Journal of Multidisciplinary Engineering Science and Technology (JMEST)
ISSN: 3159-0040
Vol. 1 Issue 5, December - 2014

Near-Capacity Joint Source-Channel Coding for Adaptive Video Transmission over Rayleigh Fading Channel

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***Abstract*—In this paper we propose an adaptive Joint Source and Channel Coding (JSCC) associated with adequate modulation algorithm to provide a near channel capacity video transmission rate to achieve better system performance. In the proposed algorithm we apply MPEG-2 video coding scheme, and turbo channel coding technique as well as BPSK/QPSK for modulation method. Rates assigned to MPEG-2 source coding and turbo channel coding schemes are based on the feedback information from Performance Control Unit (PCU) based on channel capacity limitation. Therefore, an improved performance for the suggest system design is promised over Rayleigh fading channel conditions.**

***Keywords—Joint Source-Channel Coding* (*JSCC*)*; MPEG2; Turbo Code; Markov State; Performance Control Unit* (*PCU*)**

I. INTRODUCTION

Near-capacity transmission is achievable while taking on individual source and channel coding algorithms, indicated in the source-channel separation theorem [1]. Near-capacity transmission could be accomplished by a near-entropy source code, such as an adaptive arithmetic code [2], comprised with a near-capacity channel code, e.g. a turbo code [3] or a Low Density Parity Check (LDPC) code [4]. Though, the source-channel separation theorem is dependent on numerous assumptions, which may not be valid in practice [5], [17]. In fact, the separately designed source coding and channel coding schemes have been tried to achieve the best system performance autonomously, however, by simply combining the optimum source coding scheme with the best channel coding scheme together will not promise to gain the best overall system performance. Therefore, an adaptive designed jointly source-channel coding (JSCC) scheme with adequate modulations is proposed to get the possibly best near-capacity performance, especially at varying channel conditions.

Under appropriate alliance of various source coding rates with analogous channel coding rates, an adequate designed jointly source-channel coding (JSCC) scheme may improve the overall system performance (Fig. 1). Source coding is concerned with the efficient representation of a signal. While bit errors in the uncompressed signal can cause minimal distortion, in its compressed format a single bit error can lead to meaningfully large errors. Thus, an associated channel coding is necessary to overcome the errors resulted from transmission channel. A jointly design source coding with adequate channel coding, we should be able to achieve an improved system performance. Assuming that the overall system transmission rate $r = k/n$, where k is the source coding rate and n is the channel coding rate. As shown in Fig. 1, a better system performance (with a lower distortion) can be assured when we increase the source coding rate k. However, if we raise the channel coding rate n, a higher bit error rate (a lower system performance) could happen under the same E_b/N_0 (the signal-to-noise ratio, *SNR*) criterion. Therefore, we would like to design a transmission system with higher source coding rate k but lower channel coding rate n to achieve a higher overall transmission rate r. Since the overall transmission rate r is restricted to under channel capacity, $r \leq C$, we have to vindicate the concept with proper system designing algorithm.

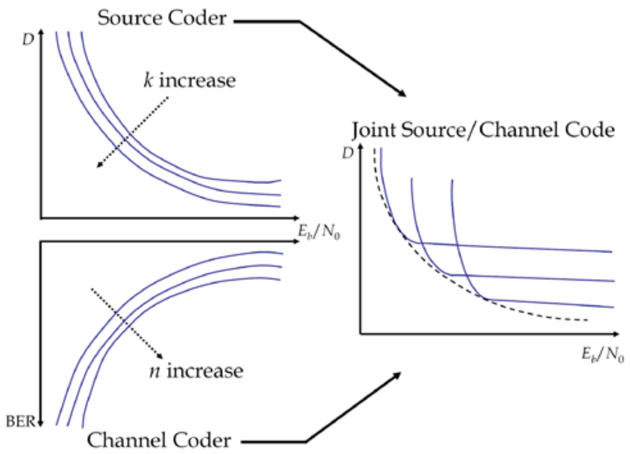

Fig. 1. Joint source and channel coding with different rates. [6]

System performance is studied for a joint MPEG-2 source coding and turbo channel coding associated with applicable modulation method (BPSK or QPSK), for video data transmission over a slow Rayleigh fading channel. The channel is modeled as an additive white Gaussian noise (AWGN) with Rayleigh fading noise. Rates assigned to MPEG-2 source coding and turbo channel coding schemes are based on the feedback information from Performance Control Unit (PCU) under system channel capacity limitation, which ensures the given system achieved an improved performance compared to conventional designed systems. Block diagram of the overall system is shown in Fig. 2.

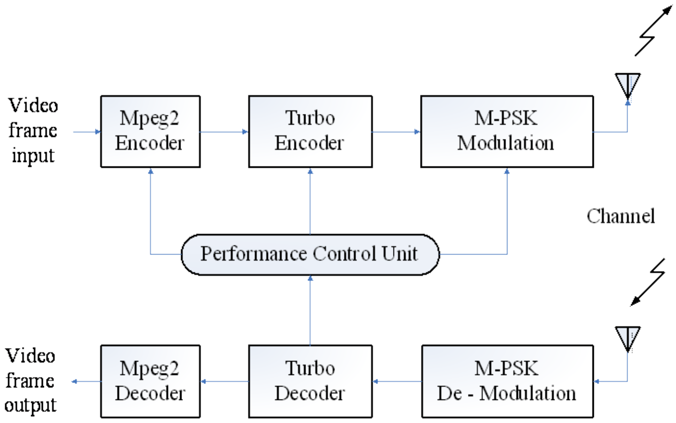

Fig. 2. Block diagram of the adaptive video transmission system.

The overall transmission rate *r* is obtained by source coding rate *k* cooperated with channel coding rate *n*. System performance for the restored video quality will not be appreciated if a high source compression rate (lower *k*) with strong channel protection (higher *n*) applied. In turns, if a low source compression ratio (higher source coding rate *k* with low distortion) but a high channel coding rate *n* (weak channel protection capability) to the transmission system is applied, which may result in higher bit error rate (*BER*) performance, we may not agree with the received signal quality from the received high BER data. It is quite clear that we have to find a better affiliation between source coding rate and channel coding rate to assure an acceptable system performance.

One of the most significant criterions in designing a transmission system is the channel capacity limitation. The available channel capacity restricts the overall transmission rate *r*, which is the rate between source coding rate *k* and channel coding rate *n*. We have to consider source coding rate, channel coding rate, and the compatible modulation type all together synchronously to cope with the channel capacity limitation. A channel capacity limitation assumed to be 1 bit/transmission is proposed in this study. We are asked to keep overall transmission rate $r \leq 1$, which can be achieved only with $k \leq n$. In this study, we will keep our system rate design with $r \approx 1$ and $r \leq 1$, that is, $k \approx n$ and $k \leq n$ to obtain the best system performance.

In this study, we have employed MPEG-2 source coding scheme and turbo channel coding technique, associated with suitable modulation method (BPSK or QPSK), for video data transmission over a wireless Rayleigh fading channel. When processing a given video data stream, we first apply the MPEG-2 scheme [7] to the input video frames. The MPEG-2 coded video is then transmitted over wireless channel. The transmission medium is modeled as a Rayleigh fading channel and BPSK/QPSK modulation is adopted in the study. Turbo channel coding algorithm is applied to improve the system transmission performance (higher *SNR*) while at high bit error rate (*BER*) fading channel condition. In this paper, Section II describes the system configuration proposed in this study. Experimental results for performance of the overall adaptive video transmission system compared with the conventional scheme over Rayleigh fading channel are shown in Section III. Finally, a summary and conclusions are presented in Section IV.

II. System Configuration

We are interested in the joint source-channel coding (JSCC) collaborated with modulation scheme design under the channel capacity constraint consideration. An integrated transmission system design method [8] is applied for digital transmission of video signals over noisy channels. Channel capacity is the supreme of all achievable transmission rates, where we have assumed in this study that the available channel capacity is 1 bit/transmission. We utilize MPEG-2 source code [6], [7], [16] with various compression rates to adjust the input test video coding rate. Before transmitting the MPEG coded video stream over the wireless channel, we consign a channel protection Turbo code to the input video steam. Turbo codes introduced by Berrou, et al. [3], [9] have been intensively studied as the channel error correction code for mobile radio communication. The applied Turbo code consists of two (or more than two) Recursive Systematic Convolutional (RSC) codes [9]-[11], with an interleaver between the two encoders (Fig. 3). The transfer function applied in this study is given as follows,

$$G(D) = \left[1, \frac{1 + D + D^2}{1 + D^2} \right]. \tag{1}$$

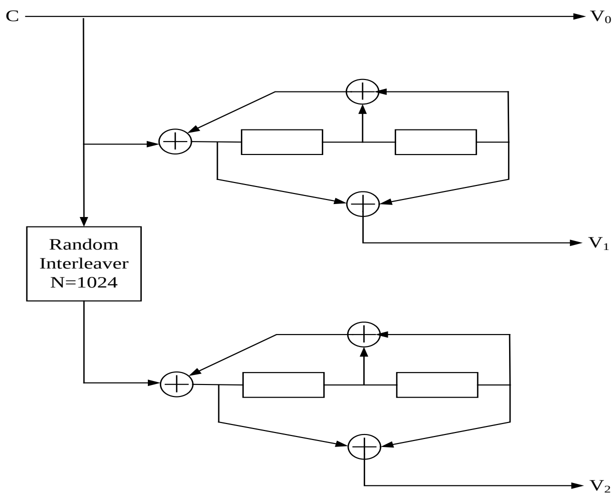

Fig. 3. Rate = 1/3 Turbo code structure (with two RSC's).

We have applied iterative SOVA algorithm [12] for the Turbo code decoding in this study. We have considered initially the first component SOVA decoder in the first iteration, where the decoder received channel sequence comprising the versions of the transmitted systematic bits, and the parity check bits, from the first encoder. The first SOVA decoder can process the soft channel inputs and produce its estimate of the "soft-output" data bits. The estimated soft-output data bits go through an interleaver and directed into the second component SOVA decoder as a-priori probability estimation. The second component SOVA decoder accepts the channel sequence containing the parity bits from the second encoder as well as the interleaved version of the received systematic bits. Along with the received channel sequence, the decoder can use the soft-output data bits delivered by the first component SOVA decoder to generate a-priori probability to be used by the second component decoder as shown in Fig. 4. In an iterative turbo decoder, the extrinsic information from the other component SOVA decoder is used as the a-priori information, after being interleaved to form the decoded data bits in the same order as they were encoded by the second encoder. The second component SOVA decoder accordingly uses the a-priori information with the received channel sequence to generate the a-posteriori information. This is to fulfil the first iteration. The estimated soft-output data bits are sent through an interleaver and fed into the first SOVA decoder as a-priori information, in which the first component decoder may process to decoder received channel sequence with additional a-priori information to produce the a-posteriori information, the soft-output data bits. This iteration process remains where on average the *BER* of the decoded bits after each iterating should be reduced. The iteration procedure conclusions once the prerequisite *BER* succeeded.

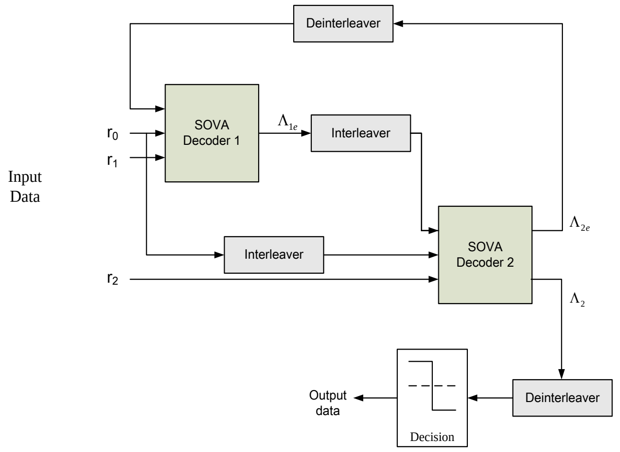

Fig. 4. Structure of a SOVA based Turbo code decoder.

The overall system configuration is shown in Fig. 1. We have applied MPEG-2 as the source video coding scheme and turbo channel coding to protect the transmitted data bits. With the joint selection of the source and channel code rates (listed in Table I, channel capacity constraint is 1 bit/transmission), we have associated each rate combination with an applicable modulation method (BPSK or QPSK), for video data transmission over a wireless Rayleigh fading channel. The channel is modeled as an additive white Gaussian noise (AWGN) with Rayleigh fading noise.

TABLE I. JOINT SOURCE/CHANNEL CODE RATE ASSIGNMENT UNDER CHANNEL CAPACITY = 1 BIT/TRANSMISSION.

State	Rate	Source Code	Channel Code	Modulation
A	0.9965 bit	0.3322 bpp	1/3	BPSK
B	0.9989 bit	0.4995 bpp	1/2	BPSK
C	0.9989 bit	0.6659 bpp	1/3	QPSK
D	0.9941 bit	0.9941 bpp	1/2	QPSK

Rates assigned to MPEG-2 source coding and turbo channel coding schemes are based on the feedback information from Performance Control Unit (PCU) under system channel capacity limitation, which ensures the given system achieved the best performance compared to conventional systems. PCU is the key components in the adaptive system design, where we have assign four states to report the variable overall transmission status from the wireless Fading channel conditions.

We adopt the first-order Markov chain to illustrate the varying channel conditions of the transmission system [8], [13]. The current state is only associated with the one-step preceding state but not all the other states. As shown in Fig. 5, State A can transfer to State B or remains in the same state, where State B may stay in State B, jump into State C, or switch back to State A, and all the other states work at the same manner. Shown in the figure, H is the output index of the PCU. We have assigned that, H = 0 is the stay-still index, system should keep in the same state; H = 1 is the jump forward index, the channel is in a good condition; H = –1 is the step backward index; system has to move back to the previous state since the channel condition has degraded.

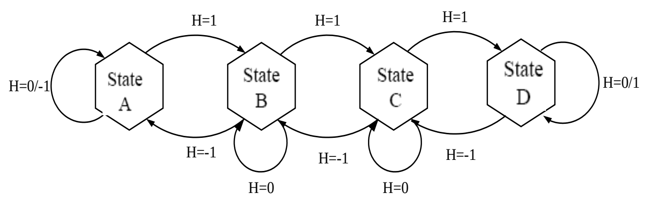

Fig. 5. System state transition

To simplify the simulation, we have set-up four states to collaborate with the variable Rayleigh fading channel conditions. In the simulation, we first send Command Testing Sequence (CTS), consisting of 10 bits stream of "1", which is attached in front of the transmitted data stream [6], [8]. After receiver been channel decoded the received data sequence, the *BER* information of the CTS is fed-back to the transmitter side as a status index, H, to adjust the next transmission status as shown in Table II.

TABLE II. PCU SYSTEM STATE ASSIGNMENT

State	Error rate of feedback testing sequence
H = 1	Error rate = 0%
H = 0	0% ≤ Error rate < 20 %
H = –1	20% ≤ Error rate

Under the channel capacity constraint, which we assumed in this study is 1 bit/transmission; we have proposed an adaptive video transmission system. Based upon the feedback information index from Performance Control Unit, we may adjust the source compression rate of MPEG-2 video coder associated with Turbo channel code rate to keep the overall transmission rate around the 1-bit channel capacity limitation. Since we have assigned four different channel condition states, there are 4 types of jointly rate assignments with corresponding modulation type as shown in Table I. According to the feedback error rate information from test sequence as shown in Table II, we can choose an adequate joint code rate assignment (system state, Fig. 5) to achieve best system performance.

III. Simulation Results and Analysis

Reconstructed video quality is measured by the peak output signal–to–noise ratio (*PSNR*, dB) defined by

$$PSNR(dB) = 10\log_{10}\left(\frac{x_{\max}^2}{MSE}\right) \tag{2}$$

where *MSE* (mean square error) is the distortion value for the reconstructed video data detracted from the original video data. Assuming the original video source is $x(i, j)$ with signal peak value x_{max} and the reconstructed video sequences is $\hat{x}(i,j)$, *MSE* between original video source and reconstructed video sequences is calculated as

$$MSE = E[(x(i,j) - \hat{x}(i,j))^2] \tag{3}$$

The *PSNR* value is employed as a subjective system performance evaluating along with the objective human's vision assessment.

For this proposed adaptive JSCC video transmission system, a transmitted video is applied the MPEG-2 coding scheme to yield a better data compression ratio and therefore to raise the system transmission rate. The MPEG-2 coded video is then applied a Turbo channel coding algorithm to transmit over wireless fading channel. The transmission medium is modeled as a Rayleigh slow fading channel and M-PSK modulation is adopted as shown in Fig. 2. For the proposed adaptive video transmission system, we have assigned four associations of the joint source-channel code rate closed to the preset channel capacity value: 1 bit/transmission ($r \approx 1$ bit, $r \leq 1$) associated with suitable M-PSK modulation scheme (as listed in Table I). The system simulation is performed as, first input 2 *Y *U *V image frames each to the MPEG-2 encoder, the resultant output after MPEG-2 encoder is a video stream (*.m2v) to be attuned with matched Turbo codes and proper modulation algorithm. The coding ratio and associated modulation type is given in Table I. Each transmission video frame will be added up a 10-bit Command Testing Sequence (CTS) before transmitted over wireless channel, consisting of Rayleigh fading noise and AWGN. After received the feedback error rate information from the Performance Control Unit (PCU), we may update the joint coding rate assignment based on indicator index, H, of PCU. The simulation transmitted video stream consisting of 20 *Y *U *V image frames in total, and every 2 image frames is added up a CTS sequence to monitor channel conditions. The resulted error rate of feedback testing sequence is served as an indicator to switch transmission rate assignment.

We can choose one of the four PCU states to assign the coding rate according to the *PSNR vs.* E_b/N_0 status shown on Fig. 6 for iteration = 1, or Fig. 7 for iteration = 5, respectively. Since we expect to have almost fixed transmission rate under channel capacity constraint, the PCU state selection may assure that the adaptive system performance is consistent. At each E_b/N_0 ratio, we have four *PSNR* values corresponding to four of the preset states (Fig. 5), the lowest *PSNR* is set to be State A while the highest *PSNR* is assigned State D.

After the states designated, we may track the PCU indicator to adjust the joint source-channel coding rate as well as the corresponding modulation. Fig.'s 8 and 9 show the system performance comparison between the PCU controlled adaptive system and the maximum *PSNR* obtained from Fig.'s 6 and 7, respectively. We found that system performance of the proposed adaptive system is not as good as the optimum *PSNR* especially at lower E_b/N_0, but getting close to the best *PSNR* at higher E_b/N_0 (within 0.8 dB). Since the total number of state change is only 10 in this simulation, a better system performance can be expected if the transmission data stream is long enough (with more states switched over the varying fading channel conditions). In Fig.'s 10 and 11, we have found the *PSNR* performance obtained by the proposed PCU controlled adaptive system is located within the *PSNR* curves accomplished by four different joint source-channel coding rate and modulation types, respectively. Especially in high E_b/N_0 conditions, the proposed adaptive system coherently achieves highest *PSNR* performance as expected.

Fig. 12 shows MPEG-2 source coding results at four compression rates compared with the original video frame. Fig. 13 illustrates the reconstructed video quality for the proposed adaptive JSCC system with Turbo decoding iteration = 1. The simulated Rayleigh fading channel condition E_b/N_0 is 11 dB and the resulted *PSNR* is 21.5812 dB at rate r = 0.99746 bits closed to the pre-set channel capacity limitation 1 bit/transmission, PCU states are (A A B B B C C C D D). Fig. 14 are the reconstructed video frames for the same simulation, except for the Turbo decoding iteration = 5 and Rayleigh fading channel condition E_b/N_0 = 9 dB. The resulted *PSNR* is 31.8544 dB at rate r = 0.99722 bits closed to the pre-set channel capacity limitation 1 bit/transmission. The feedback PCU states are (A B C B C C D D D D). It is noticeable that, with more decoding iterations, the proposed system can gain a better *PSNR* (31.8544 dB vs. 21.5812 dB) and visual quality even in a worse channel condition (E_b/N_0 = 9 dB vs. E_b/N_0 = 11 dB).

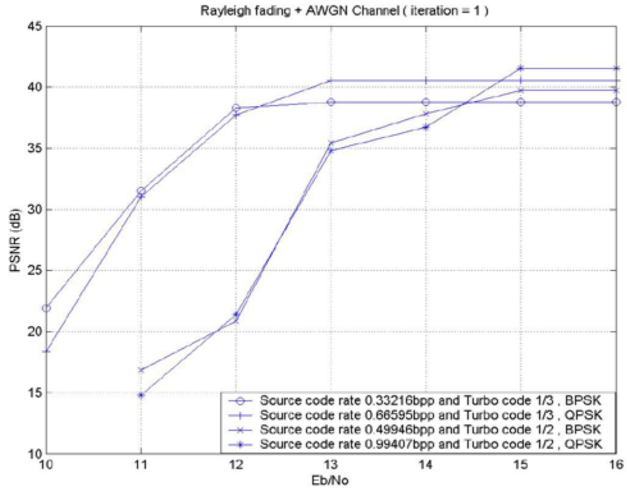

Fig. 6. *PSNR* performance for different joint source-channel coding rate and modulation type, iteration = 1.

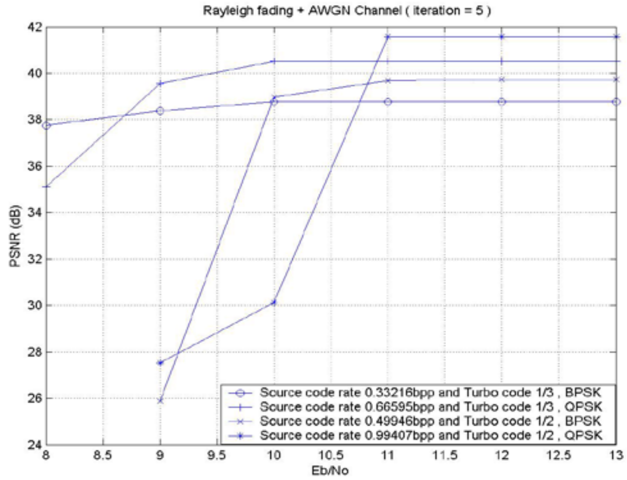

Fig. 7. *PSNR* performance for different joint source-channel coding rate and modulation type, iteration = 5.

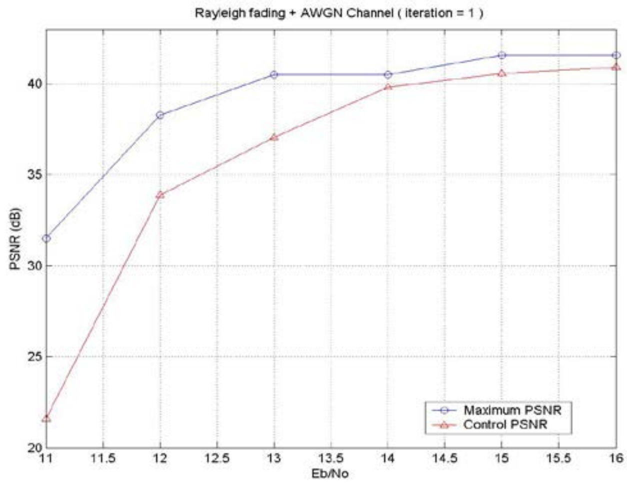

Fig. 8. Performance comparison for PCU controlled adaptive system with maximum *PSNR* joint source/channel code system, iteration = 1.

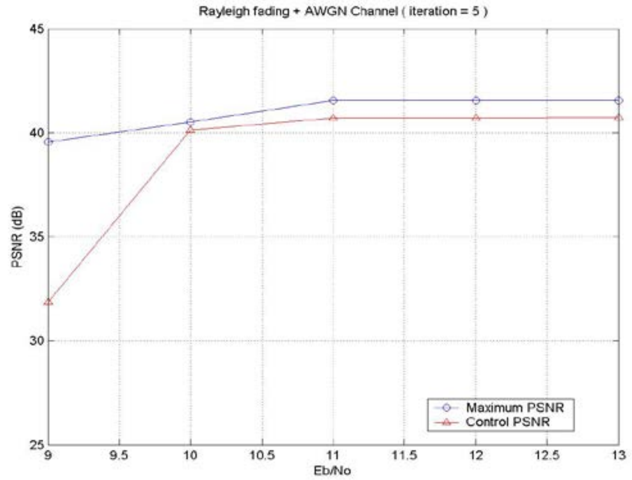

Fig. 9. Performance comparison for PCU controlled adaptive system with maximum *PSNR* joint source/channel code system, iteration = 5.

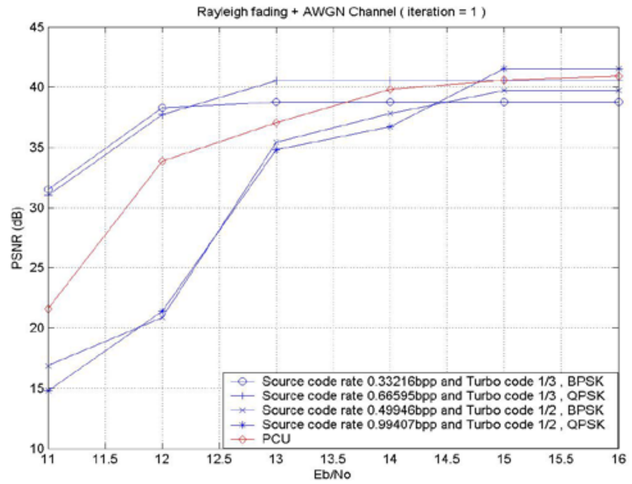

Fig. 10. *PSNR* performance comparison for PCU controlled adaptive system with four different joint source-channel coding rate and modulation types, iteration = 1.

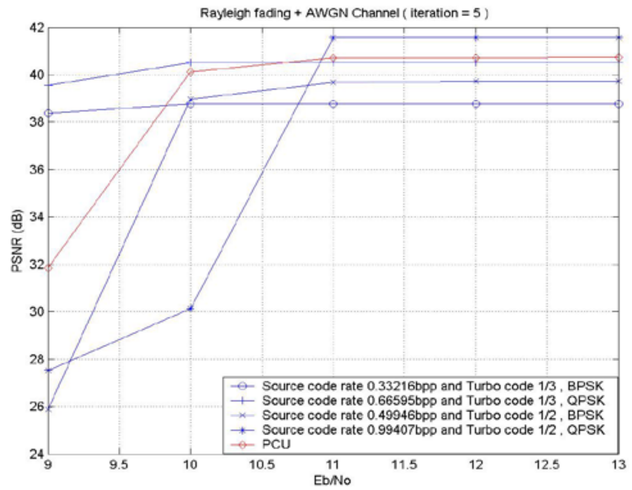

Fig. 11. *PSNR* performance comparison for PCU controlled adaptive system with four different joint source-channel coding rate and modulation types, iteration = 5.

IV. SUMMARY AND CONCLUSIONS

We have investigated an adaptive joint source-channel video transmission system to implement in wireless fading channels. We have applied MPEG-2 algorithm to code input video at rates associated with Turbo channel coding rates as well as coupled M-PSK modulation type to meet the pre-set system channel capacity requirement.

Four different combinations of source coding rate, channel coding rate, and modulation type as the system adaptive states are established. This arrangement assures overall data transmission rate closed to but less than the pre-set 1-bit channel capacity constraint. Simulation results show that we may gain a better reconstruct transmitted video quality at lower E_b/N_0 condition with increasing iteration number of Turbo decoding, therefore, to reduce power consumption for the overall system.

Under the channel capacity constraint, we are able to transmit video over Rayleigh fading channel with more consistent reconstructed quality. Based upon the feedback from Performance Control Unit, we may adjust the compression rate of MPEG-2 coder combined with Turbo channel code rate and modulation type to cope with the channel capacity limitation. System performance of the proposed adaptive system can achieve better *PSNR* at each E_b/N_0 condition compared to conventional systems.

Fig. 12. Four types MPEG-2 source coding results compared with original video frame.

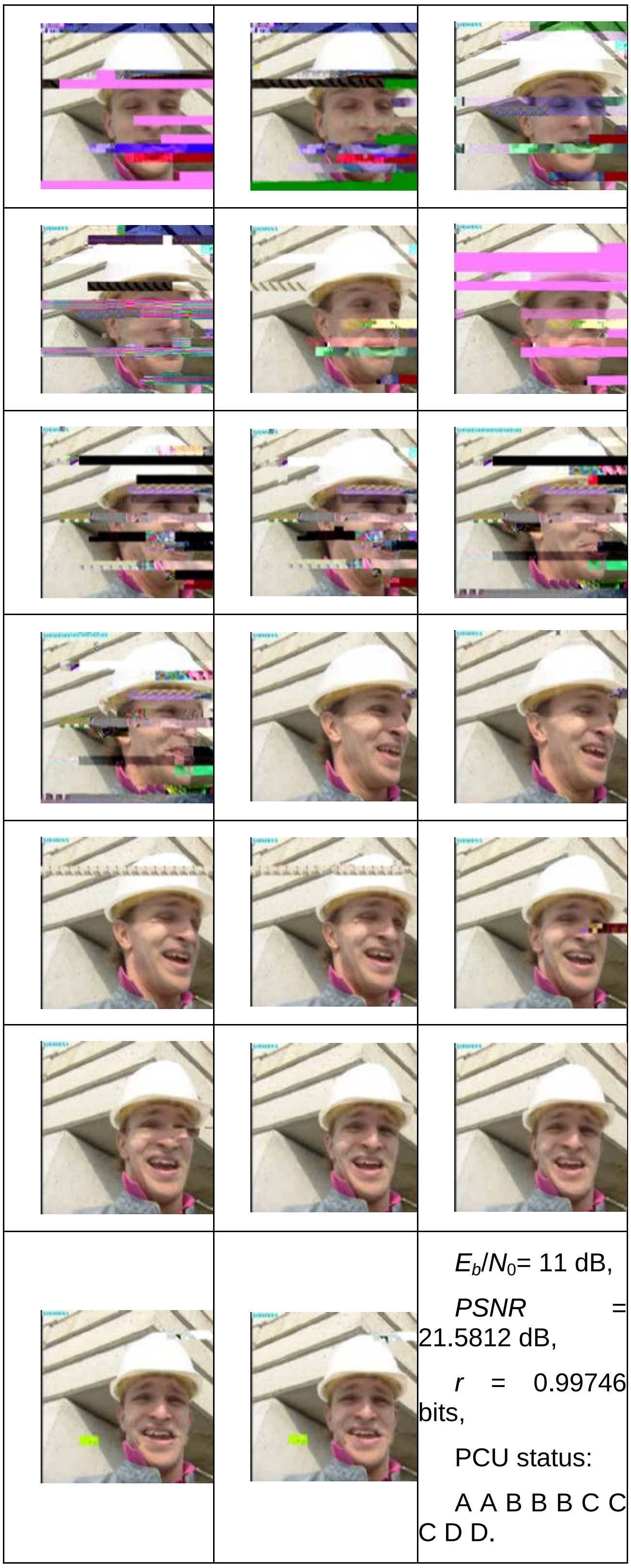

Fig. 13. Reconstructed video quality for PCU controlled adaptive system over Rayleigh fading channel, iteration = 1.

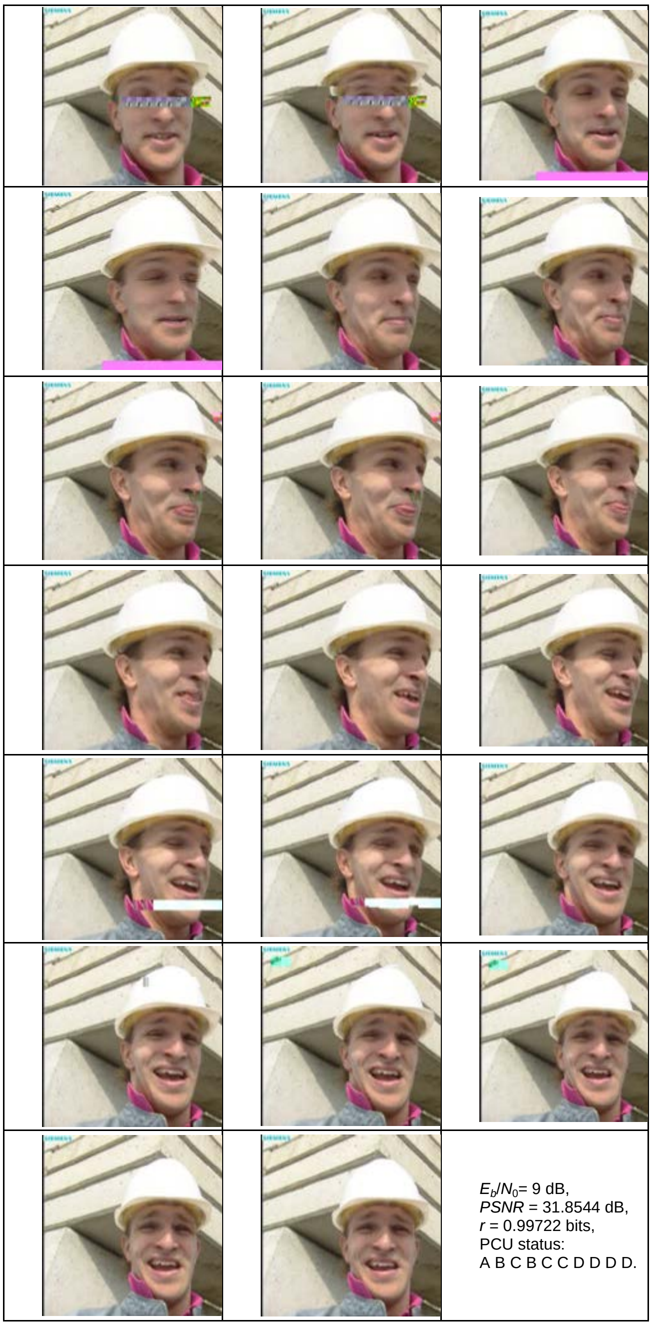

Fig. 14. Reconstructed video quality for PCU controlled adaptive system over Rayleigh fading channel, iteration = 5.

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