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Cross-Layer QoS Control for Video Communications over Wireless Ad Hoc Networks

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Assuming a wireless ad hoc network consisting of n homogeneous video users with each of them also serving as a possible relay node for other users, we propose a cross-layer rate-control scheme based on an analytical study of how the effective video transmission rate is affected by the prevailing operating parameters, such as the interference environment, the number of transmission hops to a destination, and the packet loss rate. Furthermore, in order to provide error-resilient video delivery over such wireless ad hoc networks, a cross-layer joint source-channel coding (JSCC) approach, to be used in conjunction with rate-control, is proposed and investigated. This approach attempts to optimally apply the appropriate channel coding rate given the constraints imposed by the effective transmission rate obtained from the proposed rate-control scheme, the allowable real-time video play-out delay, and the prevailing channel conditions. Simulation results are provided which demonstrate the effectiveness of the proposed cross-layer combined rate-control and JSCC approach.

Keywords and phrases: ad hoc, video transmission, throughput capacity, effective transmission rate, packet delay, joint source-channel coding.

1. INTRODUCTION

In a wireless ad hoc network, packets are sent from node to node in a multihop fashion until they eventually reach the intended destination. As multimedia is expected to be a major traffic source on next-generation wireless networks, there has been increasing research interest in the delivery of multimedia services over such wireless ad hoc networks [1, 2, 3]. A data partitioning scheme, together with multipath routing for protecting against failures of links due to topological changes and packet losses due to fading effects, was presented in [1, 2] assuming perfect network state information. In [3], a source coding-based approach using multiple description coding is presented to take advantage of path diversity as a means to improve packet-loss resilience. However, these works, as well as much previous work appearing in the literature, target the problem from an individual user's point of view without considering the overall system capacity and fairness in a multiuser environment; these are critical issues in ad hoc networks. As a result, it remains unclear what level of video quality can be supported by an ad hoc network.

Typically, for video communications over wireless ad hoc networks, there are two main factors which can greatly affect the perceived video quality: the *effective transmission rate* associated with a source-destination pair and the *transmission errors* over representative wireless links along the corresponding path. Basically, the effective transmission rate is the highest signaling rate that can be reliably supported along a path and is constrained by interference between transmissions of neighboring nodes and the burden of supporting multihop transmissions between the source and destination as demonstrated, for example, in [4]. The cause of the throughput restriction in ad hoc networks is the pervasive need for all nodes to share channels locally with other nodes. For example, nodes close to a receiver are required to be idle to avoid collisions which would otherwise cause loss of packets for the intended receiver. If the operating rate is higher than the effective transmission rate along a path, many packets will be discarded due to channel overpumping. Thus, a rate-control scheme is both desirable and necessary to limit/eliminate the amount of lost packets and achieve a satisfactory level of received video quality over ad hoc networks. On the other hand, packet losses due to transmission errors are generally caused by channel fading, multipath effects, and interference from other electronic devices, as well as node mobility. These two factors should be considered jointly since the effective transmission rate available greatly affects the performance of error-resilience tools that can be used to combat the transmission errors as shown in [5]. More specifically, in order to achieve satisfactory video quality over ad hoc networks, it is necessary to provide a tradeoff between both kinds of packet losses subject to available resources. However, to the best of our knowledge, almost all of the current literature has considered these two factors separately and independently and proposed separate techniques to improve perceived video quality. In order to achieve improved video quality supported by ad hoc networks, and to provide a more robust video delivery system, these two factors are jointly considered in this paper.

We have investigated the capacity of a wireless ad hoc network in supporting packet video transport in [6] where we studied an ad hoc network consisting of n homogeneous video users with each of them also serving as a possible relay node for other users. We quantitatively investigated how the effective video throughput, and the resulting delivered video quality, is affected by the distance between the source and destination, measured as the number of hops required for a packet to reach the destination from the source. The results indicate that appropriate video coding rate control has to be employed in order to efficiently utilize the network capacity.

Unfortunately, the wireless channel is highly error-prone due to fading, multipath attenuation, and other impairments, which often cause packet losses. Moreover, for real-time video applications, variable network delay may cause additional losses of video data due to late arrivals. Furthermore, the reconstructed video quality associated with the use of advanced hybrid video coding approaches is very sensitive to network-induced packet losses. Therefore, error-resilient video communication techniques have received significant attention in recent years and many error-mitigation techniques have been proposed and investigated. Among the error-resilience techniques proposed, forward error correction (FEC) and automatic repeat-request (ARQ) are two basic error control techniques widely used to combat transmission errors [5, 7, 8, 9, 10]. FEC is traditionally used for real-time multimedia traffic since it requires no feedback and the delay can be bounded, while the drawbacks of FEC coding are that it requires additional bandwidth to transmit the parity packets and also has the potential for introducing increased latency. ARQ, on the other hand, requires a lower overhead than FEC since retransmission is only required when needed. But in some cases, the propagation and other delays are so large that retransmission may become unacceptable due to the resulting increased latency. Therefore, in ad hoc networks, due to the multihop transmission characteristics and stringent delay requirements for real-time video applications, FEC is more appropriate than ARQ. However, FEC should be applied in an adaptive fashion which can dynamically adapt to the prevailing operating conditions, that is, the current channel conditions and the effective transmission rate.

Therefore, based on the preceding discussion, in this paper we investigate cross-layer techniques to maximize the perceived video quality employing the H.264 video coding standard operating over wireless ad hoc networks while considering the effective transmission rate and transmission imparements jointly. Specifically, based on an analysis of the effects of interference between neighboring nodes and the burden of supporting multihop transmissions, we propose a cross-layer rate-control scheme which can dynamically control the effective transmission rate[1] for video communications from source to destination. This is achieved by feedback information obtained from the underlying routing algorithm. For instance, in ad hoc routing protocols, such as ad hoc on-demand distance vector (AODV) [11] and optimized link state routing (OLSR) [12], each node is able to maintain a routing table such that for each entry (destination), information is provided on the hop count (number of hops from source to destination). With some simple and slight modifications of the feedback routing update packet format in AODV or OLSR, each node can maintain additional information for each entry, such as packet-loss rate, bandwidth and interference conditions, required to implement the proposed approach. Then, given the effective transmission rate obtained from the proposed rate-control scheme, a model-based joint source-channel coding (JSCC) approach is employed in a cross-layer manner to optimally select the channel coding strategy subject to the constraints on delay and the prevailing channel conditions. As a result, the end-to-end quality of service (QoS) for video communication over wireless ad hoc networks can be significantly improved by taking into account both the effective transmission rate and channel error effects.

The rest of this paper is organized as follows. In Section 2, we provide some technical preliminaries, which include a brief description of H.264 and the use of interlaced Reed-Solomon codes for this application. In Section 3, we first determine the throughput capacity of the ad hoc network under an assumed homogeneous traffic pattern, and then we propose a cross-layer rate-control scheme based on the obtained analytical results. In Section 4, we propose a cross-layer joint source-channel coding (JSCC) approach given the effective transmission rate and an imposed delay constraint. In Section 5, we present some selected simulation results for

[1]Note that the effective transmission rate considered in this paper only takes into account the effect of interference between neighboring nodes and the burden of supporting multihop transmissions. It does not consider the effect of packet losses occurring on wireless links.

RTP-H.264 packet video delivery over ad hoc networks. Finally, Section 6 provides a summary and conclusions.

2. PRELIMINARIES

2.1. RTP-H.264

The H.264 standard is a newly developed video coding standard resulting from a joint effort of both ITU-T and ISO. The syntax of compliant H.264 coding is expected to result in an average reduction in bit rate by at least 50% compared to previous standards for the same video fidelity. In addition, H.264 also provides several built-in error-resilience tools, such as intraupdating and data partitioning, as well as flexible network adaptation, to combat packet losses over error-prone wireless networks. This makes H.264 an attractive candidate for wireless video transport applications, as the bandwidth resource is extremely costly in wireless environments and the packet losses induced by bit errors or link outages are quite common.

Because of the ubiquity of the Internet, and its well-entrenched networking protocols, we concentrate on the use of IP at the network level. At the transport level, although traditional ARQ strategies for point-to-point multimedia transmission (such as in TCP) may be feasible in some applications, implementing these protocols while satisfying the stringent real-time delivery requirements is clearly inappropriate. As a result, real-time applications typically use the UDP/IP combination which provides an unreliable packet delivery service. the real-time transport protocol (RTP) was developed to enable real-time multimedia applications over IP networks.

For the packetization scheme employed, in this paper, the RTP/UDP/IP protocol stack is used to support video applications over wireless ad hoc networks as in [13]. Specifically, we assume QCIF formatted video and we packetize each video slice within one video frame into a single RTP/UDP/IP packet. Since one QCIF video frame has nine slices, thus one video frame is packetized into 9 RTP/UDP/IP packets as in [7].

2.2. Interlaced RS encoding

In this paper, we use interlaced Reed-Solomon (RS) channel coding as described in [5, 14]. Basically, this scheme operates by aligning k successive data packets vertically, each of which is subsequently partitioned into q-bit symbols. An RS(n, k) code is used to encode the vertically aligned q-bit symbols to produce $n - k$ parity packets. Each of the resulting n packets is then encapsulated as a RTP/UDP/IP packet to be transmitted over the wireless network. The size of the data packets is assumed fixed and taken as just large enough to contain a single slice. This requires that each slice has the same size, which can be achieved with appropriate padding bits.

With the use of the RTP protocol, if a packet is considered lost, the RTP sequence number enables the FEC decoder to identify the lost packet, so that the location of the missing packet is known. As a result, some or all of the lost packets can be recovered through the use of the erasure-correcting capability of the FEC coding employing the corresponding location information of the lost packets.

Given the stringent delay constraints for real-time video services, it is desirable to keep the additional delay introduced by interlaced RS coding to within a single video frame. Since each QCIF frame is composed of 9 slices, this suggests the use of RS$(n, 9)$ codes. For example, the use of the RS$(15, 9)$ code, with corresponding symbol size $q = 4$ bits, provides an erasure-correcting capability of $n - k = 6$, that is, up to 6 packet losses can be fully recovered. However, it should be noted that the use of FEC coding clearly introduces additional overhead which increases the actual transmission rate. On the other hand, use of larger values of n can provide improved erasure-correcting capability but at the expense of excessive overhead which reduces the bit rate available for source coding and introduces a larger delay. In previous work [5, 7], we have demonstrated that, given the employed packetization approach as discussed previously, the RS$(15, 9)$ code can provide excellent erasure-correcting capabilities in combating packet losses over wireless networks even under severe channel conditions, say packet-loss rate greater than 5%. Therefore, in what follows, we assume that the RS$(15, 9)$ code is the strongest RS code we can apply and make exclusive use of the primitive RS$(15, 9)$ code and its punctured versions resulting in a class of RS$(n, 9)$ codes with $9 \leq n \leq 15$.

The main reasons why we do not employ an ARQ scheme to provide the error-recovery mechanism for real-time video communications over wireless ad hoc network are the following. (1) FEC coding, especially using RS codes, is quite effective in dealing with bursty packet losses commonly encountered on wireless ad hoc networks while ARQ, in the face of bursty packet losses, would introduce a substantial delay due to the requirements for retransmitting the lost packets. (2) As can be seen in Section 4.3, the delay introduced by the proposed FEC coding is much lower than that achievable with ARQ since the delay introduced by FEC coding $(n - k) \times \Delta_T$ is much less than the round-trip transmission time $2 \times T_T$ that is necessary to transmit a packet from the sender to the receiver and obtain the appropriate ACK/NACK messages from the receiver in a typical multihop-transmission scenario.[2]

Based on the discussions above, in this paper we concentrate on using FEC coding as the error-recovery scheme for real-time video applications over wireless ad hoc networks.

3. PROPOSED CROSS-LAYER RATE-CONTROL SCHEME

As discussed previously, the effective transmission rate associated with a source-destination path in a wireless ad hoc network supporting packet video is affected by several

[2]The quantities Δ_T and T_T are the interarrival time between successive packets in seconds, and the delay in transmitting a packet from sender to receiver, respectively.

parameters, such as the number of hops between source and destination [15, 16], and the number of interference neighbors of intermediate nodes along the path. As shown in [15, 16], it is clear that as the number of hops between source and destination increases, the corresponding effective transmission rate decreases accordingly. In this section, we will first determine the effective transmission rate for each node in a wireless ad hoc network under a specified traffic pattern and then propose the use of a cross-layer rate-control scheme based on the resulting analysis.

We consider a wireless ad hoc network consisting of n homogeneous nodes, each of which generates the same amount of video traffic and employs the same traffic pattern as defined in what follows. Video packets are sent from node to node in a multihop fashion until they eventually reach the destination, that is, each user has to relay traffic for other users besides being the source for its own traffic. We assume that the ith node has a transmission rate of W_i bits per second and that only those nodes that are adequately spatially separated to provide no destructive interference to each other can transmit simultaneously.

We assume that the n nodes are uniformly distributed in a domain of unit area. They are considered to be homogeneous, having the same transmission power level when they communicate with each other.

3.1. Traffic pattern

While a random traffic model is assumed in [4], in this paper we propose a different traffic scenario in order to investigate the relationship between the source-destination distance and the delivered video quality. We will characterize the traffic pattern in terms of the number of hops L taken between the source and destination. Specifically, for the above-defined ad hoc network consisting of n homogeneous users, when we say that the traffic pattern is $L = k$, we mean that the destination is located exactly k hops away from the source. As a result, the video data has to be relayed through another $k - 1$ intermediate nodes in order to reach the destination. We also assume that each node is equally likely to communicate with each of the nodes that are L hops away from it. Intuitively, as L increases, more transmission bandwidth has to be allocated since the increasing relay traffic leads to less effective video throughput for each user. The purpose of this section is to quantitatively assess this effect. In this paper, we consider a homogeneous traffic pattern, that is, L is constant for all the users and traffic. An analysis of the case of heterogeneous traffic patterns will be presented in subsequent work.

3.2. Interference model

There are a number of possibilities available for an interference model to be used in assessing the performance of wireless ad hoc networks. For example, in [4], a "protocol model" is used to assess the asymptotic capacity of an ad hoc wireless network operating in a limited domain as the node density increases. According to this model, a transmission from node X_i to node X_j is successful if the following two conditions are satisfied.

(i) Node X_j is within the transmission range of node X_i, that is,

$$|X_i - X_j| \le r, \tag{1}$$

where $|X_i - X_j|$ represents the distance between nodes X_i and X_j in the domain and r is the effective communication range of each node.

(ii) For every other node X_k that is simultaneously transmitting over the same channel, it must satisfy

$$|X_k - X_j| \ge (1+\delta)|X_i - X_j|. \tag{2}$$

This condition provides a guard zone to prevent the interference between neighboring transmissions on the same channel at the same time. The parameter $\delta > 0$ defines the size of the guard zone.

Using this interference model, it is shown in [4] that the corresponding number of interference neighbors for a node, c depends only on δ and grows no faster than linearly in $(1+\delta)^2$. Based on this observation, the authors demonstrate that the asymptotic capacity goes to zero as the number of nodes n increases.

In this work, we adopt a much simpler and less abstract interference model which is more related to physically meaningful and observable network quantities. This model is directed toward the assessment of video delivery quality rather than evaluation of asymptotic capacity as in [4]. More specifically, we assume that the number of interference neighbors associated with a node can be determined and provided to each of the nodes based upon feedback information made available through the embedded routing algorithm employed. Specific implementation of a scheme for providing this information is provided in Section 3.4.

3.3. Throughput capacity

We consider the problem of estimating the supportable throughput under the above-specified traffic pattern described in Section 3.1. We provide a simple scheme to estimate the supportable throughput based on the number of interference neighbors associated with a node which we assume is known. Furthermore, we assume that the number of interference neighbors can be obtained through the underlying routing algorithm as detailed in the subsequent section.

We begin by first assuming that each node has the same number of interference neighbors c and the transmission rate for each node is constant, that is, $W_i = W$. Furthermore, we assume that there is a spatial scheduling policy such that each node gets one slot to transmit data in every $(1+c)$ slots, and such that all transmissions are received interference-free within a distance of r from their sources.[3] Without considering the boundary regions, the number of concurrent transmissions ϕ is then upper-bounded by

$$\phi \leq \frac{n}{1+c}. \tag{3}$$

As a result, the degradation of the maximum transmission rate for each node is then bounded by

$$\beta = \frac{\phi}{n} \leq \frac{1}{(1+c)}. \tag{4}$$

Therefore, the degradation of the transmission rate of any node due to the interference between adjacent neighbors is also bounded by β. This results in a transmission rate in bits per second for any node,

$$\varsigma = \beta W \leq \frac{W}{(1+c)}. \tag{5}$$

However, this transmission rate is not the same as the corresponding effective throughput for a node. This is because part of the transmission rate obtained from (5) serves to relay traffic for others. As we will demonstrate next, the effective throughput for a node will also depend on the corresponding traffic pattern as defined in the preceding section.

Specifically, when $L \geq 1$, following (5), the aggregate transmission rate of the entire ad hoc network in bits per second is given by

$$n\varsigma = n\beta W \leq \frac{nW}{(1+c)}. \tag{6}$$

Because the traffic model is homogeneous, we have the effective useful data rate, or throughput, for a single user given by

$$R_{\text{effective}} = \frac{n\varsigma}{nL} = \frac{\beta W}{L} \leq \frac{W}{(1+c)\cdot L}, \tag{7}$$

where the factor L appears in the denominator to reflect the fact that each node must transmit the relay traffic in addition to its own traffic. As a result, it follows that in an ad hoc network, the effective transmission rate for a single user depends not only on the number of interference neighbors but also depends on the hop count between source and destination. In particular, it is necessary to adaptively adjust the video coding rate for each user when the distance L between source and destination changes.

However, in the above analysis, we assume that each node has the same number of interference neighbors and the transmission rate for each node is constant. These assumptions may not be realistic in an actual network due to the rapid change of network topology and physical environments. Therefore, in what follows, we extend the preceding analysis without these two assumptions; that is, each node may have a different transmission rate W_i and a different number of interference neighbors c_i.

Therefore, corresponding to (5), the transmission rate in bits per second for the ith node is given by

$$\varsigma_i = \beta_i W_i \leq \frac{W_i}{(1+c_i)}. \tag{8}$$

Since, in general, the effective transmission rate from source to destination is constrained by the minimum transmission rate of a particular intermediate node along the path, by following the same analysis procedure as above, the resulting effective throughput for a given source-destination pair is then given as

$$R_{\text{effective}} = \frac{\min\left(\beta_i W_i\right)}{L} \leq \frac{1}{L}\min\left(\frac{1}{1+c_i}W_i\right), \tag{9}$$

where the minimization is over all nodes along the corresponding path from the source node to the destination node.

Thus, the effective video transmission rate of the source node is constrained by both the distance L between the source and destination, and the minimum value of $\beta_i W_i$ along the path from the source to destination. It should now be clear that the effective available throughput for a given node in an ad hoc wireless network is affected by a number of factors as described above. Therefore, in order to match the transmission rate to the effective transmission rate in a video coding and transmission system, and thereby avoid channel overpumping, a rate-control scheme is necessary and a specific approach is proposed in what follows.

3.4. *Cross-layer rate-control scheme*

As can be seen from (9), the effective transmission rate for video communication from a specified source to a destination is determined by the number of hops from the source to the destination (L) as well as the bandwidth (W_i) and the number of interference neighbors (c_i) of each node along the source-to-destination route which is composed of multiple intermediate links. Basically, the embedded routing algorithm can provide the above necessary information (i.e., L, W_i, and c_i) to the source node when the route is established or when a route change occurs. Generally, the value of L is easily obtained from the routing table since most current routing algorithms, such as AODV, can provide information on the hop count between source and destination. Likewise, W_i is the transmission rate for each intermediate node, and with some slight modification of the routing update packet format, this information can also be included in the routing update messages which are sent back to the source node from the destination. As for the c_i, we can use either of two alternative methods to obtain the value for each intermediate node. One is based on the RTS/CTS mechanism in IEEE 802.11b [17], which is commonly used in ad

[3]Note that interference-free transmission does not necessarily result in successful transmission, due to wireless channel fading effects.

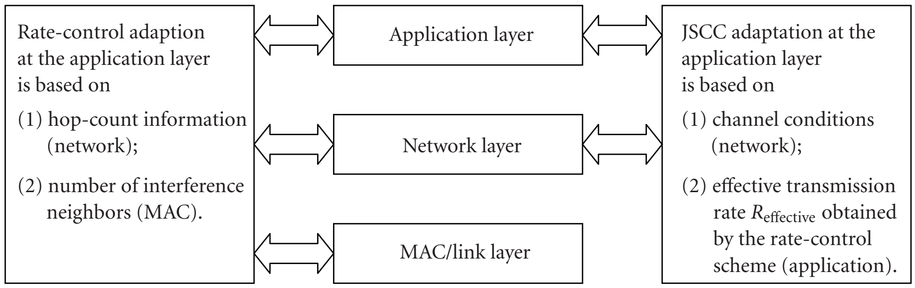


Figure 1: Illustration of the cross-layer design approach.

hoc networks. More specifically, how many different neighboring nodes sending RTS messages to a specified intermediate node can provide the value of c_i for the corresponding intermediate node. For example, if one intermediate node obtains RTS messages from 4 different neighboring nodes, this means that it has 4 interference neighbors. However, the RTS/CTS mechanism itself cannot pass this information on the number of interference neighbors to upper layers; the use of this method would result in a cross-layer design which requires some slight modifications of the layered infrastructure in order to enable the delivery of this information to upper layers as in [18]. The other method is for the node to actively send probing packets periodically, and if any other nodes receive this kind of probing packet, an acknowledgment is sent back. Based on how many different nodes send back acknowledgments, we can determine the number of interference neighbors of any intermediate node. These two methods have respective advantages/disadvantages. The first method is easy to implement and no extra bandwidth is required. But the drawback is that it may not be sufficiently accurate since if nodes have no data to send out, they will not send any RTS messages resulting in ignorance of some potential interference nodes. On the other hand, the second method is accurate but the drawback is that it needs extra bandwidth and power to send/receive probing and acknowledgment packets. However, as indicated in [18, 19], the extra bandwidth requirements generally will be small enough and should not be a burden when this method is applied.

Generally, based on connectivity, the routing algorithm can provide a set of candidate routes from the source to destination, and using (9), we can calculate the effective transmission rate for each candidate route. Instead of using the least-hop route, our routing algorithm then selects from the set of candidate routes the one that maximizes the bound on the effective transmission rate.

Since the effective transmission rate $R_{\text{effective}}$ is subject to changes in L, the number of interference neighbors, and the transmission rate of each node, in order to achieve an improved perceived video quality, it is necessary to provide a rate-control mechanism at the application layer based on the knowledge of $R_{\text{effective}}$ which is obtained through our routing algorithm.

If a route from source to destination has already been established, each time the source node encodes/sends video packets, it first checks its routing table to obtain the information on L, W_i, and c_i from the source to the desired destination. Based on the obtained information, and using (9), we can obtain the maximum effective transmission rate which is available to the source/channel coder. If the destination is no longer listed in the table, the source node initiates a route request (RRQ) to discover a new route. As soon as the new route has been established, the source node can then obtain the corresponding information on L, W_i, and c_i. On the other hand, when a route change occurs, the route error (RER) message caused by the link outage will be sent to the source node. The source node can use the reception of RER, or the initiation of RRQ, as an indication of the route change so that it can change its transmission rate accordingly.

4. CROSS-LAYER JOINT SOURCE-CHANNEL CODING

Using the rate-control scheme from the previous section, each time the source node encodes/transmits video frames, we can obtain the information on the effective transmission rate $R_{\text{effective}}$. As discussed previously, performance variations due to changes of the maximum effective transmission rate are only one of the two factors which have a major effect on perceived video quality. In this section, given the effective transmission rate $R_{\text{effective}}$ obtained from the proposed rate-control scheme, we describe the application of a cross-layer (JSCC) approach subject to a delay constraint and the prevailing operating channel conditions. We use interlaced RS codes as the channel coding strategy and employ the H.264 video coding standard as the source coding/decoding approach. This combination of rate control and JSCC represents a cross-layer approach as shown in Figure 1. More specifically, the use of the rate-control scheme requires the cooperation of the application layer, network layer, and MAC layer. First of all, the proposed rate-control scheme operating at the application layer requires information on the hop count from the routing algorithm at the network layer and information on the number of interference neighbors acquired at the MAC layer in order to determine the effective transmission rate for each source-destination pair; secondly, the proposed JSCC approach, as shown in what follows, requires information on the effective transmission rate as well as the prevailing channel conditions, including the transmission delays and information on the underlying packet-loss process, which are obtained at the network layer by the embedded routing algorithm. This information is required in order to optimally select the source/channel coding rates.

In this paper, we use RS$(n, 9)$ to denote the specific interlaced RS code used; T denotes the maximum allowable delay from the source to destination for video delivery, T_{FEC} denotes the delay introduced by FEC coding/decoding, and T_T denotes the delay in transmitting a packet from sender to receiver, that is, the sum of packetization delay, propagation delays over intermediate links, and queuing delays in intermediate nodes; R_s and R_c denote the source coding rate and channel coding rate, respectively.

The overall end-to-end performance will be measured by the resulting PSNR values for a video sequence of N_f consecutive frames and includes channel error effects as well as source coding losses. For a given effective transmission rate $R_{\text{effective}}$, PSNR(R_s, R_c) can be determined for each combination of source coding rates $\mathbf{R}_s = (R_s^1, R_s^2, \ldots, R_s^m)$, and the corresponding channel coding rates $\mathbf{R}_c = (R_c^1, R_c^2, \ldots, R_c^m)$.[4] The corresponding optimal operating parameters (R_s, R_c) are given as

$$(R_s, R_c) = \operatorname{argmax}\left\{\text{PSNR}\left(R_s^i, R_c^i\right)\right\}, \quad 1 \le i \le m, \tag{10}$$

where the maximization is performed over all possible combinations of R_s^i and R_c^i subject to the constraints

$$\begin{aligned} T_{\text{FEC}} + T_T &\le T, \\ \frac{R_s}{R_c} &\le R_{\text{effective}}, \end{aligned} \tag{11}$$

together with knowledge of the prevailing channel conditions.

In what follows, we first describe the packet-loss pattern approximation employed in this paper to represent the channel packet-loss process and analyze the delay introduced by FEC coding. Then, based on this analysis, we introduce the proposed cross-layer JSCC approach for video transmission over wireless ad hoc networks.

4.1. Loss pattern approximation

Although FEC coding is very effective in combating the effects of packet losses over wireless channels, the FEC coding gain is achieved at the cost of source coding efficiency given the total available transmission rate. Specifically, when the packet-loss rate is high, we prefer to use stronger FEC codes, while when the packet-loss rate is low, weaker FEC codes or even no FEC coding are preferred [5]. Therefore, in order to exploit FEC coding optimally, we need to specify the loss pattern of the underlying wireless links. In particular, for packet video transmission over ad hoc networks, the packet-loss patterns over all the intermediate links which make up

[4] In this paper, $R_c^i \in \{1, 9/10, 9/11, \ldots, 9/15\}$ given the packetization scheme discussed in Section 2 and $R_{\text{effective}} = R_s^i/R_c^i$.

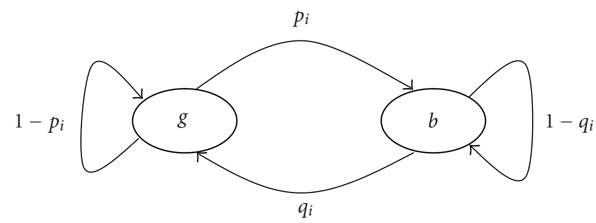


FIGURE 2: State transition diagram for the Gilbert channel.

the route from source node to destination should be tracked individually. In this paper, the loss pattern for each individual intermediate link is modeled by a two-state Gilbert channel.

4.1.1. Error pattern for individual intermediate links

The Gilbert model [20], as illustrated in Figure 2 for a two-state version, has been widely used in the literature for capturing the packet-loss patterns of wireless fading channels. In this figure, g (good) and b (bad) represent successful packet reception and packet-loss states, respectively. The two-state Gilbert model for the ith link associated with a source-destination pair can be completely specified by two parameters: the packet-loss rate P_L^i and the average burst length L_B^i.

Based on the two values P_L^i and L_B^i, we can easily calculate the associated transition probabilities of the ith link modeled by a Gilbert channel according to

$$\begin{aligned} p_i &= \frac{P_L^i}{L_B^i\left(1 - P_L^i\right)}, \\ q_i &= \frac{1}{L_B^i}. \end{aligned} \tag{12}$$

Then, the steady-state occupancy probabilities for the corresponding channel are given by

$$\begin{aligned} \pi^i(g) &= \frac{q_i}{p_i + q_i}, \\ \pi^i(b) &= \frac{p_i}{p_i + q_i}. \end{aligned} \tag{13}$$

4.1.2. Link aggregation

Generally, the route from the source to destination is a combination of several intermediate links. Although it is straightforward to compute the end-to-end loss probabilities by considering each of these links individually, this computation can be greatly simplified by using a single Gilbert channel [21] which can be used to approximate the end-to-end loss behavior of the corresponding source-destination path. Assume that the consecutive links are independent and there are a total of h intermediate links between source and destination which are represented by the channel vectors $\mathbf{P}_L = (P_L^1, P_L^2, \ldots, P_L^h)$ and $\mathbf{L}_B = (L_B^1, L_B^2, \ldots, L_B^h)$. We can directly compute the packet-loss rate P_L and the average burst length

L_B for the single Gilbert channel corresponding to this path as

$$P_L = 1 - \prod_{i=1}^{h} \pi^i(g),$$
$$L_B = \frac{1 - \prod_{i=1}^{h} \pi^i(g)}{\left(\prod_{i=1}^{h} \pi^i(g)\right)\left(1 - \prod_{i=1}^{h} (1 - p_i)\right)}, \tag{14}$$

where $\pi^i(g)$ is the steady-state occupancy probability for each intermediate link which can be obtained from (13); p_i is the transition probability calculated from (12).

After we obtain the two corresponding Gilbert parameters, the entire route from the source to destination can be modeled by this aggregate loss model. This model is employed in this paper to dynamically apply the JSCC approach as described in what follows. It should be noted that this approach is suboptimal compared to a link-by-link coding approach, since the individual intermediate link error conditions may be greatly different from each other, that is, one link may have very low packet-loss rate while another one may have a very high packet-loss rate. Generally, if we can distinguish link error conditions for each intermediate link and then design optimal source/channel coding strategies on a link-by-link basis, further performance gain can be expected. However, this requires the use of some form of transcoding scheme which will introduce much higher computational complexity, a much larger delay, and consumes more power, and is inconsistent with the IP network protocol. Therefore, it is not efficient in ad hoc networks, especially when the number of hops between source and destination is large. In this paper, despite its suboptimality, we make use of this simple aggregate Gilbert model to represent the path-loss behavior instead of individually considering each link.

4.2. FEC coding delay

As mentioned earlier, FEC coding delay is an important factor to be considered for practical operation of the proposed approach. In general, this coding delay depends on the particular code employed, the stochastic nature of traffic, and the processing speed. In this section, we incorporate the FEC coding delay as a constraint in an objective design criterion. We assume use of systematic RS(n, k) codes so that, as shown in [14], the information packets can be transmitted as generated while at the same time, they are locally buffered to allow the computation of the parity packets. Furthermore, assuming sufficient processing power, the time required to generate the parity packets at the encoder is negligible. As a result, the FEC delay is incurred solely at the decoder. In particular, if there are losses of information packets, the receiver has to wait until the arrival of the parity packets in order to make a possible recovery. The delay caused by using RS codes can then be characterized as the waiting time for the additional parity packets at the receiving end as suggested in [14].

As shown in [14], the introduced FEC delay is related to the interarrival time of packets received within a video frame. Here, we assume a particular model for the interarrival time of packets received within a corresponding frame. Specifically, packets received in a frame are assumed to be uniformly spaced. In reality, for any general video sequence, the packet delay introduced is a function of the image resolution, the frame rate, the encoder operating rate, and the network delay variability. Theoretical evaluation of this delay is generally not possible. Likewise, experimental determination of the delay caused by using FEC coding is generally not possible in most real-time applications since the encoded video material is not available prior to the start of transmission. In such cases, it is necessary to have approximate a priori estimates of the FEC delay. We now provide an expression for an approximate evaluation[5] of the FEC delay under the assumption that the packets are uniformly and periodically received over a frame, that is, we neglect the network delay variability. Let Δ_T denote the interarrival time between successive packets in seconds, let k be the number of information packets within one video frame, and let $n - k$ be the number of parity packets. Then the delay in waiting for the required FEC parity packets at the decoder is

$$T_{\text{FEC}} = (n - k)\Delta_T \tag{15}$$

with

$$\Delta_T = \frac{1}{f \cdot n}, \tag{16}$$

where f is the video frame rate in *frames/s*, and n is the number of encoded packets generated in a particular video frame. For example, if the frame rate were 30 *frames/s,* and 15 packets were generated for each frame, the interarrival time for the packets is taken as $1/(30 \times 15)$ second. This would then correspond to an interarrival time delay of 2.22 milliseconds and for the use of the RS$(15, 9)$ code, this would result in $T_{\text{FEC}} = 13.32$ milliseconds.

In later sections, this expression will prove useful in obtaining a priori estimates of the overall FEC coding delay for sequences coded at any rate.

4.3. Code selection policy

Since the application of FEC, subject to a fixed-over-transmission rate, requires throttling the coding rate to accommodate the FEC overheads, the FEC coding gain is achieved at the cost of source coding efficiency. A fixed FEC code cannot guarantee satisfactory performance for all possible channel conditions as demonstrated in [5]. Therefore, in this paper, we use a simple model-based approach to dynamically select the FEC codes, specifically RS(n, k) codes.

At the source node, the allowable delay caused by the FEC decoding at the destination is determined by the total allowable delay T together with T_T, the delay in transmitting a

[5]The expression for analytical evaluation of the FEC delay is an approximation due to the fact that it assumes that the packet-to-packet variation in the rate is negligible.

packet from sender to receiver. We assume that the transmission delay T_T is constant for the period of sending one video frame. The set of feasible RS codes capable of meeting the imposed delay constraint must then satisfy[6]

$$T_{\mathrm{FEC}} + T_T = (n-k)\Delta_T + T_T \leq T, \tag{17}$$

which is equivalent to

$$n \leq \frac{T - T_T}{\Delta_T} + k, \tag{18}$$

where the total delay T is preset as a threshold for the underlying real-time video application; T_T can be obtained by the underlying routing algorithm and is sent back to the source node. Thus, given T_T, we can find a set of feasible RS codes at the source node under the delay constraint using (18).

Since in this paper the channel coding rate R_c is determined by $R_c = k/n$, every RS code found in the previous step under the imposed delay constraint corresponds to an equivalent channel coding rate. Thus, we can obtain a set of possible channel coding rates $\mathbf{R}_c = (R_c^1, R_c^2, \ldots, R_c^m)$ from the previous step. At the same time, we can obtain a set of corresponding source coding rates $\mathbf{R}_s = (R_s^1, R_s^2, \ldots, R_s^m)$, according to

$$R_s^i = R_{\mathrm{effective}} \times R_c^i, \quad i = 1, 2, \ldots, m. \tag{19}$$

As for packet video transport over networks, the reconstructed video quality is affected by both source compression and quality degradation due to packet losses. In this paper, we assume that the two forms of induced distortion are independent and additive [22]. Thus, we can calculate the overall distortion in terms of MSE as

$$D_d = D_s + D_c, \tag{20}$$

where D_d denotes the overall distortion; D_s and D_c denote the distortion induced by source compression and channel errors, respectively.

Based on [22], the distortion caused by source compression can be approximated by

$$D_s = \left(\frac{\theta}{(R_s - R_0)}\right) + D_0, \tag{21}$$

where R_s is the source coding rate; θ, R_0, and D_0 are the parameters of the distortion model which depend on the encoded video sequence as well as on the intracoding strategy employed. These three parameters can be obtained by the method used in [6, 22].

Likewise, as in [22], the distortion caused by channel errors can be modeled by

$$D_c = \alpha P_{\mathrm{LE}}, \tag{22}$$

where α depends on the encoded video sequence as well as the encoding structure, for example, packetization scheme and intracoding ratio. P_{LE} is the residual packet-loss rate of the underlying equivalent Gilbert channel after employing an RS(n, k) code. Based on the approach proposed in [22], the residual packet-loss rate can be easily computed.

So, given the encoded video sequence as well as source/channel encoding structures, the overall distortion can be modeled as

$$D_d = D_s + D_c = \frac{\theta}{R_s - R_0} + D_0 + \alpha P_{\mathrm{LE}}. \tag{23}$$

Therefore, for each feasible pair (R_s^i, R_c^i), we compute the overall distortion at the source node using (23). The pair with the minimum D_d is selected as the source/channel coding strategy for the video frames within the current routing update interval at the source node. Then the corresponding encoded video packets plus the parity packets are sent to the destination. In Algorithm 1, we summarize the code selection procedure proposed above.

5. SELECTED SIMULATION RESULTS AND DISCUSSIONS

5.1. Simulation configuration

We performed several simulations to demonstrate the efficacy of the proposed joint rate-control and JSCC approach. In this paper, we used the QCIF Susie test sequence at frame rate 30 fps in our simulations to stream from a server to a client with a maximum allowable total delay $T = 200$ milliseconds. The sequence is coded at constant bit rate (CBR) [23]. The first frame of every group of pictures (GoPs), which is composed of 30 frames, is intracoded and the rest of the frames are intercoded as P frames without slice-based intraupdating. The use of the GoP structure is motivated by the error-prone network conditions in wireless ad hoc networks and the intracoded I frame in every GoP can effectively terminate the error-propagation effects in decoded video frames [5] resulting in improved reconstructed video quality.

In order to provide a representative evaluation of system performance, for each simulation run we generate a random ad hoc topology on the disc of unit area as a 2D Poisson point process with total number of nodes equal to 30. The transmission range r for each node is kept constant during the simulation at the value of $r = 0.2 \times (1/\sqrt{\pi})$ such that the sum of the transmission regions for all the 30 nodes (i.e., $30 \times \pi r^2 \approx 1$) almost completely covers the unit disc, thus ensuring a high degree of connectivity. This choice of the value for r can be justified by [24] where it has been shown that

[6]However, it is worthwhile to point out that in an ad hoc network, the delay in transmitting a packet from sender to receiver T_T is much greater than the interarrival time between successive packets Δ_T. As a result, the proposed FEC-based error-recovery scheme will still result in a substantially reduced delay compared to the ARQ-based scheme, which requires at least one extra round-trip transmission delay $2 \times T_T$ even if an ideal feedback channel is available.

Step 1. Using the delay constraint (18), find a feasible set of RS codes and the corresponding source coding rates.
Step 2. Use the overall distortion model (23) to approximate the overall distortion for each pair of feasible source/channel coding rates.
Step 3. Select the feasible pair with minimum overall distortion as the source/channel coding strategy for frames within the current routing update interval.

Algorithm 1: Code selection procedure.

if we assume that each node in an ad hoc network has constant power (transmission range), there is a critical transmission power required to ensure with high probability that any two nodes in the network can communicate with each other through multihop paths.

Each node in the randomly generated ad hoc network is assigned the fixed transmission rate $W_i = 2$ Mbps, which is a basic rate available in the IEEE 802.11b standard, and the number of interference nodes c_i is assigned according to the generated topology as well as the transmission range for each node. For each link in the ad hoc network, the packet-loss behavior caused by transmission errors is modeled as a two-state Gilbert model as in [21]. The available packet-loss rate for each link is uniformly assigned in the range of $0.5\% - 10\%$ and the available average burst length is selected uniformly in the range 1–4. After we obtain the two parameters of the Gilbert model for each intermediate link, the entire route from the source to destination can be modeled by an aggregate Gilbert model as discussed previously. Lastly, as shown in [25], the delay in using AODV on a per-link basis, not including queuing delay, is about 20–40 milliseconds given the packet size's range of our scenario, so the delay of each node-link pair is assigned uniformly in the range of 20–60 milliseconds. This quantity includes the propagation delay, the processing delay, as well as queuing delay in our simulation. Given a randomly generated topology, we initially choose a source-destination pair and stream the video from the source to the destination using the path with the highest effective transmission rate as described in Section 3.4. During transmission, the environments are updated every 1 second which can cause changes in the effective transmission rate and channel conditions. During successive 1-second intervals, the environments are kept constant.

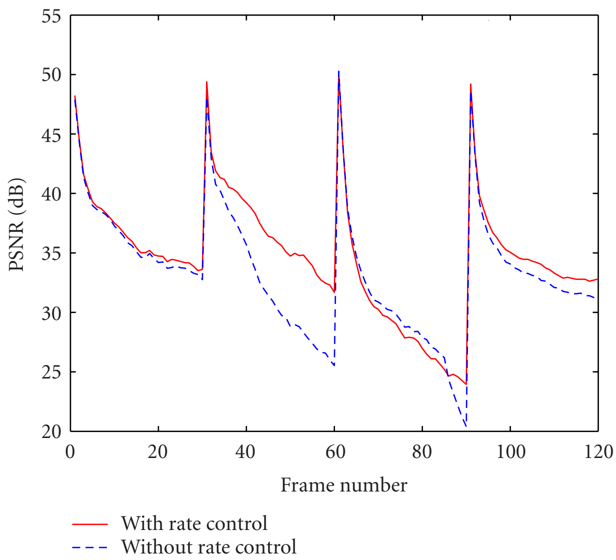
5.2. Performance evaluation of the rate-control scheme

To demonstrate the effectiveness of our proposed rate-control scheme, we use a representative drop-tail scheme for comparison which does not use rate control. More specifically, it employs a fixed source coding rate $R_s = 96$ Kbps and when the rate exceeds the current effective transmission rate available for the selected source-destination pair, it will drop the subsequent encoded packets.

In Figure 3, we show a performance comparison between our proposed rate-control scheme and the drop-tail scheme in the scenario where packet losses are caused only by channel overpumping[7] and no FEC coding is employed. It should be noted that due to the use of CBR encoding, the video quality is not constant. As a result of the CBR bit-rate control, the video quality varies periodically [7]. In Figure 3, the average PSNR using the proposed rate-control scheme is 34.77 dB while it is 33.36 dB for the case of no-rate control. Thus, a 1.5 dB performance gain can be achieved using the proposed rate-control scheme. From the channel profile, also illustrated in Figure 3, we can see that for GoP no. 1, no. 2, and no. 4, the effective transmission rate constrained by interference and multihop transmission is higher than the fixed 96 Kbps. Thus, using rate control can fully exploit the effective transmission rate resulting in improved performance compared to using a fixed-rate coding scheme. On the other hand, for GoP no. 3, it is obvious that the fixed source coding rate is higher than the prevailing effective transmission rate; therefore, packet losses will occur when the transmission buffer is full resulting in the last couple of frames being lost which cause substantial performance degradation. A lost frame is concealed by just copying the previous frame and if several consecutive frames are lost, the degradation will be even more serious since the concealed frames are then used as correctly received frames to conceal the subsequent lost frames. This results in substantial error propagation. For example, in Figure 3, we can see that there is substantial performance degradation around the 90th frame for the no-rate-control case due to channel overpumping. Furthermore, although the performance degradation caused by the channel overpumping packet losses has been partially compensated using passive error concealment (PEC), the performance is still not as good as using the rate-control scheme.

Therefore, since the proposed rate-control scheme can adapt to the changes in the transmission environments, that is, the number of interference neighbors and the number of hops between source and destination, it can enable the video encoding system to adapt to the corresponding changes in the effective transmission rate. On the other hand, if we do not use a rate-control scheme, the fixed-rate coding scheme will always cause performance loss. More specifically, if the fixed rate is lower than the effective transmission rate, performance loss is due to the source coding inefficiency result-

[7] Here, we assume that no transmission errors occurred.



(a)

Corresponding channel profile

GoP no.	1	2	3	4
R_{eff} (Kbps)	110	183	80	120
Fixed rate (Kbps)	96	96	96	96

(b)

Figure 3: Performance comparison between using rate control and without rate control (fixed source coding rate is 96 Kbps); no FEC is used; packet loss is caused only by channel overpumping; the QCIF Susie sequence.

Table 1: Simulation results for the QCIF Susie sequence.

Run	Fixed RS(15, 9) code	No FEC coding	JSCC
1	29.45 dB	30.56 dB	33.47 dB
2	30.56 dB	29.34 dB	33.01 dB
3	32.35 dB	31.86 dB	34.73 dB
4	30.05 dB	30.94 dB	34.24 dB
5	32.69 dB	30.77 dB	35.32 dB

ing from the use of an unnecessarily lower source coding rate at the video encoder. Likewise, if the fixed rate is higher than the effective transmission rate, performance loss is caused by packet losses due to buffer overflow.

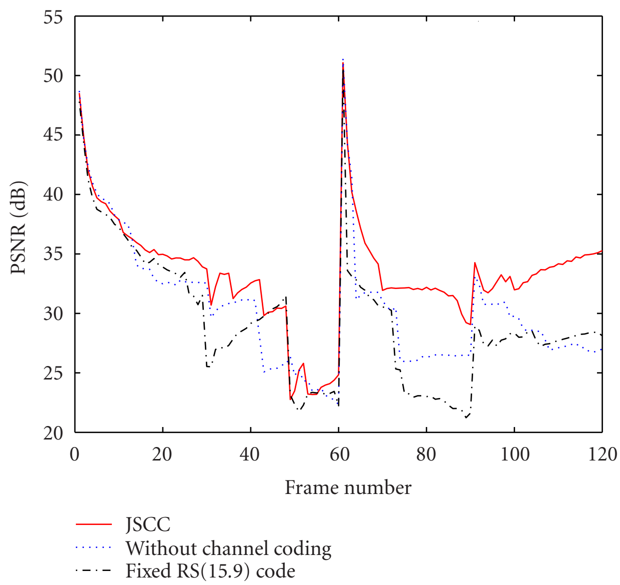
5.3. Performance evaluation of the JSCC approach

To evaluate the performance of the joint rate control and JSCC approach, 5 different simulations were conducted for the QCIF Susie test sequence. These results allow us to observe the relative performance of the proposed JSCC approach compared to two representative fixed-rate channel coding schemes where neither channel coding nor the RS(15, 9) code is used exclusively. Furthermore, the proposed rate-control scheme is used for all the above three approaches. Therefore, the packet losses are caused by transmission errors or excessive delays or both. The results are tabulated in Table 1. In each row, we show the PSNR of the corresponding reconstructed video.[8]

Clearly, the proposed joint JSCC approach outperforms the other two representative fixed channel coding schemes as can be seen from the table. More specifically, the fixed RS(15, 9) code introduces a relatively large delay as described previously and often causes packet losses when the total delay exceeds the 200-millisecond threshold. Furthermore, because of the unnecessarily large coding overheads, it can cause performance degradation in source coding efficiency although it is capable of providing stronger FEC protection. Therefore, sometimes it even provides worse performance than the no-coding case, especially when channel conditions are relatively good. On the other hand, using no channel coding generally will not introduce unacceptable delay resulting in packet losses, but when channel conditions are poor, say $P_L > 5\%$, it cannot provide the necessary protection against transmission errors. However, the proposed JSCC approach can judiciously achieve an effective tradeoff between source coding and channel coding subject to the constraints on effective transmission rate, allowable end-to-end delay, and prevailing channel conditions. Therefore, improved performance can be achieved compared to the two fixed channel coding schemes.

Another interesting point is the distribution of individual frame qualities. The plot in Figure 4 shows the PSNR values for each video frame in the test sequence. It corresponds to one pass in simulation run no. 2. As seen from Figure 4, it should be clear that using no channel coding will not provide enough protection against channel errors while using the fixed RS(15, 9) code generally will cause extra packet losses due to the large delay it induces as well as the corresponding degradation of source coding efficiency. This is demonstrated in GoP no. 3 and no. 4, that is, from the 61st frame to the 120th frame, where we can see that under the imposed delay constraint, although using the RS(15, 9) code can provide higher protection, many packets are lost due to late arrivals.

[8] All the simulations are run for 100 iterations in order to obtain statistically meaningful results.



(a)

Corresponding channel profile

GoP no.	1	2	3	4
P_L (%)	5.8	13.2	9.7	11.6
L_B	4.3	5.2	3.8	4.1
R_{eff} (Kbps)	146	146	183	183
T_T (ms)	153	179	196	191

(b)

Figure 4: Performance comparison between the JSCC approach and the representative fixed channel coding schemes; rate control used in all cases; packet loss is caused by a combination of channel overpumping and excessive delay; the QCIF Susie sequence.

(a)

(b)

(c)

(d)

(e)

(f)

Figure 5: Comparison of decoded frames for the Susie sequence at the 37th frame (first row) and the 106th frame (second row); (a), (d) the proposed JSCC approach (PSNR = 32.17 dB, 33.96 dB, resp.); (b), (e) no FEC coding (PSNR = 30.37 dB, 28.52 dB, resp.); (c), (f) the fixed RS(15, 9) scheme (PSNR = 28.52 dB, 27.22 dB, resp.).

Therefore, a substantial performance loss is observed compared to either the JSCC approach or the no-channel coding approach. On the other hand, although using no channel coding results in the best source coding efficiency, there are no error correcting capabilities which can be used to combat transmission errors. Therefore, the corresponding performance is considerably worse than the JSCC approach.

The above objective results are based on a quantitative assessment of reconstructed PSNR values. In Figure 5, we also show some subjective results based on the reconstructed frames taken from the decoded test sequences of the simulation run shown in Figure 4. From Figure 5, we can see that the proposed JSCC approach can provide improved subjective performance compared to the other two

Table 2: Performance comparison (in PSNR, dB) of the proposed joint rate-control/JSCC approach for different delay constraints T (milliseconds); the Susie sequence.

Run	$T = 150$	$T = 200$	$T = 300$	$T = \infty$
1	32.14	33.47	34.55	34.83
2	31.83	33.01	34.23	34.72
3	33.07	34.73	35.68	36.02
4	32.86	34.24	35.45	35.95
5	34.12	35.32	36.01	36.43

fixed schemes. These results again support the preceding objective assessments.

In what follows, we illustrate the performance of the proposed joint rate-control/JSCC approach under the influence of the imposed delay constraint. In Table 2, we repeated the same simulation process as in Table 1 for the proposed rate-control/JSCC approach but with varying delay constraint.

In Table 2, it can be observed that the imposed delay constraint does play a very important role. As the delay constraint T becomes looser, the performance of the proposed approach improves. For example, for simulation run no. 1, the difference between $T = 150$ milliseconds and $T = \infty$ (i.e., no delay constraint) is about 2.5 dB. The reason is that as the delay constraint becomes looser, we can find a larger set of feasible RS codes satisfying the delay constraint (18). As a result, it is possible to provide stronger FEC protection in the face of packet losses occurring on wireless links, thus providing improved performance. This table also demonstrates that for video applications over wireless ad hoc networks, the delay requirements for video applications always provide a challenging issue for system design and this issue becomes more important for real-time video applications in order to obtain a satisfactory end-to-end reconstructed video quality.

6. SUMMARY AND CONCLUSIONS

In this paper, by jointly considering the effective transmission rate constrained by the number of interference neighbors and multihop transmissions together with the transmission errors occurring on wireless links, we proposed and investigated a cross-layer rate-control scheme based on an analytical study of how the effective transmission rate is affected by the network operating parameters. Based on that, we proposed a cross-layer JSCC approach which yields the maximal video quality at the client by considering the network conditions, the delay constraint, and the available effective transmission rate. The analytical results demonstrate that it is necessary to incorporate such a rate-control scheme and JSCC approach so that the delivered video quality can be maximized. Finally, the simulation results demonstrated the effectiveness of our proposed approach for video communications over wireless ad hoc networks.

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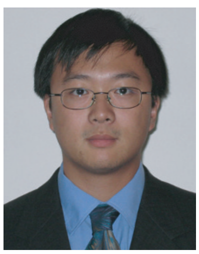

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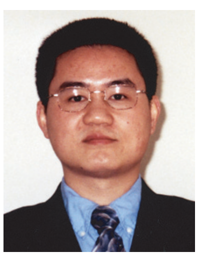

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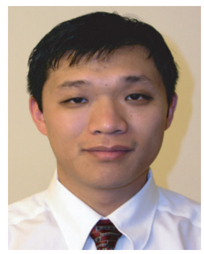

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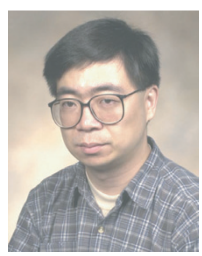

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