $w_k > w_j$ for $j > k$.

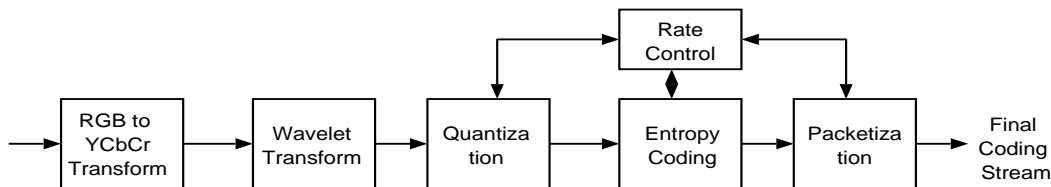


Fig. 5. Structure of JPEG2000 Encoder

B. Error Resilient Coding Schemes in JPEG2000

Since JPEG 2000 has heavily utilized the prediction and VLC, when errors are introduced into the compression bitstream, propagation is unavoidable. Besides using channel codes to alleviate the channel errors, applying error resilient schemes in the source coding stage to restrict the error propagation can be a good approach, since source encoder may have better understanding of the compressed data than the channel encoder.

The applicable source coding error resilient schemes in JPEG 2000 can be roughly classified into four categories: *artificial data partition*, *synchronization marker insertion*, *header protection*, and *modified entropy coding*. At the decoder end, besides utilizing the added redundancy to recover from errors, *error concealment* can also be used to combat the error effects.

In JPEG 2000, the artificial data partition is used to restrict the error propagation inside a small local area, such as in a code block. This is achieved by independent entropy coding on each code block. By varying the size of the code blocks, we can trade the coding efficiency with the error propagation range. Another artificial data partition technique is to assemble the final bitstream into several layers, where the errors in one layer will only affect the layers following it.

However, without the help of synchronization techniques, artificial data partition can not combat the error propagation by itself. Synchronization markers are used to synchronize the decoder when error happens, such that the error propagation can be restricted between two consecutive synchronization markers. The synchronization markers can be inserted in a hierarchical manner, such as the boundaries between adjacent layers, packets, code blocks, and coding passes, as illustrated by Fig. 7. In this paper, we call the bitstream between two consecutive synchronization markers as *segment*. The shorter the segment, the more quickly the decoder can recover from the errors, and consequently, the lower the coding efficiency.

Since for the compressed bitstream, header information plays a more important rule than body content, where if errors happen in the header of certain level, all the data this header corresponding to will become useless, it is natural to place more protection on headers than on body. In JPEG 2000, there exist two kinds of header information: main header and packet header. In the proposed system, the RS codes are used to protect these headers. To what degree the headers should be protected is decided by the JSCCRC.

Since error propagation is mainly caused by applying context-based VLC, to further restrict the error propagation, modified entropy coding schemes can be used, such as Fixed Length Coding (FLC). An extreme case is to bypass entropy

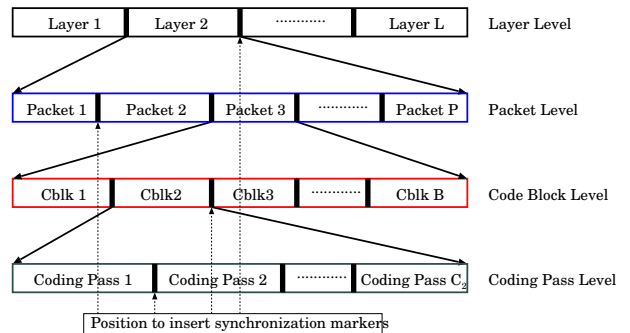


Fig. 7. Feasible Positions to Insert Synchronization Markers

coding, that is, only raw data is transmitted, we call this as *RAW coding*. For both FLC and RAW coding, error will not propagate if it happens in the packet body. Another modified entropy coding is to use reversible variable length coding (RVLC) [34], which has better coding efficiency compared with the FLC and RAW coding. However, if synchronization markers have already been used, RVLC can give us little improvement on error propagation restriction capability.

At the decoder end, once the JPEG 2000 decoder has detected and localized the errors, error concealment can be applied to further reduce the error effects.

C. Analytical Analysis of JPEG 2000 Source Error Resilient Coding Schemes

Let $s_l^s \in S_l^s$ be any feasible combination of the source error resilient coding schemes for layer l . Currently, s_l^s is a combination of code block size, synchronization marker level, and packet header protection rate. Define $E_l(s_l^s)$, the *average error propagation length* (AEPL) of s_l^s for layer l , as the average number of bits that become useless in this layer if a single bit error happens in the compressed bitstream. Let $R_l^s(s_l^s)$ denote the ratio of the total number of bits of layer l after applying s_l^s over the number of bits when no error resilient schemes are applied on this layer.

Let $e_l(s_l^s)$ denote $E_l(s_l^s)/R_l^s(s_l^s)$, which can be used as a performance criterion for the error resilient schemes. Assume the BSC is used, given the BER P_e , we say s_l^s has better performance than s_l^{s*} if $e_l(s_l^s) > e_l(s_l^{s*})$. Another performance criterion is the computation complexity, which will not be addressed in this paper.

In the following of this section we will show how to calculate $E_l^s(s_l^s)$, $R_l^s(s_l^s)$, and $G_l^s(s_l^s)$. Let P_l denote the total number of packers in layer l , C_l^p denote the number of code

blocks in the p^{th} packet of the l^{th} layer, n_l^p and h_l^p denote the body and header length of the p^{th} packet of the l^{th} layer. To fully exploit the capability of the error resilient schemes, in this paper we modify the JPEG 2000 codec such that the error resilient schemes can be applied layer by layer. The analysis is based on a BSC model with state transition probability P_e .

Based on the RS code rate, the synchronization marker insertion level, and the type of the entropy coding, $R_l^s(s_l^s)$ can be easily calculated. A general formula is

$$R_l^s(s_l^s) = \frac{1}{n_l} \sum_{p=1}^P \left(\frac{1}{r_{rs}(s_l^s)} h_l^p + a_l^p(s_l^s) \right) \quad (9)$$

where $r_{rs}(s_l^s)$ denotes the RS code rate used to protect the packet header, and $a_l^p(s_l^s)$ denotes the total number of bits for the packet body in the p^{th} packet of the l^{th} layer.

The AEPL of of scheme s_l^s for layer l can be calculated based on the synchronization marker insertion level. If no synchronization marker is used, then any error can blow up the whole bitstream, the the AEPL equals the total length of the bitstream. If only layer level markers are inserted, then any error in certain can blow up the whole layer, and $E_l(s_l^s) = n_l$. If both layer level and packet level markers have been inserted, then the AEPL of s_l^s for layer l can be approximated by

$$E_l(s_l^s) = \frac{1}{P_l} \sum_{p=1}^{P_l} n_l^p$$

Until now, we assume no RAW coding has been applied.

If markers have been inserted in all levels, then the AEPL of layer l can be approximated by

$$E_l(s_l^s) = \frac{1}{P_l} \sum_{p=1}^{P_l} \left((1 - P_e^H(s_l^s)) b_l^p(s_l^s) + P_e^H(s_l^s) n_l^p \right)$$

Here $P_e^H(s_l^s)$ denotes the probability that the packet header can be received error-freely. $b_l^p(s_l^s)$ is determined by the number of non-empty code blocks in this packet, the number of coding passes and the corresponding length in each code block, and whether RAW coding has been applied. When no RAW coding scheme is used, according to the coding passes dependence, if some errors have been introduced into a pass, then all the following passes in this code block will become useless. But if RAW coding scheme is used, then $b_l^p(s_l^s) = 1$.

Now we address to how calculate $G_l(s_l^s)$. Here we assume there is no packet dropping, and the channel is BSC with state transition probability P_e . According to the dependence of the coding passes in different layers, to precisely calculate $G_l(s_l^s)$, we need to know all the information before this layer, including all the coding and transmission strategy s_i 's with $i < l$, and all the R-D characteristics of each code block. In many situations, not all of these information are available, and even all are available, the complexity will be very high if all those information are considered.

In this paper, we make a *layer independence* assumption, that is, when calculating $G_l(s_l)$, we just simply assume that all the layers before layer l have been received error-freely. Then the calculated $G_l(s_l)$ is a upbound of the true $G_l(s_l)$. By multiplying a scalar to adjust the upbound, we

can approximate true $G_l(s_l)$, where the scalar depends on the channel condition. Based on the independence assumption, the upbound of $G_l(s_l^s)$ can be approximated by

$$G_l(s_l^s) = w_l n_l (1 - P_e E_l(s_l^s))$$

VI. OPTIMAL JOINT SOURCE CHANNEL CODING

Let $P_e(s_l)$ and $P_d(s_l)$ denote the average BER and the average PDR at the receiver end by applying strategy s_l on layer l . Based on the *layer independence* assumption, the upbound of the expected distortion reduction for layer l by applying strategy s_l can be approximated as

$$G_l(s_l) = w_l n_l (1 - P_e(s_l) E_l(s_l)) P_d(s_l) \quad (10)$$

and $R_l(s_l)$ can be calculated as

$$R_l(s_l) = n_l R_l^s(s_l^s) R_l^f(s_l^f) R_l^r(s_l^r) R_l^p(s_l^p) \quad (11)$$

where $R_l^f(s_l^f)$, $R_l^r(s_l^r)$, and $R_l^p(s_l^p)$ denote the inverse of the rates of SF code, RS code, and PE code used to combat the channel errors for layer l .

The basic idea of the optimal JSCC rate control procedure is to allocate the available rate step by step, at each step, it assigns some available bits to certain layer which can utilize these bits in a most efficient way, that is, it can achieve the largest distortion reduction. This can be done by using an greedy-like algorithm. Before describing the algorithm, we first give some definitions.

Let $\Delta G_l(s_l, s_l')$ be the *extra gain* by changing the schemes from s_l to s_l' , that is:

$$\Delta G_l(s_l, s_l') = G_l(s_l') - G_l(s_l) \quad (12)$$

and let $\Delta R_l(s_l, s_l')$ be the *extra rate consumption* by changing the schemes from s_l to s_l' :

$$\Delta R_l(s_l, s_l') = R_l(s_l') - R_l(s_l) \quad (13)$$

Now define the *normalized gain in rate* as

$$g_l(s_l, s_l') = \frac{\Delta G_l(s_l, s_l')}{\Delta R_l(s_l, s_l')} \quad (14)$$

Let $s^0, s^1, s^2, \dots, s^{|S|} \in S$ be the set of all available schemes ordered according to their rate consumption for a certain layer. Now we define the *feasible scheme set*, S_l , for each layer l as the subset of S where the necessary and sufficient condition that $s^i \in S$ belongs to S_l are

$$\min_{k < i} g_l(s^k, s^i) > 0 \quad (15)$$

and

$$\min_{k < i} g_l(s^k, s^i) > \max_{j > i} g_l(s^i, s^j) \quad (16)$$

That is, all the feasible schemes should reside on the convex hull of the rate-distortion curve for that layer. Consequently, for all $s_l^1, s_l^2, \dots, s_l^{|S_l|} \in S_l$, we have the following properties:

$$0 = R_l(s_l^0) < R_l(s_l^1) < \dots < R_l(s_l^{|S_l|}) \quad (17)$$

$$0 = G_l(s_l^0) < G_l(s_l^1) < \dots < G_l(s_l^{|S_l|}) \quad (18)$$

$$(\forall j > i) : g_l(s_l^i, s_l^{i+1}) > g_l(s_l^j, s_l^{j+1}) \quad (19)$$

For each layer l , the feasible protection scheme set can be obtained using a conventional convex hull analysis, as presented in Alg. 1.

Algorithm 1 feasible strategy set search

- 1: $S_l = \emptyset$;
 - 2: **for** ($s \in S$) **do**
 - 3: Calculate $G_l(s)$ and $R_l(s)$;
 - 4: **end for**
 - 5: Arrange the elements s 's of S in increasing order according to $R_l(s)$;
 - 6: Delete all the s' 's from S if there exists s such that $G_l(s) > G_l(s')$ and $R_l(s) \leq R_l(s')$, then add all the left s 's to S_l ;
 - 7: Perform convex hull analysis on S_l such that the elements of S_l satisfying inequalities (15) and (16);
 - 8: Return S_l ;
-

After the feasible scheme set S_l for each layer l has been obtained, the optimal rate control procedure for the optimization problem (4) can be easily solved using the proposed algorithm presented in Alg. 2. The proposed algorithm can be viewed as an extension of the resource allocation algorithm in [35]. The difference lies in that in the proposed algorithm, one scheme is selected at certain time by certain layer does not guarantee that it will not be removed from the solution, so this is not a true greedy algorithm. The similarity lies in the fact that if some bits have been assigned to certain layer, then the assigned bits will not be taken away from this layer, but used in another way. This greedy-like algorithm can also be viewed as alternative of the generalized Lagrange multiplier method [36], but are implemented in a more efficient way, where [37] has compared the difference between the greedy-like method and the generalized Lagrange multiplier method. By adjusting the length of the last layer to be transmitted, the constraint can be made tight enough.

Algorithm 2 Greedily-like JSCCRC algorithm

- 1: $R = 0$;
 - 2: **for** ($1 \leq l \leq L$) **do**
 - 3: Find S_l using Alg. 1;
 - 4: Let $s_l = s_l^0$ and mark s_l^0 ;
 - 5: **end for**
 - 6: **while** ($R < R^M$) **do**
 - 7: Find the layer l and the strategy s_l' such that $g_l(s_l, s_l')$ is maximized among all layers and all unmarked strategies in this layer ;
 - 8: $R = R + \Delta R_l(s_l, s_l')$;
 - 9: Let $s_l = s_l'$ and mark s_l' ;
 - 10: **end while**
 - 11: **if** ($R > R^M$) **then**
 - 12: Let l be the last layer with $s_l \neq s_l^0$, reduce the length of this layer from n_l to n_l' such that $R - R^M = (n_l - n_l')R_l(s_l)$;
 - 13: **end if**
 - 14: Return \bar{s} and $\{n_l\}$.
-

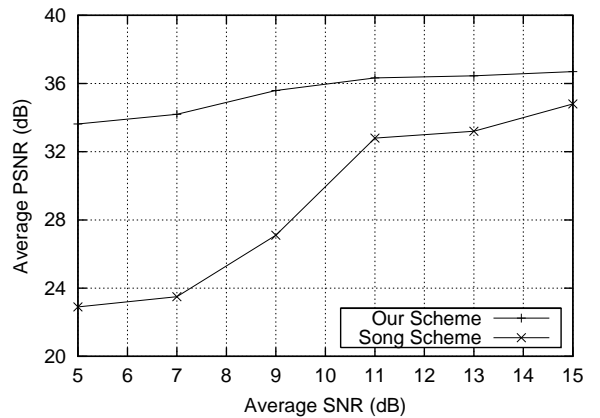


Fig. 8. Performance comparison of the proposed system with [28] for 2 Tx antennas and 2 Rx antennas

The computation complexity of the proposed algorithm is analyzed below. Assume the total number of layer is L , the total number available strategies is N , and the number of marked strategies is M . Let the time needed to calculate $G_l(s_l)$ and $R_l(s_l)$ be T_g . By using quick sort, the computation complexity of the feasible strategy set search for each layer becomes $O(NT_g + N \log_2 N + 2N)$, where NT_g comes from the gain and rate calculation, $N \log_2 N$ from the quick sort, and $2N$ comes from convex hull analysis. Then the overall computation complexity becomes $O(LNT_g + LN \log_2 N + 2NL + M \log_2 L)$. If $L = 10$, $N = 100$, $M = 50$, then the bound becomes $1000T_g + 10000$. The computation complexity can be further reduced if the layers have similar profiles.

VII. EXPERIMENTAL RESULTS

In this section we present the simulation results. In our simulations, we use a two-ray channel model with delay spread from $0\mu s$ to $20\mu s$. Two transmit antennas and one received antenna are used. The entire channel bandwidth is divided into 128 subchannels. The used variable rate SF code is an extension of the Blum's 4-state ST trellis code with QPSK modulation [6]. The reference image is gray 512×512 *lenna* with and 8-bit per pixel. The transmit rate used for all the simulations are 0.5bpp and 0.225bpp. The image is first compressed into 10 quality layers, for each layer, the packet headers are grouped together into layer header. The length of the main header is 153 bytes, and the length for each layer is about 1600 bytes. The main header can also be viewed as the first layer.

The parameters used to perform JSCC-based rate control center is as below: the RS codes are derived from RS(255, 223), and packet erasure codes are from [9], the rates of the SF code are 1, 1/2, 1/3, 1/4, and 1/5, and the available source error coding schemes for each layer are packet level synchronization marker, coding pass level synchronization marker, and RAW coding. The code block size can vary between 32×32 and 64×64 . The SNR varies from 1 to 15 in dB, and the packet dropping rate is 5% or 10%.

Fig. 8 illustrates the performance comparison between the proposed system and the system presented in [28] for 2

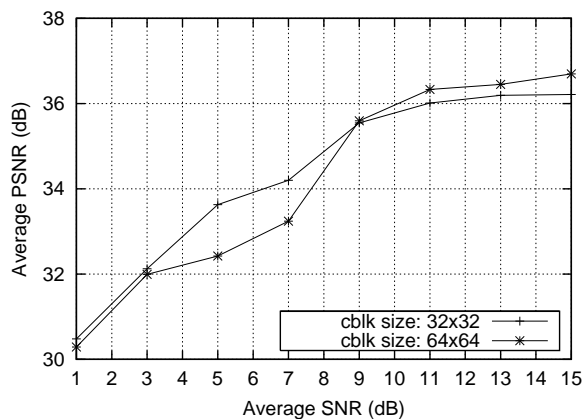


Fig. 9. Adaptivity to code block size

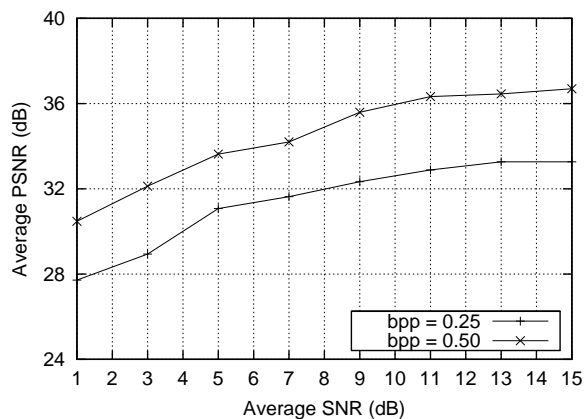


Fig. 11. Adaptivity to the rate constraint

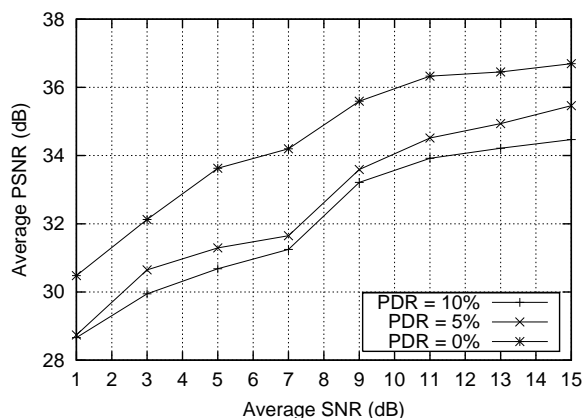


Fig. 10. Adaptivity to packet dropping

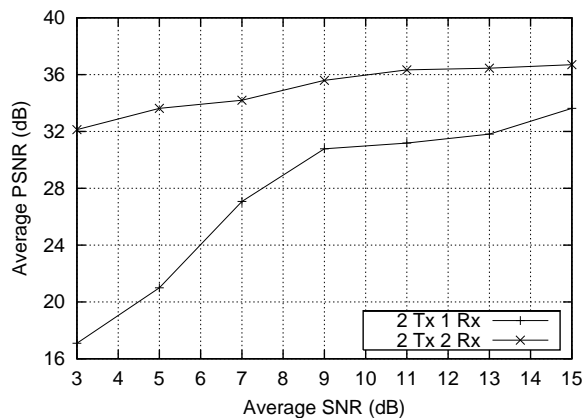


Fig. 12. Performance comparison for different system configuration

transmit antennas and 2 receive antennas with rate 0.5bps, and without packet dropping. In [28], space time block codes plus RS codes are used to protect the scalable bitstream generated using the SPIHT [38] encoder. From the results we can see that the proposed system has much better performance than the system in [28], which comes from several reasons. First, our system is a true JSCC system, which their system is a JSCM system, so no source level error resilience schemes have been exploited. Second, our system targets on minimizing the expected distortion, while their targets on maximizing the number of correctly received bits.

To show the adaptivity of the system, we compare the performance by fixing the code block size. Intuitively, when the BER is high, small code block size should be used, and when the BER is low, large code block size should be used. The simulation results presented in 9 also confirms the results. When the SNR ranges from 1 to 8, code block size 32×32 is preferred, when the SNR ranges from 10 to 15, code block size 64×64 is preferred, which illustrates the adaptivity of the proposed system.

Fig. 10 presents the simulation results for different packet dropping rate. From these results we can see that the system can still achieve good performance when the packet dropping rate is high, and the performance degrades smoothly with the

increase of the packet dropping rate.

Fig. 11 presents the performance comparison for two available rate constraint: 0.5bps and 0.25bps. The results show that increase rate constraint can also improve the performance.

Fig. 12 shows the performance comparison for different MIMO system configuration, when two types of system configurations are considered: MIMO system with 2 transmit antennas and 1 receive antenna vs. MIMO system with 2 transmit antennas and 2 receive antenna. The simulation results show that increasing the number of antennas can dramatically improve the performance.

VIII. CONCLUSION AND FUTURE WORK

In this paper, we have addressed the issues to design efficient multimedia transmission systems over error-prone channels, and proposed an efficient image transmission system over hybrid wireless network using JSCC, where the efficiency lies in three aspects: adaptivity, optimality, and low complexity. The proposed system are adaptive to the image content, the current channel conditions, and the resource constraint. Based on these information, the system can approach or approximate the optimal strategies by applying a low-complexity JSCCRC algorithm.

The proposed system is a *true* JSCC scheme. However, unlike the previous *true* JSCC schemes, in the proposed system, the source coding part is done in a more efficient way: we first quality scalable bitstream in a R-D optimal sense, then extra redundancy is added adaptively by applying certain error resilient coding schemes. The advantage of this separation lies in that it can reduce the encoding complexity, and the R-D optimal bitstream can be generated one time and used many times.

Another contribution of this paper is that we have presented an method to quantitatively analyze the performance of difference error resilience coding schemes.

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REFERENCES

- [1] W. C. Jakes, Ed., *Microwave Mobile Communications*. New York: John Wiley and Sons, 1974.
- [2] M. A. Labrador and S. Banerjee, "Packet dropping policies for atm and ip network," *IEEE Communications Surveys*, vol. 2, no. 3, pp. 2–14, Third Quarter 1999.
- [3] ISO/IEC, "ISO/IEC 15444-1:2000: Information technology — JPEG 2000 image coding system — part 1: core coding system," 2000.
- [4] D. Taubman, "High Performance Scalable Image Compression with EBCOT," *IEEE Transactions on Image Processing*, vol. 9, no. 7, pp. 1158–1170, 2000.
- [5] C. Christopoulos, A. Skodras, and T. Ebrahimi, "The JPEG2000 Still Image Coding System: An Overview," *IEEE Transactions on Consumer Electronics*, vol. 46, no. 4, pp. 1103–1127, November 2000.
- [6] Z. Safar and K. J. R. Liu, "Variable rate space-time trellis codes," in *IEEE Vehicular Technology Conference (VTC)*, vol. 1, September 2002, pp. 406–409.
- [7] W. Su, Z. Safar, M. Olfat, and K. J. R. Liu, "Obtaining full diversity space-frequency codes from space time codes via mapping," *Submitted for publication*, 2003.
- [8] S. Lin and D. Costello, *Error Control Coding*. Prentice-Hall, 1983.
- [9] L. Rizzo, "Effective erasure codes for reliable computer communication protocols," *ACM SIGCOMM Computer Communication Review*, vol. 27, no. 2, pp. 24–36, 1997.
- [10] I. E. Telatar, "Capacity of multi-antenna gaussian channels," *European Transactions on Telecommunications*, vol. 10, no. 6, pp. 585–595, Nov/Dec 1999.
- [11] G. Foschini and M. Gans, "On limits of wireless communications in a fading environment when using multiple antennas," *Wireless Personal Communications*, vol. 6, no. 3, pp. 311–335, 1998.
- [12] L. J. Cimini, "Analysis and simulation of a digital mobile channel using orthogonal frequency division multiplexing," *IEEE Transactions on Communications*, vol. 33, no. 7, pp. 665–675, July 1985.
- [13] Y. Li, L. J. Cimini, and N. R. Sollenberger, "Robust channel estimation for ofdm systems with rapid dispersive fading channels," *IEEE Transactions on Communications*, vol. 46, pp. 902–915, July 1998.
- [14] J. G. Proakis, *Digital Communications*. New York: McGraw Hill, 2001.
- [15] W. W. Peterson and E. J. Weldon, *Error-Correcting Codes*, 2nd ed. The MIT Press, 1972.
- [16] S. Alamouti, "A simple transmit diversity technique for wireless communications," *IEEE Journal on Selected Areas in Communications*, vol. 16, no. 8, pp. 1451–1458, October 1998.
- [17] V. Tarokh, N. Seshadri, and A. R. Calderbank, "Space-time codes for high data rate wireless communication: Performance criterion and code construction," *IEEE Transactions Information Theory*, vol. 44, no. 2, pp. 744–765, March 1998.
- [18] V. Tarokh, H. Jafarkhani, and A. R. Calderbank, "Space-time block codes from orthogonal designs," *IEEE Transactions on Information Theory*, vol. 45, no. 5, pp. 1456–1467, July 1999.
- [19] J. Hagenauer, "Rate-compatible punctured convolutional codes (rcpc codes) and their applications," *IEEE Transactions on Communications*, vol. 36, no. 4, pp. 389–400, April 1988.
- [20] C. E. Shannon, *Collected Papers*. New York: IEEE Press, 1993.
- [21] P. G. Sherwood and K. Zeger, "Progressive image coding for noisy channels," in *Proc. Data Compression Conf.*, 1997, pp. 72–81.
- [22] P. Sherwood and K. Zeger, "Error Protection for Progressive Image Transmission over Memoryless and Fading Channels," *IEEE Transactions on Communications*, vol. 46, no. 12, pp. 1555–1559, December 1998.
- [23] S. Appadwedula, D. Jones, K. Ramchandran, and I. Kozintsev, "Joint source channel matching for a wireless communication link," vol. 1, 1998, pp. 482–486.
- [24] J. Lu, A. Nostratinia, and B. Aazhang, "Progressive source channel coding for image over bursty error channels," in *IEEE Int. Conf. Image Progression*, vol. 2, 1998, pp. 127–131.
- [25] V. Chandre and N. Farvardin, "Joint source-channel coding for progressive transmission of embedded source coders," in *Proc. Data Compression Conf.*, 1999, pp. 52–61.
- [26] J. Hagenauer and T. Stockhammer, "Channel coding and transmission aspects for wireless multimedia," *Proceedings of the IEEE*, vol. 87, no. 10, pp. 1764–1777, October 1999.
- [27] H. Zheng and K. J. R. Liu, "The subband modulation: A joint power and rate allocation framework for subband image and video transmission," *IEEE Trans. Circuits Syst. Video Technol.*, vol. 9, p. 823838, Aug. 1999.
- [28] J. Song and K. J. R. Liu, "Robust progressive image transmission over ofdm systems using space-time block code," *IEEE Transactions on Multimedia*, vol. 4, no. 3, September 2002.
- [29] N. Farvardin, "A study of vector quantization for noisy channels," *IEEE Trans. Inform. Theory*, vol. 36, p. 799809, July 1990.
- [30] P. Cosman, J. Rogers, P. Sherwood, and K. Zeger, "Combined Forward Error Control and Packetization Zerotree Wavelet Encoding for Transmission of Image over Varying Channels," *IEEE Transactions on Image Processing*, vol. 9, no. 6, pp. 982–993, June 2000.
- [31] Y. Eisenberg, C. E. Luna, T. N. Pappas, R. Berry, and A. K. Katsgengel, "Joint source coding and transmission power management for energy efficiency wireless video communications," *IEEE Transactions on Circuits and Systems for Video Technology*, vol. 12, no. 6, pp. 411–424, June 2002.
- [32] Q. Zhang, Z. Ji, W. Zhu, and Y.-Q. Zhang, "Power-minimized bit allocation for video communication over wireless channels," *IEEE Transactions on Circuits and Systems for Video Technology*, vol. 12, no. 6, pp. 398–410, June 2002.
- [33] M. Antonini, M. Barlaud, P. Mathieu, and I. Daubechies, "Image Coding Using Wavelet Transform," *IEEE Transactions on Image Processing*, vol. 1, no. 2, pp. 205–220, 1992.
- [34] T.-C. Yang, S. Kumar, Y. Bao, and C.-C. J. Kuo, "Robust Coding of Images using EBCOT and RVLC," in *Proceedings of the SPIE's 44th annual meeting, Applications of Digital Image Processing XXII*, vol. 3808, 1999, pp. 395–400.
- [35] B. Fox, "Discrete optimization via marginal analysis," *Management Science*, vol. 13, no. 3, pp. 210–216, Nov. 1966.
- [36] H. Everett, "Generalized Lagrange Multiplier Method for Solving Problems of Optimum Allocation of Resources," *Operations Research*, vol. 11, pp. 399–417, 1963.
- [37] W. Yu, F. Sun, and J. Fritts, "Joint Rate Control and Entropy Coding in JPEG2000," *submitted for publication*, 2003.
- [38] A. Said and W. Pearlman, "A New, Fast, and Efficient Image Codec Based on Set Partitioning in Hierarchical Trees," *IEEE Transactions on Circuits and Systems for Video Technology*, vol. 6, no. 3, pp. 243–250, June 1996.