JOINT SOURCE-CHANNEL CODING AND POWER ALLOCATION FOR ENERGY EFFICIENT WIRELESS VIDEO COMMUNICATIONS

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ABSTRACT

We consider an energy-efficient video streaming system where source coding, channel coding, and transmission power allocation are jointly designed to compensate for channel errors. Our focus is on streaming applications with relatively strict delay constraints; for such applications, forward error correction (FEC) is the preferred channel coding technique to recover from packet losses. We propose a framework of joint source-channel coding and power allocation (JSCCPA), where resources such as bandwidth and transmission power are optimally allocated to provide unequal error protection (UEP) for achieving the best video quality. An efficient algorithm based on Lagrangian relaxation and dynamic programming is proposed to solve the constrained optimization problem. Simulation results illustrate the advantage of the proposed framework.

1. INTRODUCTION

A key problem with sending video over error-prone networks is recovering from channel errors. One solution to this problem is to add redundancy at the source coding level to prevent error propagation and limit the distortion caused by packet losses on the reconstructed video sequence. Such techniques are referred to as *error resilient* source coding techniques. However, these techniques cannot always overcome frequent packet losses. Another solution is to use channel coding either in the form of forward error correction (FEC) and/or retransmissions, to protect against packet loss. Such channel coding modifies the channel characteristics seen by the video encoder. Conventional retransmission-based schemes such as automatic repeat request (ARQ) are not used in most real-time streaming applications because they can not accommodate the delay requirements of the application. Likewise, such approaches may not be appropriate for multicast scenarios due to inherent scalability problems. For these reasons, FEC-based techniques have been widely examined in the literature [1–3], and are currently being considered by the Internet Engineering Task Force (IETF) as a proposed standard for supporting error resilience [4]. Since video packets are usually of different importance, the optimal bit allocation should vary across the packets, resulting in different packets receiving unequal error protection. The challenge is to add redundancy so that the receiver can optimally utilize it to detect and correct the errors and improve the video quality.

For video transmission over wireless networks, the efficient utilization of transmission energy is a critical design consideration [5]. Besides FEC, adjustment of transmitter power is another tool that can be used to change the characteristics of the wireless channel as seen by the video encoder. Thus, if the physical layer is accessible at the encoder, prioritized transmission can be achieved through the adjustment of the transmitter power for each packet [6].

In this work we jointly consider error resilient source coding, FEC and power allocation to provide UEP for real-time wireless video communication applications. Specifically, we consider how to optimally allocate bits between source and channel coding, together with the power allocation to achieve the best trade-off between video quality and resource allocation.

With regard to related work, joint error resilient source coding (quantization parameter and mode selection) and power allocation has been studied for energy efficient wireless video transmission in [6], assuming on a perfect channel coding mechanism. In [7], a joint source coding and power control approach is presented for optimally allocating source coding rate and bit energy normalized to the multiple-access interference noise density, E_b/N_0 in the context of 3G CDMA networks. The work in [7] did not address error resilient source coding and channel coding. Joint source-channel coding and power allocation has been studied in [8] for progressive image transmission. In this paper, we present a JSCCPA framework for real-time video transmission, where the error resilient source coding, FEC, and power allocation are all jointly designed. To tackle the optimization problem with two constraints, an efficient algorithm based on Lagrangian relaxation and dynamic programming is proposed.

The rest of this paper is organized as follows. We first present some background on network-adaptive video streaming systems and FEC in Sect. 2. In Sect. 3 the problem formulation of JSCCPA is presented. Section 4 presents the proposed solution algorithm. Experimental results are discussed in Sect. 5, and Sect. 6 contains our conclusions.

2. BACKGROUND

2.1. Video Transmission over IP-based Wireless Networks

In a video streaming system over an IP-based wireless network, video packets (referred to as *source packets*) are generated by a video encoder. In the application layer, parity packets used for FEC may also be generated if applicable. After passing through the network protocol stack (e.g. RTP/UDP/IP), *transport packets* are formed to be sent over the network. In the link layer, parity bits are added within packets to further protect against channel bit errors (e.g., CRC is used to provide error check). Then the resulting packets are transferred to the RLC (radio link control) layer, where they are segmented into smaller RLC frames. Finally, the RLC frames are delivered through the physical layer. IP-based wireless networks typically operate using a 32-bit Ethernet (802.2) CRC, and all packets failing that CRC check are rejected [9, 10]. Thus, some transport packets may be dropped in the network (due to congestion) or at the receiver (due to unrecoverable bit corruption). For streaming applications, packets arriving at the receiver later than the scheduled playback time are also discarded. Packets that reach the decoder in time are buffered. The video decoder reads video packets from the decoder buffer and displays the resulting video frames in real-time (i.e., the video is displayed continuously without interruption). Lost packets are concealed at the decoder.

One of the main services provided by the physical layer is the measurement of various quantities, such as physical-channel BER, transport-channel block error rate, and transport-channel bit rate. These measurements are reported to the application layer for channel state estimation. Transmission rate bounds for the streaming applications are usually obtained through a congestion controller, which can be a model-based scheme to estimate the channel throughput for UDP (User Data Protocol) packet (see [3] for details).

2.2. Channel Model

For IP-based wireless applications, channel errors are usually in three forms: packet loss, packet truncation, and bit error. Packet loss and truncation are usually due to network traffic and clock drift. Bit corruption is due to the noisy wireless channel [9]. This work focuses on the "last hop", i.e., the transmission between mobile devices and the base station, because it is likely to be the bottleneck of the whole video streaming system. Thus, we mainly study bit corruption. As discussed in [9, 10], the wireless channel can be viewed as a packet erasure channel at the IP level, as it is "seen" by the video streaming applications, since we assume that packets with errors are unavailable to the multimedia application. At the IP level, by assuming constant network delay, we model the network as a packet erasure channel, where the probability of packet loss, ϵ, is only due to unrecoverable bit errors.

Consider using uncoded Binary Phase Shift Keying (BPSK) modulation scheme over a flat Rayleigh fading channel plus an Additive White Gaussian Noise (AWGN) process. The Bit Error Rate (BER),

p_e, assuming ideal interleaving, can be expressed as

$$p_e = \frac{1}{2}\left(1 - \sqrt{\frac{\alpha E_b}{N_0 + \alpha E_b}}\right), \quad (1)$$

where E_b is the bit energy, N_0 the noise power spectrum density, and α the expected value of the square of a Rayleigh fading [11]. Usually, in wireless channels, video packets are protected by adding redundant bits within packets. A packet will be treated as lost if the corrupted bits in this packet cannot all be recovered. Depending on the specific channel code chosen, the details of the derivation of packet loss probability ϵ will be addressed in the next subsection.

2.3. Forward Error Correction (FEC)

The FEC method used depends on the requirement of the system and the nature of the channel. For wireless video transmission applications, FEC can usually be applied across packets (in the application or channel layer) and within packets (in the link layer) [12]. In inter-packet FEC, parity packets are usually generated in addition to source packets to perform cross-packet FEC, which is usually achieved by erasure codes. In the link layer, redundant bits are added within a packet to perform intra-packet prediction from bit errors. In this work, we focus on link layer FEC.

A popular family of codes used to perform link-layer FEC with variable code rates are rate-compatible punctured convolutional (RCPC) codes [12]. RCPC codes are adopted in the level 3 of H.223 and H.324 annex C (mobile multiplexer), as a part of the mobile version of H.324 [13]. In this work we consider using RCPC codes to perform link-layer FEC, but the proposed framework could easily be applied to other codes as well.

RCPC codes were first introduced in [14]. A family of RCPC codes is described by the mother code of rate $1/N$ and memory M with generator tap matrix of dimension N by $(M+1)$. Together with N, the puncturing period P determines the range of code rates as $R = P/(P+l)$ where l can vary between 1 and $(N-1)P$. The RCPC codes are punctured codes of the mother code with puncturing matrices of dimension $N \times P$. The decoding of convolutional codes is most commonly achieved using the Viterbi algorithm. The Viterbi upper bound for the bit error probability is

$$p_b \leq \frac{1}{P} \sum_{d=d_{free}}^{\infty} c_d p_d$$

where d_{free} is the free distance of the convolutional code, p_d is the probability that the wrong path at distance d is selected, and c_d is the number of paths at Hamming distance d from the all-zero path. d_{free} and c_d are parameters of the convolutional code, while p_d depends on the type of decoding (soft or hard) and the channel. The theoretical bounds of BER for RCPC codes can be found in [14, 15]. In this work, we use the simulated BER. The method for simulation can be found in [14, 15].

2.4. Packetization Scheme

In this section, we discuss the packetization schemes employed in the data link layer for generating transport packets.

As shown in Fig. 1, we assume the source bits in each transport packet corresponds to one GOB (group of blocks) [1] and every packet is independently decoded. By using a particular RCPC code with rate r_k, the length of the packet is $B_k = B_{s,k} + B_{c,k} = B_{s,k}/r_k$. Assuming independent bit errors (i.e., the additive noise and fading are each i.i.d. and independent of each other), the packet loss probability for the k-th packet can be calculated by

$$\rho_k = 1 - (1 - p_b)^{B_k}, \quad (2)$$

[1] Since in the H.263 standard [16], one row of MBs (macro-blocks) is called a GOB, we use GOB to denote one row of blocks in the following text.

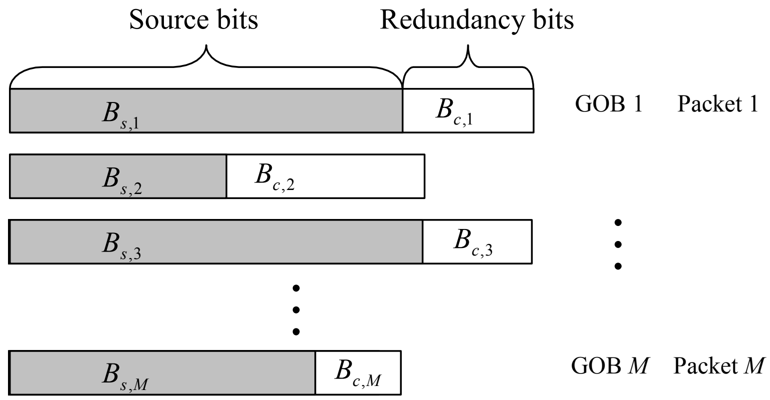

Fig. 1. Illustration of packetization.

where p_b is the BER after RCPC decoding. Note that the probability of packet loss ρ_k is a function of transmission power level, source coding parameter, and the channel coding rate selected for this packet.

3. PROBLEM FORMULATION

Different source coding parameters and different error protection ratios result in different levels of coding efficiency and robustness; the goal of joint source-channel coding (JSCC) is to find the optimal trade-off between coding efficiency and error robustness. In addressing this issue, an end-to-end approach has been employed, and three factors have been identified as affecting the video delivery quality at the receiver end: the source behavior (quantization and packetization), channel characteristics, and receiver behavior (error concealment) [1, 2, 6, 17, 18].

Besides FEC, the adjustment of transmitter power also effects the characteristics of the wireless channel as seen by the video encoder. Specifically, at a fixed transmission rate, increasing the transmission power can increase bit energy and consequently decrease BER, as shown in (1). On the other hand, at a fixed bit energy (keeping the same BER), increasing transmission power leads to higher transmission rate. Therefore, allocating different transmitter power level to the different packets results in different loss probabilities or delay for these packets. It is important to note that an increase in power may lead to increased interference to other users in the networks or inefficient use of the available battery energy. Thus, transmission power needs to be balanced against video delivery quality and delay. If the encoder can specify the transmission power (at the physical layer) for each transmission bit or packet, an interesting question is how to minimize the end-to-end distortion by optimally allocating bandwidth (bits) between source coding and channel coding, and optimally allocating energy (power) to each packet.

By jointly considering error resilient source coding, FEC, and power allocation, we formulate a JSCCPA problem given below,

$$
\begin{aligned}
\min_{\{\boldsymbol{\mu},\boldsymbol{\nu},\boldsymbol{\eta}\in\mathcal{Q}\times\mathcal{R}\times\mathcal{P}\}} & E[D] = \sum_{k=1}^{M} E[D_k(\boldsymbol{\mu},\boldsymbol{\nu},\boldsymbol{\eta})] \\
\text{s.t.} \quad & C = \sum_{k=1}^{M} B_k(\mu_k,\nu_k)P_k(\eta_k)/R_T \le C_0 \\
& T = \sum_{k=1}^{M} B_k(\mu_k,\nu_k)/R_T \le T_0,
\end{aligned} \tag{3}
$$

where $\mathcal{Q}$, $\mathcal{R}$, and $\mathcal{P}$ are the sets of allowable source coding parameters, channel coding parameters, and transmission power levels, respectively, and $\mu_k \in \mathcal{Q}$, $\nu_k \in \mathcal{R}$, and $\eta_k \in \mathcal{P}$ are the parameters selected for the k-th packet. The source bits and power level for the k-th packet are denoted by B_k and P_k, respectively. M is the number of source packets in one frame, R_T is the transmission rate, and C_0 and T_0

are the energy and transmission delay constraint for the frame, respectively. Let $\boldsymbol{\mu} = \{\mu_1, \mu_2, ..., \mu_M\}$, $\boldsymbol{\nu} = \{\nu_1, \nu_2, ..., \nu_M\}$, and $\boldsymbol{\eta} = \{\eta_1, \eta_2, ..., \eta_M\}$ denote the vector of source coding parameters, channel coding rates, and power levels for the M source packets in a frame, respectively. The expected distortion is calculated by

$$E[D_k] = (1 - \rho_k)E[D_{r,k}] + \rho_k E[D_{l,k}], \tag{4}$$

where $E[D_k]$ is the expected distortion for the k-th packet, $E[D_{r,k}]$ and $E[D_{l,k}]$ are the expected distortion when the packet is either received correctly or lost, respectively, and ρ_k is the loss probability of the k-th source packet. Note that both $D_{l,k}$ and $D_{r,k}$ are random variables. This is because, due to channel losses, the reference frames at the decoder and the encoder may not be the same. In using the packetization scheme discussed above, the packet loss probability ρ_k is given by (2) and is a function of μ_k, ν_k, η_k, as discussed in Sect. 2.4.

The calculation of $D_{l,k}$ depends on the specific error concealment strategy used at the decoder. If the lost MB is concealed using the mean motion vector of that of its received neighboring MBs (the top-left, top, and top-right), the expected distortion can be generally written as

$$E[D_k] = (1 - \rho_k)E[D_{r,k}] + \rho_k(1 - \rho_{k-1})E[D_{c,k}] + \rho_k\rho_{k-1}E[D_{z,k}], \tag{5}$$

where $E[D_{c,k}]$ and $E[D_{z,k}]$ are the expected distortions after concealment when the previous packet is either received correctly or lost, respectively. Assuming the mean squared error (MSE) criterion, the distortion measurement based on an algorithm called ROPE (Recursive Optimal Per-pixel Estimate) [17] can be used to recursively calculate the overall expected distortion level of each pixel.

4. SOLUTION ALGORITHM

In this section, we present a solution approach for (3) based on Lagrangian relaxation and deterministic dynamic programming (DP).

4.1. Lagrangian Relaxation

First, we formulate a Lagrangian dual for (3) by introducing Lagrange multipliers, $\lambda_1 \geq 0$ and $\lambda_2 \geq 0$, for the transmission energy and delay constraints, respectively. The resulting Lagrangian is

$$L(\boldsymbol{\mu}, \boldsymbol{\nu}, \boldsymbol{\eta}, \lambda_1, \lambda_2) = \sum_{k=1}^{M} \{E[D_k] + \lambda_1[B_k(\mu_k, \nu_k)P_k(\eta_k)/R_T - C_0] + \lambda_2[B_k(\mu_k, \nu_k)/R_T - T_0]\} \tag{6}$$

and the corresponding dual function is

$$g(\lambda_1, \lambda_2) = \min_{\{\boldsymbol{\mu}, \boldsymbol{\nu}, \boldsymbol{\eta} \in \mathcal{Q} \times \mathcal{R} \times \mathcal{P}\}} L(\boldsymbol{\mu}, \boldsymbol{\nu}, \boldsymbol{\eta}, \lambda_1, \lambda_2). \tag{7}$$

For simplicity, let $D(\cdot)$, $C(\cdot)$, and $T(\cdot)$ denote the expected distortion, transmission energy and transmission delay, respectively. The Lagrangian in (6) can be expressed as $L(\boldsymbol{\mu}, \boldsymbol{\nu}, \boldsymbol{\eta}, \lambda_1, \lambda_2) = \sum_{k=1}^{M} J(k)$, where

$$J(k) = D(k) + \lambda_1[C(k) - C_0] + \lambda_2[T(k) - T_0].$$

Note that the Lagrangian may not be separable because the distortion, $D(k)$, may depend on the parameters chosen for the previous packets. The dual problem to (3) is then given by

$$\max_{\lambda_1 \geq 0, \lambda_2 \geq 0} g(\lambda_1, \lambda_2) \tag{8}$$

Solving (8) will provide a solution to (3) within a convex hull approximation. Assuming we can evaluate the dual function for a given choice of λ_1 and λ_2, a solution to (8) can be found by choosing the correct Lagrange multipliers. This can be accomplished by using a variety of methods such as cutting-plane methods or sub-gradient methods [19]. Alternatively, based on the observed structure of this

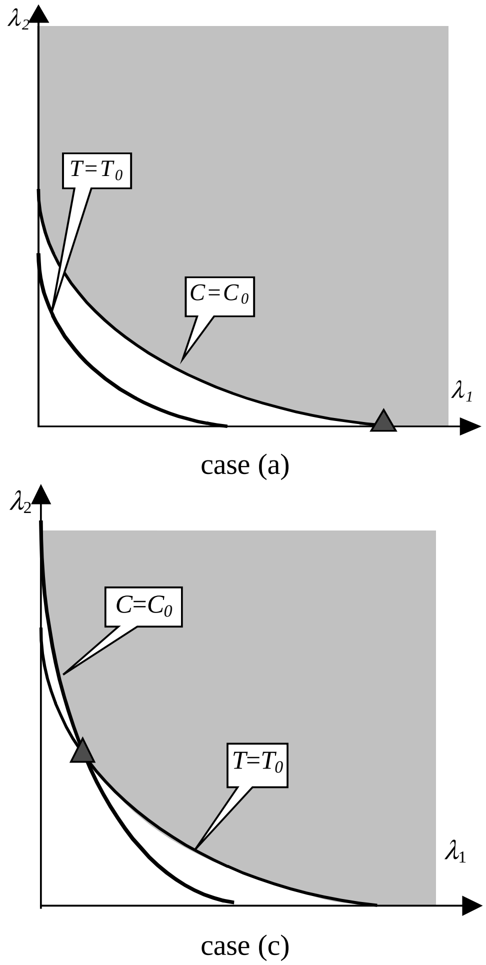

Fig. 2. Four cases of cost and delay contours.

problem, we propose the following heuristic approach, which is considerably more efficient than the above-mentioned methods.

Figure 2 illustrates four possible cases of the energy contour $C = C_0$, and the delay contour $T = T_0$ in the $\lambda_1 - \lambda_2$ plane, where C_0 and T_0 are the transmission energy and transmission delay constraints for one frame, respectively. The shaded area indicates the valid choices of (λ_1, λ_2) which satisfy both constraints. The triangle point in each figure represents the location of the optimal solution. From complementary slackness, the optimal solution must lie at one of the points where the contours intersect the axis or at the intersection of the contours [2]. In Fig. 2, we show the contours intersecting at only one point; this is the only case we have observed in practice, and we assume that it is true in the following. Let $\mathbf{h} \in \mathcal{Q} \times \mathcal{R} \times \mathcal{P}$, then the appropriate (λ_1, λ_2) can be obtained using the algorithm described below.

Step 1 (*case a, d*): Let $\lambda_2 = 0$, find the largest λ_1^* such that $C(H(\lambda_1^*, 0)) \leq C_0$. If $T(H(\lambda_1^*, 0)) \leq T_0$, $H(\lambda_1^*, 0)$ corresponds to the optimal solution. Otherwise,

Step 2 (*case b, d*): Let $\lambda_1 = 0$, find the largest λ_2^* such that $T(H(0, \lambda_2^*)) \leq T_0$. If $C(H(0, \lambda_2^*)) \leq C_0$, $H(0, \lambda_2^*)$ corresponds to the optimal solution. Otherwise,

Step 3 (*case c*):

i. Let $\lambda_1^l = 0$, $\lambda_1^r = \lambda_1^*$, $\lambda_2^b = 0$, $\lambda_2^t = \lambda_2^*$ (where λ_1^* and λ_2^* are given in steps 1 and 2).

ii. Let $\lambda_1^m = (\lambda_1^l + \lambda_1^r)/2$, find λ_2^* within $[\lambda_2^b, \lambda_2^t]$ to satisfy $T(H(\lambda_1^m, \lambda_2^*)) \leq T_0$.

iii. If $C(H(\lambda_1^m, \lambda_2^*)) > C_0$, then let $\lambda_1^l = \lambda_1^m$, $\lambda_2^t = \lambda_2^*$, and go to step 3ii. Otherwise,

iv. If $C(H(\lambda_1^m, \lambda_2^*)) < C_0 - \delta$ (δ is a relatively small number), then let $\lambda_1^r = \lambda_1^m$, $\lambda_2^b = \lambda_2^*$, and go to step 3ii. Otherwise,

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[2]Here we are referring to the optimal solution to the dual problem, which is a concave problem and so complementary slackness applies.

v. The optimal solution corresponds to $H(\lambda_1^m, \lambda_2^*)$.

In the proposed solution, when one Lagrange multiplier is fixed, the primal problem becomes a typical Lagrangian problem with only one Lagrange multiplier. This can be solved by changing this Lagrange multiplier to satisfy its corresponding constraint, using a bisection method [20]. Note that in cases (a) and (b), one of the constraints is inactive. Case (d) indicates that different combinations of (λ_1, λ_2) may result in the same minimum distortion.

Next, we consider evaluating the dual function in (7). From (2), (4), and (6), the cost of each packet $J(k)$ is a function of μ_k, ν_k, η_k and $E[D_{l,k}]$. As shown in (5), the error concealment scheme introduces dependences between packets. Specifically, the calculation of $E[D_{l,k}]$ depends on the coding parameter and power levels of the neighboring packet(s), if temporal concealment based on the motion vectors of neighboring packet(s) is used. For the particular error concealment scheme discussed above, $E[D_{l,k}]$ depends on the prediction mode (INTRA/SKIP or INTER), channel coding rate ν_{k-1}, and power level η_{k-1}, for the previous packet, through the calculation of ρ_{k-1} and $E[D_{c,k}]$, as shown in (5). The cost of the k-th packet can be described as

$$J(k) = J(\mu_{k-1}, \nu_{k-1}, \eta_{k-1}, \mu_k, \nu_k, \eta_k).$$

The dual can then be evaluated via dynamic programming. The time complexity of this is $O(M \cdot |\mathcal{Q} \times \mathcal{R} \times \mathcal{P}|^2)$, where $|\mathcal{Q} \times \mathcal{R} \times \mathcal{P}|$ denotes the cardinality of the set, $\mathcal{Q} \times \mathcal{R} \times \mathcal{P}$ [20]. Note that if a simpler error concealment scheme is used, i.e., the lost MB is recovered from the MB with the same spatial location in the previously reconstructed frame, the cost for packet k is in the form of

$$J(k) = J(\mu_k, \nu_k, \eta_k),$$

resulting in a time complexity of $O(M \cdot |\mathcal{Q} \times \mathcal{R} \times \mathcal{P}|)$.

5. EXPERIMENTAL RESULTS

5.1. Implementation Issues

In our simulations, we choose an H.263+ codec [16] to perform source coding. The test sequence is Foreman with QCIF (176×144) format at frame rate of 30 fps. We employ a simple but efficient temporal replacement strategy for error concealment. That is, at the decoder, the lost MB is replaced by the MB with the same spatial location in the previously reconstructed frame. The distortion measurement is based on a per-pixel distortion calculation, which ensures accurate estimation of the end-to-end distortion [17]. In order to clearly illustrate the proposed JSCCPA framework, we assume that channel feedback is available to the encoder in the form of which packets are received or lost. Feedback is only utilized in the calculation of the expected distortion and limits the impact of error propagation. In all experiments, the feedback delay is set as two frames and the transmission rate is 360 Kbps. We emphasize that the feedback delay is long enough to preclude retransmissions in this setting. Rate control is not implemented in the video streaming system. Thus, every frame has the same transmission delay constraint of one frame time. The image quality is measured by the peak signal to noise ratio (PSNR), which is defined by PSNR$= 10\log\frac{255^2}{\mathrm{MSE}}$ dB.

We use an RCPC code with generator polynomials (133, 171), mother code rate 1/2, and puncturing rate $P = 4$. This mother rate is punctured to achieve the 4/7, 2/3, and 4/5 rate codes. At the receiver, soft Viterbi decoding is used in conjunction with BPSK demodulation. We present experiments on Rayleigh fading channels, and the channel parameter is defined as $\mathrm{SNR} = \alpha\frac{E_b}{N_0}$. In the simulations, the bit error rates for the Rayleigh fading with the assumption of ideal interleaving were obtained experimentally using simulations, as shown in Table 1.

5.2. Performance Comparison of JSCPA and OERSC Systems

In this experiment, we compare the performance of two systems: the proposed framework in (3) with fixed channel coding rate, which is referred to as JSCPA (joint source coding and power allocation)

Channel SNR (dB)	2	6	10	14	18	22
Channel rate=1/2	1.4×10^{-3}	2.2×10^{-5}	2.1×10^{-6}	2.4×10^{-7}	6.4×10^{-8}	2.8×10^{-9}
Channel rate=4/7	1.1×10^{-1}	5.3×10^{-4}	4.1×10^{-5}	1.1×10^{-5}	3.8×10^{-6}	1.3×10^{-6}
Channel rate=2/3	3.2×10^{-1}	7.4×10^{-3}	1.7×10^{-4}	3.5×10^{-5}	1.2×10^{-5}	4.2×10^{-6}
Channel rate=4/5	4.2×10^{-1}	4.0×10^{-2}	6.6×10^{-4}	1.1×10^{-4}	3.6×10^{-5}	1.2×10^{-5}

Table 1. Performance of RCPC (in BER) over a Rayleigh fading channel with interleaving.

system, and OERSC (optimal error resilient source coding) system. In both approaches, channel coding rates are fixed. In the JSCPA approach, error resilient source coding and power allocation are jointly considered within the proposed framework (3). We refer to the OERSC system as the reference system, and run it under different channel SNR (referred to as reference channel SNR) to generate the energy constraints for the JSCPA system. Thus, the two systems have the same transmission delay constraints and use the same amount of transmission energy. The difference is that in JSCPA, the power level can vary. We illustrate the performance of the two systems in Fig. 3, where we plot the average decoded PSNR for the Foreman sequence averaged over 50 random channel error realizations under different reference channel SNR. As shown in Fig. 3, with the above simulation setup, JSCPA outperforms the OERSC system by up to 6 dB. The gain comes from the higher flexibility of the JSCPA approach, where the power level can be optimally assigned to different packets to achieve UEP for video packets that are of different importance. In addition, from Fig. 3, we can see that under some settings, little gains of JSCPA over OERSC can be obtained (e.g., when channel SNR is 12dB and the channel coding rate used is low, as shown in Fig. 3(b)). This observation can help us identify the effective components in designing a practical video streaming system. Table 2 shows how transmission power is selected for transmitting video packets in the proposed JSCPA system. Each value in parentheses denotes the percentage of packets using the corresponding transmission power level. The transmission power is proportional to the power level parameter.

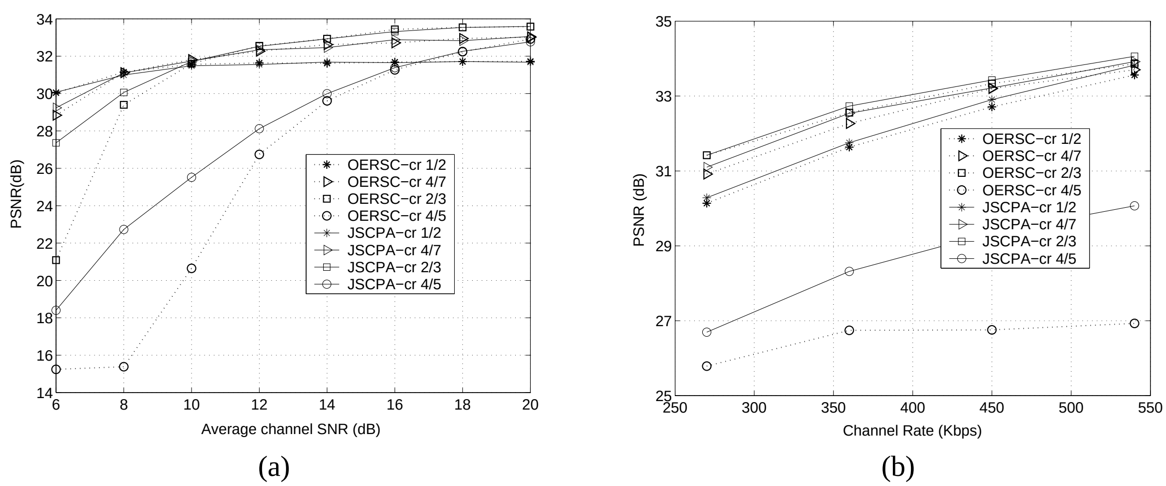


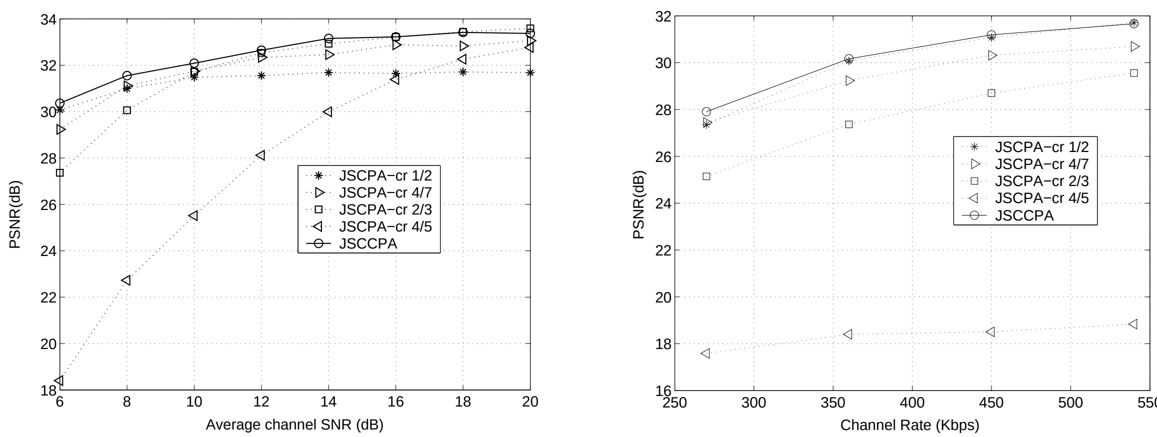
Fig. 3. JSCPA vs. OERSC (a) PSNR vs. average channel SNR (b) PSNR vs. transmission rate (cr denotes channel rate in the legend; the reference channel SNR is 12dB in (b)).

Reference channel SNR (dB)	6	12	20
Channel rate=1/2	(2.4,18.5,73.9,5.1,0)	(12.6,32.4,33.9,19.6,1.4)	(62.3,0,12.9,0,24.8)
Channel rate=4/7	(18,0,14.3,66.1,1.6)	(2.3,29.9,56.4,11.0,0.3)	(10,35,39.2,13.4,2.3)
Channel rate=2/3	(40,0,0,13,47)	(0.7,13.9,66.1,18.7.0.6)	(11.6,10.8,69.1,6.9,1.6)
Channel rate=4/5	(45.8,0,0,0,54.2)	(2,4.1,41.8,47.3,4.9)	(8.2,31.5,43.8,15.3,1.3)

Table 2. Allocation of power level (1,2,3,4,5) in percentage in the JSCPA system (the reference power level is 3).

5.3. Performance Comparison of JSCCPA and JSCPA Systems

In the second experiment, we compare the performance of the proposed framework JSCCPA (3) and JSCPA, where the channel coding parameter is fixed. Note that the two systems have the same transmission delay constraints and energy constraints, which are obtained from the corresponding reference OERSC systems. For the two systems, we plot the average decoded PSNR for the Foreman sequence under different channel SNR in Fig. 4(a) and at different transmission rate in Fig. 4(b). It can be seen that JSCCPA achieves the upper bound of all JSCPA approaches. The gain comes from the higher flexibility of the JSCCPA approach, where channel coding parameters can be optimally assigned to different packets to achieve UEP. Table 3 shows how channel coding rates are selected in the JSCCPA system. As we can see, as the channel SNR improves, less channel coding protection is needed.



(a) (b)

Fig. 4. JSCCPA vs. JSCPA (a)PSNR vs. average channel SNR (b) PSNR vs. channel transmission rate (the reference channel SNR is 12dB in (b)).

Reference channel SNR (dB)	6	8	10	12	14	16	18	20
Channel rate=1/2	96.2	67.7	41.2	19.6	6.7	4.7	1.0	1.6
Channel rate=4/7	3.8	31.9	57.3	69.6	61.3	35.0	17.8	5.6
Channel rate=2/3	0	0.4	1.5	10.8	31.3	57.7	73.9	69.6
Channel rate=4/5	0	0	0	0	0.7	2.6	7.3	23.2

Table 3. Channel coding rates in percentage in JSCCPA system.

6. CONCLUSIONS

We present a cross-layer joint source-channel coding and power allocation approach for trading off video quality and resource allocation for energy efficient wireless video communications. Through simulations, we have illustrated the advantages of the proposed JSCCPA framework, where error resilient source coding at the application layer, FEC at the link layer, and power allocation at the physical layer are jointly considered in providing UEP and consequently achieving the best video delivery performance.

7. REFERENCES

[1] Y. Wang, G. Wen, S. Wenger, and A. K. Katsaggelos, “Review of error resilience techniques for video communications,” *IEEE Signal Processing Magazine*, vol. 17, pp. 61–82, July 2000.

[2] M. Gallant and F. Kossentini, “Rate-distortion optimized layered coding with unequal error protection for robust Internet video,” *IEEE Trans. Circuits and Systems for Video Technology*, vol. 11, pp. 357–372, March 2001.

[3] Q. Zhang, W. Zhu, and Y.-Q. Zhang, “Network-adaptive scalable video streaming over 3G wireless network,” in *Proc. IEEE Int.Conf. Image Processing*, Thessaloniki, Greece, Oct. 2001.

[4] J. Rosenberg and H. Schulzrinne, “An RTP payload format for generic forward error correction,” Tech. Rep., Internet Engineering Task Force, Request for Comments (Proposed Standard) 2733, Dec. 1999.

[5] N. Bambos, “Toward power-sensitive network architectures in wireless communications: Concepts, issues, and design aspects,” *IEEE J. Select. Areas commun.*, vol. 18, pp. 966–976, June 2000.

[6] Y. Eisenberg, C. E. Luna, T. N. Pappas, R. Berry, and A. K. Katsaggelos, “Joint source coding and transmission power management for energy efficient wireless video communications,” *IEEE Trans. on Circuits and System for Video Technology*, vol. 12, no. 6, pp. 411–424, June 2002.

[7] Y. S. Chan and J. W. Modestino, “A joint source coding-power control approach for video tranmssion over CDMA networks,” *IEEE J. Select. Areas Commun.: Recent advances in wireless multimedia*, Sept. 2002, Submitted.

[8] S. Appadwedula, D. L. Jones, K. Ramchandan, and L. Qian, “Joint source channel matching for wireless image transmission,” in *Proc. IEEE Int. Conf. Image Processing*, Chicago, IL, Oct. 1998.

[9] D. A. Eckhardt, *An Internet-style approach to managing wireless link errors*, Ph.D Thesis, Carnegie Mellon University, Pittsburg, PA, May 2002.

[10] A. Majumdar, D. G. Sachs, I. V. Kozintsev, K Ramchandran, and M. M. Yeung, “Multicast and unicast real-time video streaming over wireless LANs,” *IEEE Trans. Circuits and Systems for Video Technology*, vol. 12, pp. 524–534, June 2002.

[11] T. S. Rappaport, *Wireless communications principle and practice*, Prentice Hall, 1998.

[12] N. Celandroni and F. Pototì, “Maximising single connection TCP goodput by trading bandwidth for BER,” *International Journal of Communication Systems*, vol. 16, pp. 63–79, Feb. 2003.

[13] *Multiplexing protocol for low bitrate multimedia communication over highly error-prone channels*, ITU-T Draft of H.223 Annex C, Sept. 1997.

[14] J. Hagenauer, “Rate-compatible punctured convolutional codes (RCPC codes) and their applications,” *IEEE Trans. Commun.*, vol. 36, pp. 389–400, Apr. 1988.

[15] J. G. Proakis, *Digital Communications*, McGraw-Hill, New York, Aug. 2000.

[16] ITU, *Video Coding for Low Bitrate Communication*, ITU Telecom. Standardization Sector of ITU, Sept. 1997, Draft ITU-T Recommendation H.263 Version 2.

[17] R. Zhang, S. L. Regunathan, and K. Rose, “Video coding with optimal inter/intra-mode switching for packet loss resilience,” *IEEE J. Select. Areas Commun.*, vol. 18, pp. 966–976, June 2000.

[18] D. Wu, Y. T. Hou, B. Li, W. Zhu, Y.-Q. Zhang, and H. J. Chao, “An end-to-end approach for optimal mode selection in Internet video communication: theory and application,” *IEEE J. Select. Areas Commun.*, vol. 18, no. 6, pp. 977–995, June 2000.

[19] D. Bertsekas, *Nonlinear Programming*, Athena Scientific, Belmont, MA, 1995.

[20] G. M. Schuster and A. K. Katsaggelos, *Rate-Distortion Based Video Compression: Optimal Video Frame Compression and Object Boundary Encoding*, Kluwer Academic Publishers, 1997.