ADAPTIVE-RATE IMAGE COMPRESSION FOR WIRELESS DIGITAL DATA TRANSMISSION SYSTEMS

J. E. Kleider and G. P. Abousleman
Motorola, Space and Systems Technology Group
Scottsdale, Arizona, USA
p22319, p26994@email.mot.com

ABSTRACT

The vast amount of data needed to represent digital imagery motivates the use of advanced compression systems to reduce the bandwidth required to transmit high-resolution source imagery. We propose two methods to provide optimal image quality at a fixed image delivery rate. The first method, channel-controlled variable-rate (CCVR) image coding, operates within the constraint that the modulation symbol rate is fixed. The second method, adaptive-rate coding-modulation (ARCM), utilizes adaptive modulation, and is less complex, while providing increased performance. Both methods use a variable-compression-ratio image coder and variable-rate channel coding. The objective is to maximize the quality of the reconstructed image at the receiver when transmitted through Rayleigh fading and additive white Gaussian noise (AWGN). The reconstructed image quality is maximized through adaptive source-channel coding in the context of a bit rate tradeoff between the source and channel coders (CCVR), and then, in addition, adaptive-rate modulation (ARCM). Both methods require knowledge of the channel state which is used by the receiver to inform the transmitter, via a feedback channel, of the optimal strategy for image compression, channel coding, and modulation format. The resulting system using ARCM achieves up to a 17 dB improvement over the peak signal-to-noise ratio (PSNR) performance of a system using a fixed-compression-ratio image coder and fixed-rate channel coding.

1. INTRODUCTION

The benefits of adaptive-rate modulation (ARM) were established with digital multi-media communication systems in [4], and for digital voice communication systems in [3]. The basic idea from [3] is derived from a combination of the principles of channel capacity and rate-distortion theory for lossy voice coding. A lower rate of source encoding for a vocoder causes an increase in the speech distortion, and is a function of the rate-distortion characteristics of the vocoder. The speech distortion is minimized by jointly minimizing the distortion in the vocoder due to lower encoding rates and to channel bit errors. Using this technique, a significant gain in speech quality can be achieved using ARCM and CCVR (as compared to fixed-source- and fixed-channel-coding rates).

This paper proposes to apply this technique to variable-compression-ratio image coding. ARCM is shown to be desirable when considering performance and complexity trade-offs, while the CCVR system offers performance gains with additional complexity. Both methods produce vastly improved reconstructed image quality for noisy channels exhibiting a large variance in bit error rate (BER). Early joint source-channel coding schemes suffer from designs which are highly channel dependent. The method proposed in [2] was shown to provide robust image quality to channel variation, but is limited to finite-state channels, and was shown only for source coders using vector quantization. In the interest of minimizing complexity and providing optimal PSNR performance at any BER, we present a method which maximizes source quality at the receiver by using source coding, channel coding, and modulation adjustments. CCVR, as addressed in [6], will perform a bit rate trade-off between the image coder rate-distortion performance, and forward error correction (FEC) utilization, based on the state of the channel. However, a better approach is ARCM, which uses adaptive modulation. Although not a constraint, utilization of fixed-rate channel coding reduces the complexity of ARCM.

PSNR is maximized by using a fixed-rate wavelet/TCQ image coder operated at multiple compression ratios by adapting the system in an optimal fashion to the noisy communication channel. The channel model consists of both AWGN and Rayleigh fading. The CCVR method produces improved quality through 4 bit rate tradeoffs between the source and channel coding. In general, the ARCM system produces improved image quality through the optimal tradeoff between source and channel coding, and modulation adjustments. A reduction in BER is obtained through the use of adaptive modulation, and translates into higher average PSNR. Adaptive modulation used in conjunction with channel coding and a multiple-compression-ratio image coder produces a large increase in the reconstructed image quality at the receiver.

The adaptive-rate modulation approach is shown in Figure 1. CCVR is similar, but is limited to fixed symbol rates. Reconstructed image quality is evaluated through both quantitative and subjective measures using peak signal-to-noise ratio and visual analysis, respectively. Quality improvements of both the CCVR and ARCM techniques are compared to a system using a fixed compression ratio of 1/2 bits per pixel, and 1/2-rate channel coding. All systems assume a fixed image delivery rate of one image per second and use maximal-distance-property convolutional codes for error protection. Simulation results show that both methods provide up to a 17 dB improvement in PSNR over the fixed system. Additionally, ARCM provides up to 2.5 dB improvement in PSNR over CCVR in AWGN. For the Rayleigh channel, both systems provide improvements with respect to the fixed system.

Section 2 describes each block of Figure 1. Experimental results are presented in Section 3, with conclusions given in Section 4.

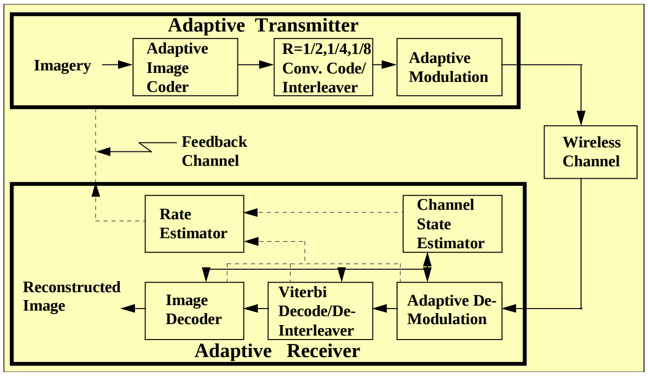

Figure 1: Block diagram of adaptive-rate image compression system.

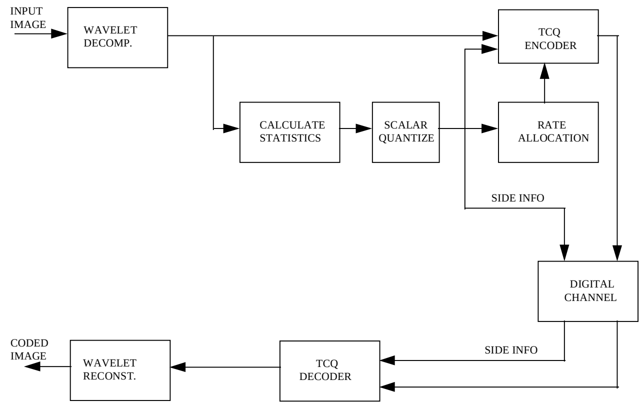

Figure 2: Wavelet image coder

2. SYSTEM DESCRIPTION

This section provides a detailed description of the ARCM system. As mentioned previously, the CCVR system is similar, with the inclusion of a fixed-modulation constraint.

2.1 Modulation Format. The ARCM technique is advantageous because it is applicable to both bandwidth efficient (MPSK or MQAM) and power efficient (MFSK) modems. In this work, we chose differentially encoded/decoded coherent 4PSK (D4PSK) because of its simplicity and popularity for mobile wireless applications. Additionally, a phase reference is not required at the receiver. For D4PSK, N=2 data bits are gray encoded onto the set of basis functions consisting of {(a + ja), (-a + ja), (-a - ja), (a - ja)}, where $a = \sqrt{2}/2$. The probability of a bit error for D4PSK can be found in [5].

For the Rayleigh channel, we assume a "quasi-static" system. The Rayleigh fading performance of D4PSK can also be found in [5].

2.2 Channel Model. A fully interleaved Rayleigh fading channel with complex additive white Gaussian is used for all simulations. The channel can be defined such that the i^{th} received symbol, $z(i)$, is related to the i^{th} transmitted symbol, $x(i)$, according to the relation:

$$z(i) = R(i)x(i) + n(i), \qquad (1)$$

where $x(i)$ is the transmitter output after differential encoding of the gray-encoded 4PSK basis signal set, and $n(i)$ is a zero-mean complex Gaussian random variable with variance $N_0/2$. The multiplicative fading coefficient, $R(i)$, is Rayleigh distributed with $\mathbf{E}[R^2] = 1$. We assume $R(k)$ and $R(i)$ are independent for $k \neq i$, resulting in a "fully interleaved" Rayleigh channel. $R(i)$ is assumed to be approximately constant over the symbol duration.

2.3 Image Coder Description. Figure 2 shows a block diagram of the image coder utilized in the adaptive-rate system. The coder is similar to the intraband coder presented in [1], and is based on wavelet decomposition and trellis-coded quantization (TCQ). The image to be encoded is decomposed into 22 subbands using a modified Mallot tree configuration. That is, the image is initially transformed into 16 equal-sized subbands, with two additional levels of transformation being performed on the lowest-frequency subband. Each subband is treated separately as a sequence and quantized using fixed-rate trellis-coded quantization (FRTCQ). All subbands are normalized by subtracting their mean (only the sequence corresponding to the lowest-frequency subband is assumed to have non-zero mean) and dividing by their respective standard deviations.

The total side information to be transmitted consists of the mean of the lowest-frequency subband, and the standard deviations of all 22 subbands. These quantities are quantized using 16-bit uniform scalar quantizers with a total of 368 bits. These quantities are assumed to be transmitted with no channel errors. Additionally, the initial trellis state for each encoded sequence requires 2 bits (for a 4-state trellis). The total side information then consists of 412 bits per image.

Fixed-rate TCQ codebooks are designed in one-bit increments from 1 to 8 bits/sample. The training sequence consisted of 100,000 samples derived from a Laplacian pseudo random number generator. Codebook design uses a modified version of the generalized Lloyd algorithm for vector quantizer design.

Rate allocation is performed by using an iterative technique which allocates bits based on the rate-distortion performance of the various trellis-based quantizers, and the energy content of the wavelet coefficients. This rate allocation procedure allows precise bit rate specification, independent of the image to be coded.

2.4 Channel Coding and Interleaver. Multiple standard convolutional coders are used to produce possible candidate codes for the variable channel coding. The channel coding is applied with equal weighting to the source data bits. The error protection can be applied at rates 1/2, 1/4, and 1/8, with K=7, as defined in [5]. Viterbi decoding is used with an optimal differentially detected correlation receiver to decode the received data.

2.5 Rate and System State Estimators. The receiver performs near-optimal demodulation by sampling the output of the system state estimator. The purpose of the system state estimator is to decode the current image coding, channel coding, and modulation strategy from the transmitter. The function of the rate estimator is to monitor the bit error and symbol error statistics, and the reconstructed image PSNR. These statistics are used by the rate estimator to monitor the channel's short- and long-term statistics, and determine the new system operating parameters.

2.6 Switching Algorithm. The transmitter decodes the information on the feedback channel to determine the new image coding, channel coding, and modulation strategy. Confirmation of the new strategy is sent from the receiver. For the purposes of this work, the switching strategy for the modem is straightforward and is performed on an image-by-image basis. Switching within an image is also possible, but it not considered here. For our purposes, the channel is considered to be approximately constant within each image transmission.

3. EXPERIMENTAL RESULTS

Typically, the BER performance analysis of a digital communication system is measured versus the energy per bit divided by the noise spectral density (E_b/N_0). This parameter, however, does not reflect the performance increase achieved by an adaptive-rate modem. In the adaptive-rate system, performance comparisons can be illustrated by using the relationship between the received average carrier power, C, and the energy per bit times the bit rate, R_b, or $C = E_bR_b$. Dividing each side by N_0 gives $C/N_0 = (E_b/N_0)R_b$, where C/N_0 can be used as the parameter to provide image coder PSNR performance comparisons.

The maximum bit rate of the system is assumed to support transmission of a 512 by 512 pixel image, compressed at approximately 1 bit per pixel (≈262 kbits/sec). When channel coding is used, $C/N_0 = (E_{cb}/N_0)R_{cb}$, where R_{cb} is the coded bit rate, and E_{cb} is the energy per coded bit. R_{cb} is equal to the sum of the image coder bit rate and channel coder bit rate, respectively, or $R_{cb} = R_i + R_c$. Because D4PSK is used, the symbol rate, R_s, is determined by $R_s = R_{cb}/2$ with channel coding. The designs for the two systems are shown in Tables 1 and 2 for both the AWGN and Rayleigh fading channels. Listed are the image coder bit rate, I_r, and the channel code rate, r, for each strategy. Note that $r = k/n$, where k equals the number of bits into the channel coder, and n equals the number of bits out of the channel coder.

Figures 3 and 4 are plots of the average PSNR of the reconstructed image. From Tables 1 and 2, it is evident that strategies 1 and 5 are equivalent, as are strategies 2 and 6.

Table 1. Fixed-symbol-rate CCVR System.

Strategy	*I_r (bits/pixel)	r (k/n)	Modulation Rate
1	1	1	R_s
2	1/2	1/2	R_s
3	1/4	1/4	R_s
4	1/8	1/8	R_s

Table 2. Adaptive-rate coding-modulation system (ARCM).

Strategy	*I_r (bits/pixel)	r (k/n)	Modulation Rate
5	1	1	R_s
6	1/2	1/2	R_s
7	1/4	1/2	$R_s/2$
8	1/8	1/2	$R_s/4$
9	1/8	1/4	$R_s/2$

* Note: I_r is very close to the values listed in Tables 1 and 2 for each strategy.

For the fixed system, the source was coded at 1/2 bits/pixel, and 1/2-rate channel coding was employed, which is the same as strategies 2 and 6.

Results show that both the CCVR and ARCM systems require approximately 6 dB less received signal power to achieve the same PSNR performance as the approach with a fixed compression ratio. The system with adaptive modulation produces better image quality at low C/N_0 (< 54 dB-Hz) than the CCVR system. Results for Rayleigh fading are also shown in Figure 5 for the adaptive modulation method, and requires approximately 7.5 dB less received signal power to provide the same PSNR as the fixed system.

Modem rate changes are straightforward in terms of complexity. Thus, it is usually more desirable to choose an adaptive modulation approach as opposed to the computational complexity involved with channel code rate reductions (rate 1/2 to 1/8), as is done with the fixed modulation system of Table 1.

The objective of the switching strategy is to maximize the average PSNR of the reconstructed image. By utilizing the principles of channel capacity and rate-distortion theory, the optimal trade-off between the bit-error-induced distortion, and distortion due to lower source coding rates is obtained. Table 3 shows the performance of the adaptive-modulation system, which produces the best overall performance in AWGN. We believe that further gains are possible for lossy source coders exhibiting less sensitivity to bit errors. To illustrate the subjective improvement in image quality, Figures 6 and 7 show the reconstructed image for strategies 6 and 8, respectively, at 56 dB-Hz in AWGN.

Table 3. Adaptive-rate system performance versus strategy.

Strategy	C/N_o (dB-Hz) [AWGN] from	to
8	$-\infty$	56.5
7	56.5	58.5
6	58.5	64.4
5	64.4	$+\infty$

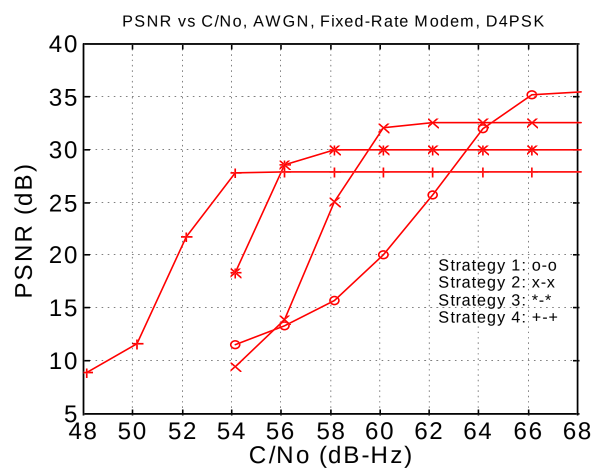

Figure 3: CCVR system performance in AWGN.

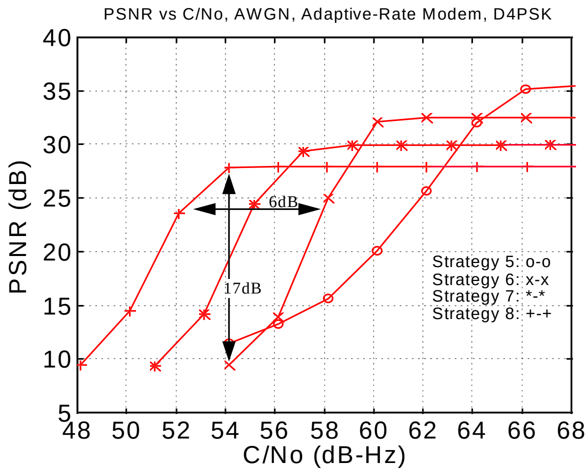

Figure 4: Adaptive-rate modem system performance in AWGN.

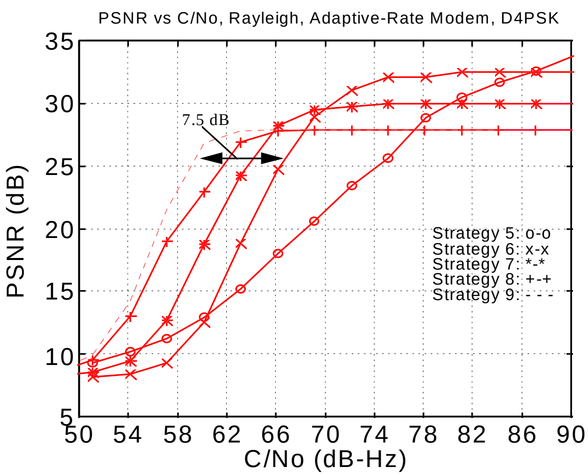

Figure 5: Adaptive-rate modem system performance in fading.

Figure 6: Reconstructed image, strategy 6, AWGN, at 56 dB-Hz.

Figure 7: Reconstructed image, strategy 8, AWGN, at 56 dB-Hz.

4. CONCLUSION

We have presented an adaptive-rate image compression system which chooses the optimal compression ratio, and rate of channel coding and modulation, for operation over noisy channels. The system combines the principles of channel capacity and rate-distortion theory to maximize the reconstructed image quality for any given channel condition. The system selects the optimal trade-off between distortion induced by bit errors, and distortion resulting from lower source coding rates. As shown through subjective and quantitative measures, a significant gain in image quality at the receiver is produced.

5. REFERENCES

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