Routing-aware multiple description video coding over mobile ad-hoc networks

Yiting Liao, *Student Member, IEEE* and Jerry D. Gibson, *Fellow, IEEE*

***Abstract*—Supporting video transmission over error-prone mobile ad-hoc networks is becoming increasingly important as these networks become more widely deployed. We propose a routing-aware multiple description video coding approach to support video transmission over mobile ad-hoc networks with multiple path transport. We build a statistical model to estimate the packet loss probability of each packet transmitted over the network based on the standard ad-hoc routing messages and network parameters. We then estimate the frame loss probability and dynamically select reference frames in order to alleviate error propagation caused by the packet losses. We conduct experiments using the QualNet simulator that accounts for node mobility, channel properties, MAC operation, multipath routing, and traffic type. The results demonstrate that our proposed method provides 0.7-2.3 dB gains in PSNR for different video sequences under different network settings and guarantees better video quality for a selectably high number of users of the network. Furthermore, we examine the estimation accuracy of our proposed estimation model and show that our model works effectively under various network settings.**

***Index Terms*—multiple description coding, video communications, multipath routing, mobile ad-hoc networks, error resilience**

I. Introduction

There has been a growing interest in video communications over mobile ad-hoc networks due to the deployment of ad hoc networks in military, homeland defense, and disaster recovery applications. However, mobile ad-hoc networks impose significant challenges to video transmissions since node mobility and the lack of infrastructure in the network can lead to frequent link failures and route changes. Furthermore, the link quality is affected by fading and interference in the wireless channels. Given the error-prone nature of the mobile ad-hoc networks and the vulnerability of compressed video to packet losses, it is important to provide effective error resilience for video transmission over mobile ad-hoc networks.

Multiple description coding (MDC) has been shown to be a promising technique for video transmission over lossy networks [1]. With MDC, a video sequence is encoded into multiple descriptions such that each description can be used to reconstruct the video with low but acceptable video quality. When more number of descriptions are received for reconstruction, higher video quality can be achieved. As long as all descriptions are not lost simultaneously, somewhat acceptable quality can be maintained. In order to reduce the likelihood of simultaneous loss of descriptions, different descriptions are transmitted through different paths. This is referred to as MDC with multiple path transport (MPT). MPT helps to reduce the possibility of simultaneous loss of different descriptions and enables load balancing in networks. Many studies show that combining MDC with MPT leads to substantial performance gains for video transmission over these networks [2]–[5].

However, most MDC approaches employ motion compensated prediction to exploit the temporal correlation between video frames. This introduces a condition called "mismatch", in which the decoder cannot correctly reconstruct the current frame due to the loss of a reference frame. In other words, the errors in one frame may propagate to the subsequent frames and significantly degrade the video quality. Many approaches have been proposed to mitigate or eliminate mismatch [6]–[8], yet it comes at the cost of coding redundancy. Other solutions to address the problem are traffic allocation and path selection for MDC with MPT [9]–[12], in which video packets are spread over different paths based on the error characteristics of paths to minimize end-to-end distortion.

In this paper, we propose a **routing-aware MDC** (RA-MDC) approach with MPT to alleviate error propagation caused by packet losses. This approach uses ad-hoc routing messages available in the standard routing protocols to estimate the packet loss and then select the reference frames accordingly. We first explore the relationship between packet losses and routing messages. We build a model to estimate the packet loss probability of each packet according to the routing messages received by the transmitter and the transmission delay determined by the MAC layer access mechanism and network parameters. Based on this model, we then estimate the frame loss probability, and apply a threshold-based algorithm to select the reference frames to mitigate error propagation. Unlike common reference picture selection (RPS) work [13], [14], our approach does not require any extra channel feedback but retrieves information from normal routing messages.

We implement and evaluate our design using the Qualnet simulator for different video sequences and network settings. Simulation results show that our proposed method achieves gains in average PSNR of up to 2.3 dB and gains in a perceptual quality metric $\text{PSNR}_{r,f}$ [15], [16] of up to 13.6 dB, which indicates fewer bad-quality frames for most users over the network. We also study the estimation accuracy of our proposed model and find that our model works effectively under various network settings.

This paper is organized as follows. We discuss related work

This research has been supported by the UC Discovery Grant Program and Applied Signal Technology, Inc.

The authors are with the Department of Electrical and Computer Engineering, University of California, Santa Barbara, CA, 93106, USA (e-mail: yiting@ece.ucsb.edu; gibson@ece.ucsb.edu).

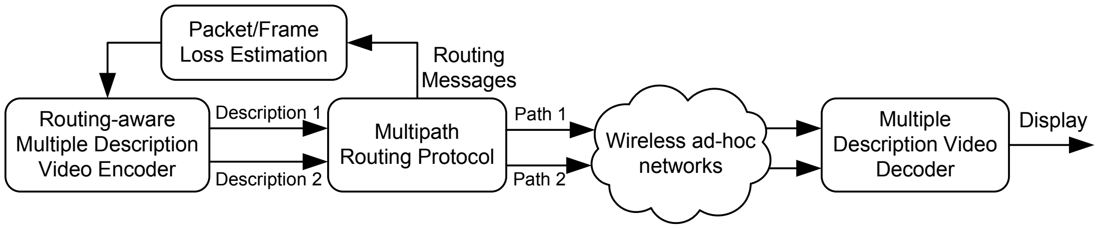


Fig. 1. System architecture of the proposed system using routing-aware multiple description coding and multipath routing

in Section II. In Section III, we describe the architecture of a routing-aware MDC system. Section IV and Section V present our proposed packet loss estimation method based on routing messages and discuss our reference frame selection algorithm for MDC using the estimated packet loss. We introduce the simulation setup for the routing-aware MDC system in Section VI-A and analyze our experimental results in Section VI-B. Section VII concludes the paper.

II. Related Work

Combining MDC with path diversity for video communications over wireless ad hoc networks has drawn significant attention in recent years. The research in this area can be generally divided into two categories. One category studies the effectiveness of MDC methods based on a specific network model with path diversity [3]–[5]. In [3], the authors proposed a MDC method based on the lapped orthogonal transform and examined the performance on a two-path system with the same capacity and error characteristics. An adaptive MD mode selection approach is proposed in [5] to adapt to the network conditions as well as to the video characteristics. This approach selects the optimal MD mode by calculating the end-to-end distortion based on the Gilbert packet loss model. In [4], Mao et. al. compared feedback based reference picture selection, layered coding, and MDC schemes with multipath transport and found that MDC is preferable when a feedback channel cannot be set up.

The other category addresses the path selection and rate allocation problem for MDC given a particular MDC scheme [9]–[12]. Begen et. al. proposed a multi-path selection method that chooses a set of paths maximizing the overall quality at the client based on the network parameters, media characteristics and application requirements [9]. The authors in [10] formulated a routing optimization problem that minimizes the application layer video distortion and provided a genetic-algorithm based approach to compute two disjoint paths for video transmission. In [11], the authors formulated the video distortion as a function of network layer behavior and proposed a branch-and-bound framework to produce optimal solutions. Different metrics used for the path selection for MDC are discussed in [12], and a practical interference aware distributed routing protocol is proposed.

Our proposed method falls into the first category; however, instead of assuming that two node-disjoint paths with the same error characteristics are available or the set of paths is given, we consider multipath routing in a more practical network and utilize the route messages to select the proper reference frames. Our work is inspired by the reference picture selection (RPS) methods proposed in [13], [14]. Most of the RPS work assumes an extra feedback control channel from the video receiver to the sender, and the receiver thus sends an ACK/NACK for every video packet [13]. Such an approach can lead to extra overhead and cost, especially in a large network. Our routing-aware MDC method, on the other hand, does not require any additional control packets or an extra channel connection in the network. We only extract and utilize the information embedded in typical routing messages, thus saving network bandwidth.

III. System Architecture

The system architecture of the proposed system is shown in Fig. 1. The routing-aware multiple description (MD) video encoder generates two video descriptions based on the multiple state video coding (MSVC) method [2] and routing-aware reference frame selection. The two descriptions are transmitted through two paths established by the multipath routing protocol. At the receiver, the descriptions are decoded by the MD video decoder, in which the refined error concealment method proposed in [17] is applied to reconstruct the video sequence from received descriptions. We discuss the MD video encoder/decoder and the multipath routing protocol in this section and present our packet loss estimation approach in Section IV.

A. Multiple Description Video Encoder/Decoder

Multiple description video coding (MDC) is an effective approach to enhance the error resilience of video transmission over lossy networks. The general idea is to encode the video sequence into several descriptions with equal importance. Each description can be decoded independently or combined with other descriptions for reconstruction. In general, the reconstructed video achieves better video quality when more descriptions are received.

Among the many proposed MDC algorithms [1], multiple state video coding (MSVC) proposed by Apostolopoulos in [2] is a very popular method since it is easy to implement and compatible with different video standards. Thus, we apply MSVC to our MD video encoder. At the encoder, the video sequence is temporally downsampled into two subsequences consisting of odd and even frames, and the odd and even frames are encoded as two descriptions using an H.264 encoder. During the encoding process, we use routing messages from the routing protocol to help the encoder select the reference frames. The details are presented in Section V.

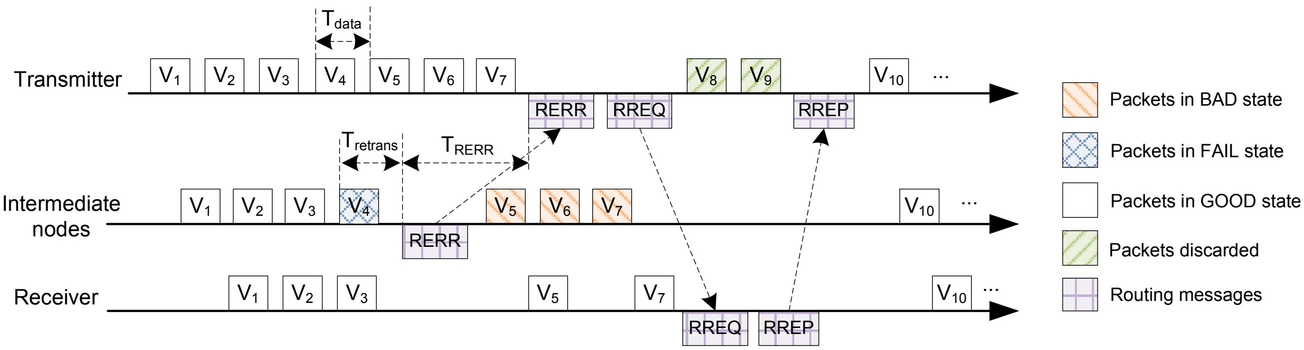


Fig. 2. An example to illustrate the packet losses in the network and the corresponding routing messages

At the decoder, we utilize the MSVC decoder with the refined error concealment method as proposed in [17]. When the decoder receives the corrupted descriptions, it decodes the correctly received MBs and conceals the lost MBs with the refined MB concealment method that considers the information from both descriptions for better recovery. The refined intra MB concealment reconstructs the lost MBs in the intra frames by using the temporal correlation between adjacent intra frames in two descriptions, while the refined inter MB concealment uses an additional reference list to perform the motion-compensated concealment. Finally, the concealed descriptions are interleaved to achieve the final reconstruction.

B. Multipath Routing Protocol

As mentioned in Section I, combining MDC with multipath transport is an appealing approach because it provides error resilience as well as load balancing for video transmission over networks. To support MDC with path diversity, a multipath routing protocol is required to build multiple paths between the source and destination nodes through the ad hoc network.

Many multipath routing protocols have been proposed to support multipath transport in wireless ad hoc networks [18]–[21]. In [18], the authors proposed a multipath extension of dynamic source routing (DSR) [22], in which a set of alternate link-disjoint routes are maintained. Another extension of DSR called split multipath routing (SMR) is proposed in [19]. It focuses on building and maintaining multiple maximally disjoint paths. AOMDV [20] and AODVM [21] are two multipath protocols extended from the ad hoc on-demand distance vector (AODV) routing protocol, in which AOMDV computes multiple loop-free and link-disjoint paths [20] and AODVM finds multiple node-disjoint paths [21].

Although these on-demand multipath routing protocols have different optimization criteria to establish routes, they all consist of two basic mechanisms: route discovery and route maintenance. A route discovery process is triggered when a source node needs a route to transmit packets to a destination node. A route request (RREQ) message is flooded to the entire network to find the routes. When the RREQ reaches the destination node, a route reply (RREP) message is sent back to the source node to build a new route. Route maintenance deals with the situation that a route becomes worse or even broken. When a route breaks, the node that detects the link failure sends a route error (RERR) message to the source node. Once the source node receives the RERR, either a new route is built from the route table or a route discovery is initiated to reconstruct a new route.

We notice that the routing discovery and maintenance mechanisms in most of the routing protocols provides feedback concerning the network conditions. This inspires us to utilize these feedback messages to estimate the packet losses in the network and to adapt the video coding accordingly. In this paper, we implement SMR as our multipath routing protocol due to its popularity and simplicity [23]. However, our solution is not limited to this particular protocol and can be extended to other multipath routing protocols.

IV. Packet Loss Estimation

In this section, we present how to use routing messages to estimate the packet losses in the network. Based on the routing mechanisms, a RERR message is initiated when the MAC layer fails all retransmission attempts to transmit a packet to the next hop destination. This RERR indicates that a link becomes unreliable and the packets transmitted through this link suffer a high packet loss rate. Before the source node receives the RERR, video packets sent from the source node are still transmitted through this error-prone link and are susceptible to losses. When the source receives the RERR, it either reconstructs the route from the route cache or initiates the route recovery process to find a new route. Packets scheduled to be transmitted in the broken route during the route recovery process are discarded and marked as lost.

In our previous work [24], we use a simple method to utilize the routing messages, that is, every time a route error (RERR) message is received by the source node, we assume that the previously transmitted packet is lost. However, due to the transmission delay of the video packets and routing messages, a RERR may indicate possible losses of several previously transmitted video packets. Therefore, here we propose a model to estimate the packet loss probability of the packets sent through an unreliable link.

Figure 2 illustrates how the RERR message correlates to the packet losses in the network. As shown in Fig. 2, a RERR is initiated at the intermediate node when video packet v_4 exhausts all retransmission attempts and still fails to transmit

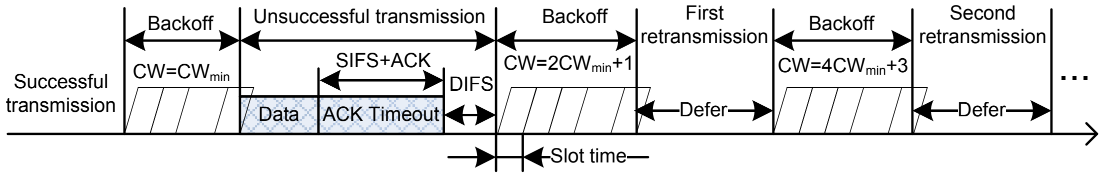


Fig. 3. Packet retransmission procedure based on basic access mechanism

to the next hop destination. We define the retransmission delay of this packet as T_{retrans}. After time T_{RERR}, the source node receives the RERR and stops transmitting video packets through the unreliable link. We see that packets v_5, v_6, v_7 sent during time period $T_{\text{retrans}}+T_{\text{RERR}}$ are still transmitted through the unreliable link and are very susceptible to packet loss.

We assume that anytime the source receives a RERR, the preceding video packets sent from the source follow the same packet loss distribution under the same network conditions. Therefore, we denote Pr(n) as the packet loss probability of the n^{th} preceding packet sent from the source before the source node receives a RERR. Our main goal is to model Pr(n) and utilize it to determine the potential corrupted frames. Due to the random delay between link failure and RERR reception at the source, the n^{th} preceding packet before RERR can be sent at a time before, right at, or after the link failure happens. We use three states to represent these three cases: GOOD means the packet is sent before the link failure, FAIL means the packet fails to transmit and triggers RERR, and BAD means the packet is sent after the link failure. According to our above analysis, we define Pr(n) as

$$\text{Pr}(n) = \lambda_g \cdot p_g(n) + \lambda_f \cdot p_f(n) + \lambda_b \cdot p_b(n) \tag{1}$$

where λ_g, λ_f, and λ_b represents the packet loss probability in GOOD, FAIL, or BAD state respectively, and $p_g(n)$, $p_f(n)$, and $p_b(n)$ denotes the probability of the n^{th} preceding packet in these three states, respectively. In the following, we estimate the state probability distribution and packet loss probabilities in these three states.

A. Estimation of State Probability Distribution

The state of a video packet depends on the delay of the link failure feedback and the transmission interval of video packets. For example, as shown in Fig. 2, v_4 is the packet that triggers RERR and hence is in FAIL state. The packets sent before v_4 (e.g. v_3) are in GOOD state while the packets sent after v_4 are in BAD state. Therefore, we can compare the video packet transmission interval T_{data} and the delay of the link failure feedback T_{delay} to determine the state of the packets sent before receiving the RERR by

$$\begin{cases} p_g(n) = p\left(T_{\text{delay}} \le (n-1)T_{\text{data}}\right) \\ p_f(n) = p\left(nT_{\text{data}} \ge T_{\text{delay}} > (n-1)T_{\text{data}}\right) \\ p_b(n) = p\left(T_{\text{delay}} > nT_{\text{data}}\right) \end{cases} \tag{2}$$

We can calculate the video packet interval T_{data} by

$$T_{\text{data}} = L/R_t \tag{3}$$

where R_t is the transmission bitrate of the video sequence and L is the payload size. Then in order to calculate Eq. (2), we need to estimate the probability distribution of T_{delay}.

As shown in Fig. 2, we see that T_{delay} consists of two parts: the retransmission delay of a packet that fails all retransmission attempts (denoted as T_{retrans}) and the time period to transmit the RERR to the source (denoted as T_{RERR}). So we have

$$T_{\text{delay}} = T_{\text{retrans}} + T_{\text{RERR}} \tag{4}$$

The values of both T_{retrans} and T_{RERR} depend on the MAC layer access mechanism. The basic access method of the IEEE 802.11 MAC layer is the distributed coordination function (DCF) based on the carrier sense multiple access with collision avoidance (CSMA/CA) scheme [25]. The DCF method provides a basic access mechanism and an optional RTS/CTS access mechanism. In this paper, our estimation is based on the basic access mechanism. Our method can also be applied to the RTS/CTS mechanism. Next, we estimate T_{retrans} and T_{RERR} based on the basic 802.11 DCF mechanism.

1) T_{RERR}: We first estimate the time period to transmit the RERR to the source, T_{RERR}, by

$$T_{\text{RERR}} = n_{\text{hop}} \cdot T_C \tag{5}$$

where n_{hop} is the average number of hops to transmit RERR to the source, and T_C is the transmission time for a successful RERR transmission. For the basic 802.11 access mechanism, we have

$$T_C = T_{\text{DIFS}} + T_H + T_{\text{ctl}} + T_{\text{SIFS}} + T_{\text{ACK}} \tag{6}$$

where T_{DIFS} is the DIFS time, T_H represents the transmission time of MAC and PHY header, T_{ctl} is the transmission time of RERR payload, T_{SIFS} is the SIFS time, and T_{ACK} denotes the ACK transmission time.

2) $T_{retrans}$: We then estimate the transmission delay T_{retrans} of a packet that fails to transmit from the current station to the next hop destination after exhausting all retransmission attempts. As shown in Fig. 3, each transmission period consists of a defer access and a backoff process. The transmission procedure starts when the station senses an idle distributed inter-frame space (DIFS) and invokes a backoff procedure. The backoff time is uniformly chosen in the range of $[0, CW]$, where *CW* is the current contention window (CW) size. Then the station sends out the video packet. If the transmitting station does not receive the acknowledgment (ACK) within the ACK timeout interval, the station concludes that the transmission has failed and invokes a retransmission process until the retransmission limit is reached. Note that CW takes an initial value of $CW_{\min}$ and exponentially increases after each unsuccessful transmission, until it reaches the maximum CW size of $CW_{\max}$.

Based on the above analysis, the transmission delay of a packet that fails all retransmission attempts is

$$T_{\text{retrans}} = mT_D + T_{\text{backoff}} \tag{7}$$

where m is the retransmission limit, T_D is the time period of a defer access, and T_{backoff} is the overall backoff time.

Similar to Eq. (6), we have

$$T_D = T_{\text{DIFS}} + T_H + T_{\text{data}} + T_{\text{SIFS}} + T_{\text{ACK}} \tag{8}$$

where T_{data} is the transmission time of the data payload.

The overall backoff time is a random variable that is the sum of a series of independent random variables uniformly distributed in the range of $[0, W_i]\cdot T_{\text{slot}}$ and W_i is the *CW* size in the i^{th} retransmission defined by

$$W_i = \begin{cases} 2^i \cdot (CW_{\min}+1) - 1 & i \le m' \\ 2^{m'} \cdot (CW_{\min}+1) - 1 & i > m' \end{cases} \tag{9}$$

where $m' = \log_2 \dfrac{(CW_{\max}+1)}{(CW_{\min}+1)}$.

We define T_{B_i} as the backoff time in the i^{th} retransmission, so we have $T_{B_i} \sim U(0, W_i \cdot T_{\text{slot}})$, where $U(0, W_i \cdot T_{\text{slot}})$ represents an uniform distribution in the range $[0, W_i] \cdot T_{\text{slot}}$. Thus the overall backoff time is

$$T_{\text{backoff}} = \sum_{i=0}^{m-1} T_{B_i} \sim U_{\text{s}}(0, \sum_{i=0}^{m-1} W_i \cdot T_{\text{slot}}) \tag{10}$$

where $U_{\text{s}}(\cdot)$ represents the probability distribution of the overall backoff time T_{backoff}, which is the sum of m uniform random variables. We use $P_s(t)$ to represent the CDF of T_{backoff}, *i.e.* the probability that T_{backoff} is shorter than time t is represented by $P_s(t)$.

Finally, based on Eqs. (2)-(10), we have the state probability distribution by

$$\begin{cases} p_g(n) = P_s(\Delta T) \\ p_f(n) = P_s(T_{\text{data}} + \Delta T) - P_s(\Delta T) \\ p_b(n) = 1 - P_s(T_{\text{data}} + \Delta T) \end{cases} \tag{11}$$

where $\Delta T = (n-1)T_{\text{data}} - T_{\text{RERR}} - mT_D$.

B. Estimation of Packet Loss Probability λ_g, λ_f, and λ_b

λ_g refers to the packet loss rate of a good link, in which the ACK is received to indicate a successful transmission. Therefore, we assume $\lambda_g = 0$. λ_b is defined as the packet loss rate of an unreliable link, which is the probability that the video packet does not reach the next hop destination successfully. λ_f is the packet loss rate for the video packet that fails all transmission attempts and triggers the RERR. Based on the MAC layer mechanism, we know that each time a video packet fails a transmission, it means either the video packet fails to transmit to the next hop destination or the ACK message is not received by the transmitter. Thus, λ_f is the conditional packet loss probability for the video packet that fails all transmission attempts.

Let A_0 denote the event that the video packet is lost and A_1 denote the event that the video packet fails all transmission attempts. We assume that each transmission is independent and the loss probability of a video packet and an ACK for an unreliable link are p_{data} and p_{ACK} respectively. Then we have $p(A_0) = p_{\text{data}}^m$ and $p(A_1) = [p_{\text{data}} + (1-p_{\text{data}}) \cdot p_{\text{ACK}}]^m$, where m is the retransmission limit. Finally, λ_f and λ_b are represented by

$$\begin{aligned} \lambda_f &= p(A_0|A_1) = \frac{p(A_1|A_0) \cdot p(A_0)}{p(A_1)} \\ &= \frac{p_{\text{data}}^m}{[p_{\text{data}} + (1 - p_{\text{data}}) \cdot p_{\text{ACK}}]^m} \qquad (12) \\ \lambda_{\text{b}} &= p(A_0) = p_{\text{data}}^m \qquad (13) \end{aligned}$$

By Eqs. (12) and (13), we have

$$\frac{\lambda_b}{\lambda_f} = [p_{\text{data}} + (1 - p_{\text{data}}) \cdot p_{\text{ACK}}]^m \le 1 \tag{14}$$

i.e. λ_f is generally larger than λ_b.

V. Proposed Routing-aware MDC

Given the packet loss probability estimated from routing messages, we seek to design a routing-aware MDC method that can improve the error resilience of the reconstructed video. Since error propagation is probably the most important problem for video transmitted over error-prone channels [26], our design goal is to reduce error propagation caused by the video packet losses. Our design achieves this goal by using the reference frame selection technique to reduce error propagation, *i.e.*, select proper reference frames that do not suffer packet losses. For every video frame that may be corrupted, our design can estimate the frame corruption probability and avoid using this frame as a reference frame if the corruption probability is higher than a certain threshold. In the following, we describe the details of the frame corruption estimation and the reference selection algorithm for our MDC method.

In Section IV, we discussed the process to estimate the packet loss probability of each transmitted packet based on the routing messages. Based on the received RERR message, we estimate the packet loss probability of the n^{th} preceding packet sent from the source as $\Pr(n)$, which corresponds to the packet loss probability of the video packet with index v_i. We now use the estimated packet loss probabilities to determine the frame corruption probability of each frame by

$$p(f_k) = 1 - \prod_{v_i \in f_k} (1 - p(v_i)) \tag{15}$$

where $p(v_i)$ is the packet loss probability for packet v_i in frame f_k, and the frame corruption probability of f_k is defined by the probability that any packet in frame f_k is lost. That is, we consider the whole frame as corrupted as long as part of the frame is lost and the corrupted frame is removed from the reference frame list. This simplifies the reference frame selection algorithm. As a future work, we can further improve our algorithm by performing reference frame selection on partial frames.

We use the frame corruption estimation results to assist the reference frame selection. During the encoding process, we initialize the reference list that consists of previously encoded

Procedure 1 Encode a video packet for transmission

while have video context to send **do**
  Initiate reference list for current frame
  for all reference frames in current description **do**
    if the frame corruption probability of the reference is larger than a threshold **then**
      Remove the frame from the reference frame list
    end if
  end for
  if no frames are available in the reference list **then**
    Add available frames in the other description to the reference list
  end if
  Encode a packet of video using selected references and transmit it through one of the two paths
  if receive a RERR message that implies a link failure **then**
    Estimate the RERR delay and determine the packet loss probability for affected packets
    if a route is available in the route cache **then**
      Reconstruct a new route from the route cache
    else
      Initiate the route recovery process
      repeat
        Mark the packets scheduled to be sent through the broken route as lost
      until receive a RREP to build a new route
    end if
    Estimate the frame corruption probability based on the estimated packet loss probability
  end if
end while

frames in the same description. Next, we remove the frames with a frame corruption probability greater than a threshold p_{thres} from the list. If all frames are removed from the reference list, we check the previously encoded frames in the other description and add frames with $p(f_k) < p_{\text{thres}}$ to the list. The current frame is encoded using the reference frames in the list and transmitted over the networks. The process is shown in Procedure 1. By not using the possible damaged frame as reference, we expect to reduce error propagation due to packet losses. Moreover, our proposed approach only relies on the standard ad-hoc routing messages and it does not incur any extra overhead.

Despite minimizing frame corruption estimation errors through careful modeling, estimation errors may still occur due to the random feedback delay of routing messages. These unexpected estimation errors can reduce the gains of our design: failing to detect corrupted frames (miss-detections) can lead to error propagation, and incorrectly identifying good frames as corrupted (false-alarms) can reduce video coding efficiency. To address the estimation errors, we can change the threshold p_{thres} used for frame corruption detection to achieve a flexible tradeoff between the error resilience and coding efficiency. By configuring p_{thres}, we can adapt our design to different scenarios, *e.g.*, we can reduce p_{thres} for the applications that are sensitive to error propagation. In Section VI-B, we study the overall estimation accuracy of our proposed approach under various p_{thres} values.

TABLE I
SIMULATION PARAMETERS

Region	500 m× 500 m
Number of nodes	50
Mobility model	Random waypoint model: node speed 0 $\sim$ 10 m/s, pause time 120 s
PHY data rate	5.5 Mbps
Transmission Power	15 dBm
MAC layer protocol	802.11b CSMA/CA
Playout deadline	350 ms

VI. SIMULATION RESULTS AND DISCUSSION

In this section, we first introduce the simulation setup for our routing-aware MDC system and then discuss the performance of our proposed method under various conditions.

A. *Implementation and Simulation Setup of the Routing-aware MDC scheme*

We have simulated the routing-aware MDC system using the modified JM codec and the Qualnet simulator, and we have examined the performance of the reconstructed video at the receiver under varying network settings. First, we simulated a two-path transport system over a mobile ad-hoc network using the Qualnet simulator. The routing information received at the transmitter is recorded and fedback to the JM encoder. Based on the MAC layer parameters, routing information, and our packet loss estimation model, we estimated the packet loss probability for each packet and used it to guide the reference frame selection during encoding. Then we generated the corrupted video bitstream based on the encoded video sequence and the network simulations. Finally, we decoded the bitstream using our refined error concealment method for MDC [17]. Details of the network settings, parameters for the estimation model, and video source statistics are described below.

1) Network Settings: We use a QualNet simulator to evaluate our routing-aware MDC over a mobile ad-hoc network. Unless otherwise specified, we choose network parameters as shown in Table I. In the ad-hoc network, nodes are uniformly placed in a $500m \times 500m$ region, where the connectivity of any two nodes is determined by the network topology and the communication range. The movement of each node is characterized by a random waypoint model [27] with parameters shown in the table. A pair of source and destination nodes is randomly chosen to transmit video packets. We use IEEE 802.11b, which employs CSMA/CA as the MAC layer protocol and we implement SMR as the multipath routing protocol. Packets are dropped if they do not reach the destination by the playout deadline of 350 ms.

QualNet uses a wireless communication medium model to simulate the propagation of signals between nodes [28]. This model takes into account propagation delays and signal

TABLE II
PARAMETERS OF IEEE 802.11B

Parameter	Value
Slot time T_{slot}	20 μs
PHY header	192 bits
MAC header	224 bits
ACK packet	112 bits + PHY header
DIFS time T_{DIFS}	50 μs
SIFS time T_{SIFS}	10 μs
$\text{CW}_{\min}$	31
$\text{CW}_{\max}$	1023
Retransmission limit	7

attenuation due to path loss, fading, and shadowing. In our simulation, we choose a two-ray path loss model that considers a line-of-sight path and a reflection from flat earth in the pathloss calculation. We use a Rayleigh fading model to calculate the effect of a propagation path on the signal strength and lognormal shadowing model to calculate the signal attenuation caused by obstruction on a propagation path.

2) Parameters for The Estimation Model: In Section IV, we propose a statistical model to estimate the packet loss probability of each packet transmitted over the networks based on the routing messages and MAC layer parameters. We assume that all the nodes in the network employ the DCF basic access mechanism for packet transmission. The parameters for packet loss estimation are shown in Table II.

3) Video Source and Performance Metrics: We consider five video sequences “Foreman”, “Coastguard”, “Mother-daughter”, “News”, and “Silent”, which are all at CIF format with 150 frames at a frame rate of 15 fps. The video sequences are encoded into RTP packets with a packet size of 500 bytes. We generate two descriptions for each video sequence and the bitrate of each video sequence is 400 kbps, which corresponds to a bitrate of 200 kbps for each description. The two descriptions are transmitted through two paths over the network. For each network scenario, each video sequence is sent repeatedly 500 times to generate statistically meaningful quality measures.

We use average PSNR of all frames over all realizations to evaluate the objective video quality of the decoded video sequences. In addition, we introduce $\text{PSNR}_{r,f}$ proposed in [15], [16] to evaluate the perceptual video quality because video sequences with close average PSNR may reveal different perceptual video quality for human viewers due to the non-linear behavior of the human visual system.

$\text{PSNR}_{r,f}$ is defined as the PSNR achieved by $f\%$ of the video frames for $r\%$ of the realizations, which shows the video quality guaranteed for $r\%$ of realizations among $f\%$ of the frames. The definition of $\text{PSNR}_{r,f}$ can be written as

$$\text{PSNR}_{r,f} = \arg_x P_{\text{real}}(P_{\text{frame}}(\text{PSNR} > x) \geq f) \geq r \quad (16)$$

where $P_{\text{frame}}(\text{PSNR} > x)$ is the percentage of frames that have PSNR higher than x in one realization and $P_{\text{real}}(\Omega)$ is the percentage of realizations that satisfy the condition Ω. For example, $\text{PSNR}_{r=80\%,f=90\%} = 35$ dB means that there are 80% of the realizations having 90% of the frames with PSNR higher than 35 dB.

TABLE III
AVERAGE PSNR FOR CODED FOREMAN SEQUENCE AT 400 KBPS WITHOUT AND WITH TRANSMISSION LOSSES

PSNR (dB)	SDC	MDC	RA-MDC
Without losses	35.77	34.56	34.50
With losses (p=4.5%)	32.20	32.45	33.51

We use $\text{PSNR}_{r,f}$ to evaluate the perceptual video quality because of two findings [15], [16], [29]: (1) The bad-quality frames dominate users' experience with the video; (2) For PSNRs higher than a certain threshold, increasing PSNR does not help to enhance the perceptual video quality. We know that average PSNR treats every frame equally and does not perfectly correlate with the perceptual video quality because of the non-linear behavior of the human vision system. While $\text{PSNR}_{r,f}$ can capture the performance loss due to damaged frames in a video sequence ($f\%$).

Furthermore, $\text{PSNR}_{r,f}$ captures the performance experienced by a user for multiple uses ($r\%$) of the network, or alternatively, it can be interpreted as a performance indicator for multiple users ($r\%$) of the network.

B. Results and Discussion

Using the simulation setup described in Section VI-A, we simulated the routing-aware MDC (RA-MDC) method with MPT and compared the end-to-end performance with single description coding (SDC) and MDC with MPT. For SDC and MDC , we use the same MPT strategy such that even and odd frames are transported through two separate routes. For both RA-MDC and MDC methods, we apply the refined error concealment method proposed in [17] to conceal MB-level losses while the frame loss concealment method in JM is applied to conceal frame-level losses. We first study the overall video performance of the three methods and we examine the estimation accuracy of our proposed model under various network settings.

1) Overall Performance: First, we examine the case that the transmission power of each node is 15 dBm and the overall packet loss rate in the network is around 4.5%. We show the PSNR values of each frame in one realization in Fig. 4. When the packets are transmitted successfully, SDC achieves slightly higher PSNR than MDC and RA-MDC because the coding efficiency of MDC and RA-MDC decreases due to the decreased correlation between adjacent frames. When packet loss happens, the PSNR value of the corrupted frame drops and the errors propagate to all subsequent frames of SDC (dashed line with triangle marker) until an I-frame is received. For MDC (dash-dot line with cross marker), the errors only propagate in the description on the broken route and the PSNR oscillates as shown in Fig. 4. Meanwhile, the RERR packets indicate the packet losses in the network fairly accurately and our proposed RA-MDC (solid line with point marker) method can effectively stop the error propagation in the subsequent frames.

Then we look at the PSNR performance of the three methods without and with transmission losses in Table III. We see that for the coded Foreman sequence without transmission

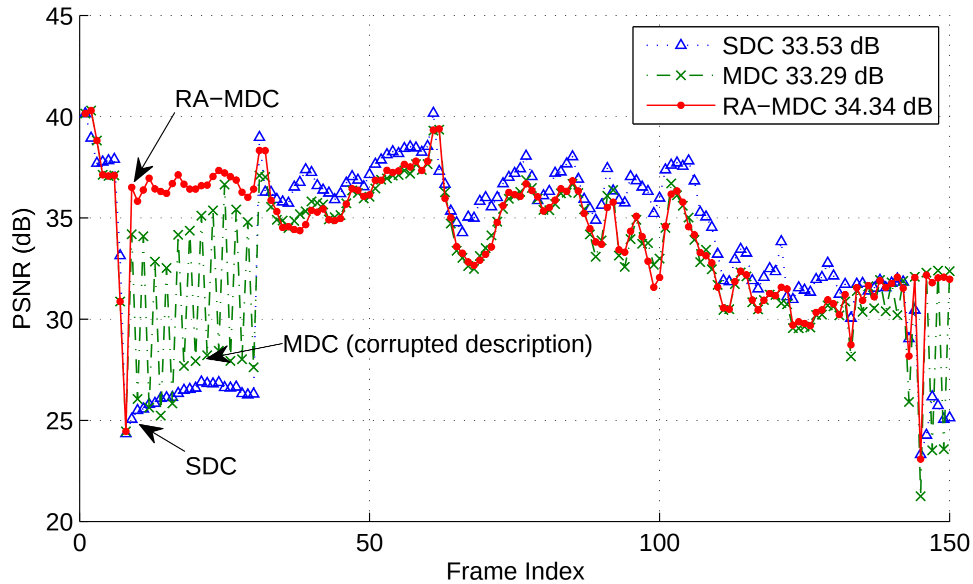
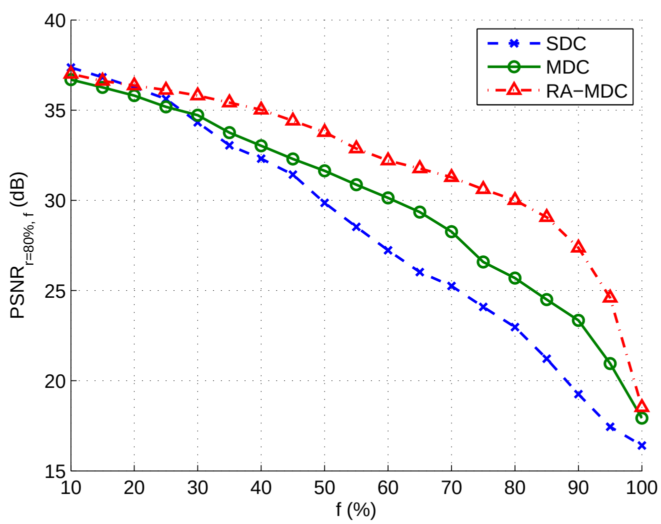
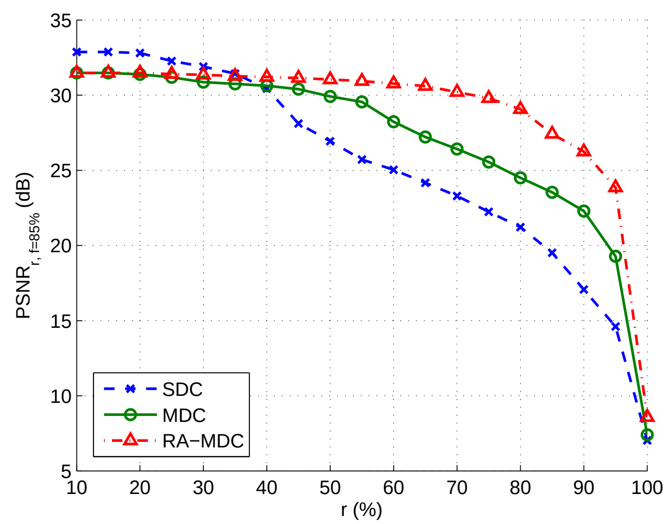


Fig. 4. PSNRs of each frame in one realization for Foreman sequence (CIF, 15fps) coded at 400 kbps. Average PSNRs for SDC, MDC, and RA-MDC in this realization are 33.53 dB, 33.29 dB, and 34.34 dB respectively



(a) $\text{PSNR}_{r=80\%,f}$



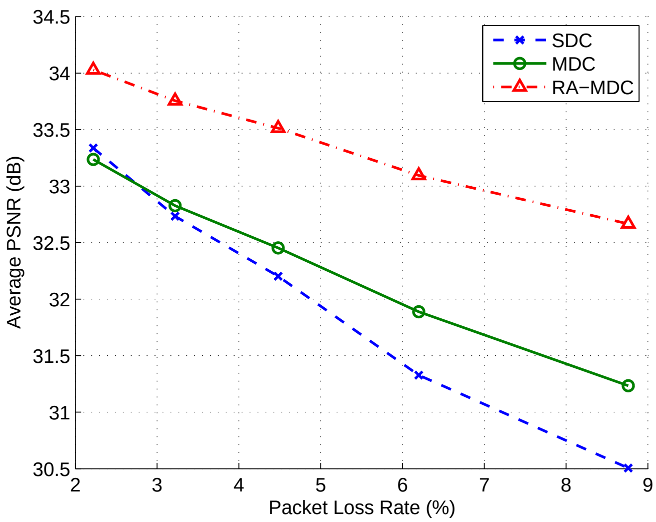
(b) $\text{PSNR}_{r,f=85\%}$

Fig. 5. Comparing $\text{PSNR}_{r,f}$ of SDC, MDC, and RA-MDC, Foreman sequence (CIF, 15fps) at 400 kbps, packet loss rate 4.5%. Average PSNRs of SDC, MDC, and RA-MDC are 32.20 dB, 32.45 dB, and 33.51 dB respectively

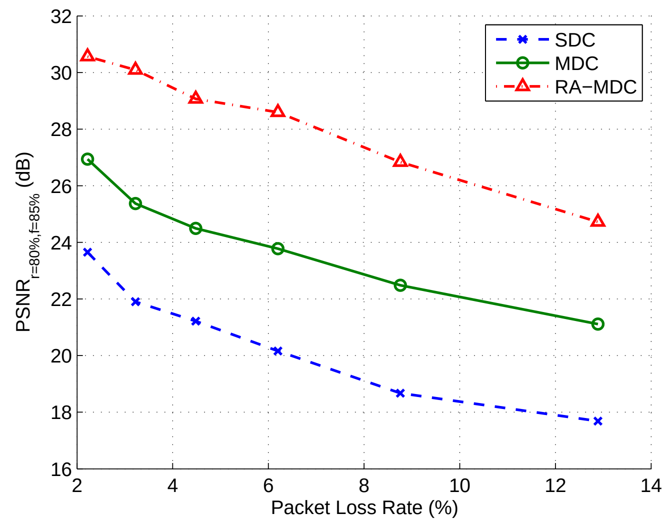
losses, SDC achieves highest PSNR under the same bitrate, while MDC has a PSNR slightly higher than RA-MDC. On the other hand, RA-MDC achieves the highest average PSNR in the presence of moderate transmission losses. The results show that both MDC and RA-MDC trade coding efficiency for the reconstructed video quality under transmission losses, while RA-MDC provides a better tradeoff between coding efficiency and error resilience. Based on the frame loss estimation in RA-MDC, fewer frames are used as reference for RA-MDC, which leads to a 0.06 dB lower PSNR than MDC when there is no transmission loss. However, the RA-MDC achieves 1 dB gain in PSNR under transmission losses, since it can effectively stop error propagation by not using corrupted frames as reference.

As discussed in Section VI-A3, average PSNR among all frames over all realizations does not correlate very well with the perceptual video quality. Therefore, we present $\text{PSNR}_{r,f}$ [15] to assess better the perceptual video quality of the three methods.

Figure 5(a) and 5(b) compare the $\text{PSNR}_{r,f}$ results for Foreman sequence with fixed values of r and f, respectively. Figure 5(a) shows the $\text{PSNR}_{r,f}$ values for the three coding methods with fixed $r = 80\%$, which indicates the delivered video quality guaranteed for 80% of the users for f percentage of the frames. In Fig. 5(a), we see that about 28%, 16%, and 5% of the frames in 80% of the realizations have a PSNR lower than 25 dB for SDC, MDC, and RA-MDC, respectively. This shows that RA-MDC has the fewest bad-quality frames for 80% of the channel uses. Figure 5(b) presents $\text{PSNR}_{r,f}$ for SDC, MDC, and RA-MDC with fixed $f = 85\%$. This figure shows that RA-MDC guarantees a better video quality for most



(a) Average PSNR vs packet loss rate



(b) $\text{PSNR}_{r=80\%,f=85\%}$ vs packet loss rate

Fig. 6. Performance under different packet loss rates for Foreman sequence (CIF, 15fps) at 400 kbps. Transmission power varies from 10 dBm to 15 dBm to achieve different packet loss rates.

TABLE IV
PERFORMANCE FOR DIFFERENT VIDEO SEQUENCES UNDER DIFFERENT PACKET LOSS RATE

	packet loss rate 4.5%						packet loss rate 8.8%					
	PSNR			$\text{PSNR}_{r=80\%,f=85\%}$			PSNR			$\text{PSNR}_{r=80\%,f=85\%}$		
	SDC	MDC	RA-MDC	SDC	MDC	RA-MDC	SDC	MDC	RA-MDC	SDC	MDC	RA-MDC
Coastguard	28.30	28.38	29.03	21.42	23.77	27.76	27.16	27.62	28.52	18.79	22.30	25.16
Foreman	32.20	32.45	33.51	21.22	24.49	29.07	30.51	31.23	32.67	18.67	22.49	26.84
Mother-daughter	39.40	39.86	40.63	27.86	34.20	38.66	37.91	38.91	39.99	24.29	32.00	34.28
News	37.24	37.50	38.53	22.47	29.63	36.06	35.43	36.38	37.72	20.42	27.89	31.41
Silent	35.13	35.20	35.89	24.93	29.66	33.85	33.61	34.32	35.24	20.32	26.59	29.32

of the realizations compared to the other two methods. For example, RA-MDC guarantees a PSNR of 29.07 dB for 85% of the frames in 80% of the realizations, while SDC and MDC can only guarantee a PSNR of 24.50 dB and 21.22 dB for the same values of r and f. This indicates that RA-MDC provides better video quality for most of the users over the network.

Next, we examine the performance of the three methods under different packet loss rates. In the simulations, we varied the transmission power from 15 dBm to 10 dBm to achieve packet loss rates in the range of 2.2%-12.9%. Figure 6(a) presents the average PSNR under different packet loss rates. We see that the gains in PSNR provided by RA-MDC increase as the packet loss rate increases. This is because the gain of error resilience for RA-MDC overcomes the reduction in coding efficiency. In Fig. 6(b), we present $\text{PNSR}_{r,f}$ under different packet loss rates with $r = 80\%$ and $f = 85\%$. The results show that RA-MDC outperforms MDC by about 5 dB and outperforms SDC by about 8 dB, which indicates that RA-MDC provides better video quality for most of the users under various packet loss rates.

Finally, we present the performance of the three methods for five different video sequences under two typical packet loss rates in Table IV. These results show that RA-MDC achieves gains in PSNR in the range of 0.7-2.3 dB compared to SDC, and gains in PSNR in the range of 0.7-1.4 dB compared to MDC. Furthermore, RA-MDC increases $\text{PSNR}_{r=80\%,f=85\%}$ by up to 13.6 dB as compared to SDC and by up to 6.4 dB as compared to MDC.

2) Model Estimation Accuracy: We showed that RA-MDC improves both objective and perceptual video quality of delivered videos in Section VI-B1. This indicates that our proposed method can accurately estimate the frame corruption based on the routing messages and network parameters. Now we examine the performance of our estimation process under various network settings to verify its robustness.

We can consider the frame corruption estimation problem as a binary classification problem, in which we try to determine whether a frame is corrupted or not based on the routing information and network conditions. Therefore, we can run a binary hypothesis test to evaluate the performance of our frame corruption estimation. There are two hypotheses: H_0 corresponds to the situation that a frame is correctly received; H_1 corresponds to the situation that a frame is corrupted. Based on our estimation model, we have our estimation outcomes A_0 and A_1, where A_0 means we estimate the frame to be correctly received and A_1 means we treat the frame as corrupted. Then we can use two error probabilities to measure the accuracy of our estimation. $P_{\text{FA}} = P(A_1|H_0)$ is referred to as a false alarm, which corresponds to the probability of detecting a corrupted frame when the frame is actually correctly received. $P_{\text{MISS}} = P(A_0|H_1)$ is referred to as a miss-detection, which corresponds to the probability of detecting a

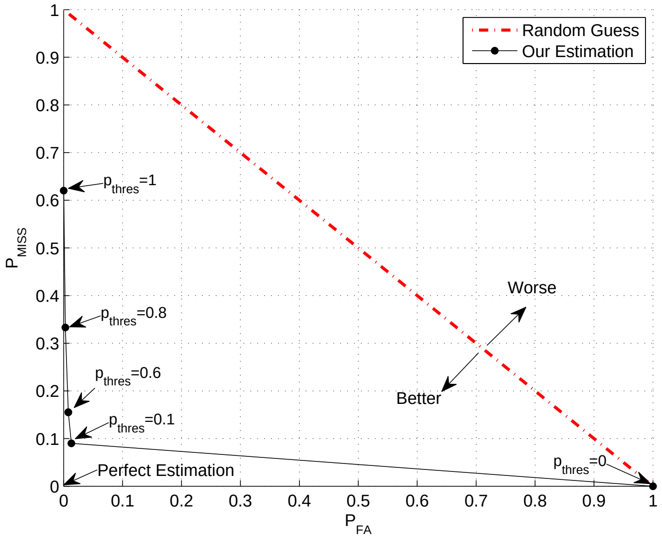


Fig. 7. ROC curve for $p_{\text{thres}} \in [0, 1]$. Varying p_{thres} provides a trade-off between P_{FA} and P_{MISS}.

TABLE V
P_{FA}, P_{MISS} UNDER DIFFERENT TRANSMISSION POWERS

Transmission Power	P_{FA}	P_{MISS}
15 dBm	0.008	0.155
14 dBm	0.010	0.195
13 dBm	0.012	0.229
12 dBm	0.014	0.267
11 dBm	0.016	0.315
10 dBm	0.019	0.372

correctly received frame when the frame is actually corrupted.

First, we plot a receiver operating curve (ROC) [30] to represent the possible values of P_{FA} and P_{MISS} under various p_{thres} in the range of [0, 1] in Fig. 7. This figure is generated using the default network settings presented in Section VI-A with transmission power of 15 dBm. In this ROC space, the (0, 0) point represents perfect estimation and the diagonal line denotes a completely random guess. Therefore, the closer the point is to the lower left corner, the better the overall accuracy of the estimation. In Fig. 7, we see that the ROC curve of our estimation is close to the lower left corner for the p_{thres} values in the range of [0.1, 0.6], where we achieve a very low probability of a false classification while maintaining a fairly low probability of missing a corrupted frame. This suggests that by choosing a p_{thres} value in that range, our estimation method yields fairly good performance.

Next, we examine the accuracy of our estimation method under different network settings. Here we choose $p_{\text{thres}} = 0.5$. Table V presents P_{FA} and P_{MISS} values under different transmission powers. We see that the false alarm probability is constantly low under different transmission powers, which leads to negligible unnecessary reduction in coding efficiency. Meanwhile, the miss-detection probability indicates that our proposed estimation method can detect most of the corrupted frames. We notice that the performance of the estimation becomes worse as the transmission power reduces. This is because when the transmission power decreases, the network connectivity becomes worse. The loss probability of routing

TABLE VI
P_{FA} AND P_{MISS} UNDER DIFFERENT NUMBER OF NODES

Number of Nodes	P_{FA}	P_{MISS}
20	0.008	0.144
40	0.008	0.164
50	0.008	0.155
60	0.008	0.169
80	0.008	0.164

messages may increase, which leads to more miss-detections.

We also investigate the impact of the number of nodes on our estimation accuracy. As shown in Table VI, varying the number of nodes has little impact on the false alarm probability and miss-detection probability. Similarly, our experiments show that the estimation accuracy is insensitive to the network size, transmission bitrate, and number of retransmissions. Thus, we omit these results due to space limitations.

VII. Conclusions

We propose a routing-aware MDC approach with multipath transport to enhance the error robustness of video transmission over wireless ad-hoc networks. We establish a model to estimate the packet loss probability of each packet based on routing messages and network parameters. Then we use the estimated packet loss probability to select the proper reference frames for MDC in order to reduce error propagation. Our proposed method does not require any additional feedback channel or extra overhead while it nicely captures the potential frame corruption during transmission.

We examine our proposed RA-MDC method using a modified JM coder and the QualNet simulator. The simulation results show that our method achieves up to 2.3 dB gains in PSNR for different video sequences under different network conditions. Using PSNR$_{r,f}$ as a multiuser perceptual quality measure, the results also indicate that RA-MDC guarantees better perceptual video quality for multiple users. In addition, we show that our proposed method has good estimation accuracy under various network settings, which leads to the improvement of the delivered video.

References

[1] Y. Wang, A. R. Reibman, and S. Lin, "Multiple description coding for video delivery," *Proceedings of the IEEE*, vol. 93, no. 1, pp. 57–70, Jan. 2005.

[2] J. G. Apostolopoulos, "Reliable video communication over lossy packet networks using multiple state encoding and path diversity," in *SPIE Proceedings on Visual Communications and Image Processing*, vol. 4310, no. 1, 2001, pp. 392–409.

[3] N. Gogate, D. M. Chung, S. S. Panwar, Y. Wang, F. N. Commun, and P. River, "Supporting image and video applications in a multihop radio environment using path diversity and multiple description coding," *IEEE Transactions on Circuits and Systems for Video Technology*, vol. 12, no. 9, pp. 777–792, 2002.

[4] S. Mao, S. Lin, S. S. Panwar, Y. Wang, and E. Celebi, "Video transport over ad hoc networks: multistream coding with multipath transport," *IEEE Journal on Selected Areas in Communications*, vol. 21, no. 10, pp. 1721–1737, 2003.

[5] B. A. Heng, J. G. Apostolopoulos, and J. S. Lim, "End-to-end rate-distortion optimized MD mode selection for multiple description video coding," *EURASIP Journal on Applied Signal Processing*, 2006.

[6] V. A. Vaishampayan and S. John, "Balanced interframe multiple description video compression," in *Proceedings of International Conference on Image Processing*, vol. 3, 1999, pp. 812 –816.

[7] T. Nguyen and A. Zakhor, "Matching pursuits based multiple description video coding for lossy environments," in *Proceedings of International Conference on Image Processing*, vol. 1, Sept. 2003, pp. I–57–60.

[8] Y.-C. Lee, Y. Altunbasak, and R. M. Mersereau, "A drift-free motion-compensated predictive encoding technique for multiple description coding," in *Proceedings of International Conference on Multimedia and Expo*, vol. 3, July 2003, pp. III–581–584.

[9] A. C. Begen, Y. Altunbasak, and O. Ergun, "Multi-path selection for multiple description encoded video streaming," in *IEEE International Conference on Communications*, vol. 3, May 2003, pp. 1583–1589.

[10] S. Mao, Y. T. Hou, X. Cheng, H. D. Sherali, S. F. Midkiff, and Y. Q. Zhang, "On routing for multiple description video over wireless ad hoc networks," *IEEE Transactions on Multimedia*, vol. 8, no. 5, pp. 1063–1074, 2006.

[11] S. Kompella, S. Mao, Y. T. Hou, and H. D. Sherali, "Path selection and rate allocation for video streaming in multihop wireless networks," in *Proceedings of Military Communication Conference*, Oct. 2006, pp. 1–7.

[12] S. Murthy, P. Hegde, V. Parameswaran, B. Li, and A. Sen, "Improved path selection algorithms for multipath video streaming in wireless ad-hoc networks," in *Proceedings of Military Communications Conference*, 2007, pp. 1–7.

[13] S. Lin, S. Mao, Y. Wang, and S. Panwar, "A reference picture selection scheme for video transmission over ad-hoc networks using multiple paths," in *Proceedings of IEEE International Conference on Multimedia and Expo*, Aug. 2001, pp. 96–99.

[14] Y. J. Liang, E. Setton, and B. Girod, "Channel-adaptive video streaming using packet path diversity and rate-distortion optimized reference picture selection," in *IEEE Workshop on Multimedia Signal Processing*, Dec. 2002, pp. 420–423.

[15] J. Hu, S. Choudhury, and J. D. Gibson, "PSNR$_{r,f}$: Assessment of delivered AVC/H. 264 video quality over 802.11a WLANs with multipath fading," in *Proceedings of the First Multimedia Communications Workshop*, 2006.

[16] J. Hu, S. Choudhury, and J. Gibson, "Video capacity of WLANs with a multiuser perceptual quality constraint," *IEEE Transactions on Multimedia*, vol. 10, no. 8, pp. 1465–1478, Dec. 2008.

[17] Y. Liao and J. D. Gibson, "Refined error concealment for multiple state video coding over ad hoc networks," in *Proceedings of the 42nd Asilomar Conference on Signals, Systems and Computers*, Oct. 2008, pp. 2243 –2247.

[18] A. Nasipuri and S. R. Das, "On-demand multipath routing for mobile ad hoc networks," in *Proceedings of Eight International Conference on Computer Communications and Networks*, 1999, pp. 64 –70.

[19] S. J. Lee and M. Gerla, "Split multipath routing with maximally disjoint paths in ad hocnetworks," in *IEEE International Conference on Communications*, vol. 10, 2001, pp. 3201–3205.

[20] M. K. Marina and S. R. Das, "On-demand multipath distance vector routing in ad hoc networks," in *Ninth International Conference on Network Protocols*, Nov. 2001, pp. 14 – 23.

[21] Z. Ye, S. V. Krishnamurthy, and S. K. Tripathi, "A framework for reliable routing in mobile ad hoc networks," in *Twenty-Second Annual Joint Conference of the IEEE Computer and Communications*, vol. 1, 2003, pp. 270 – 280.

[22] D. B. Johnson and D. A. Maltz, "Dynamic source routing in ad hoc wireless networks," in *Mobile Computing*, T. Imielinski and H. F. Korth, Eds. Springer, 1996, vol. 353, pp. 153–181.

[23] S. Mueller, R. P. Tsang, and D. Ghosal, "Multipath routing in mobile ad hoc networks: Issues and challenges," in *Performance Tools and Applications to Networked Systems*. Springer, 2004, vol. 2965, pp. 209–234.

[24] Y. Liao and J. D. Gibson, "Routing-aware multiple description video coding over wireless ad-hoc networks using multiple paths," in *International Conference on Image Processing (ICIP)*, Sep. 2010.

[25] "IEEE Standard Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications," *IEEE Std 802.11-2007 (Revision of IEEE Std 802.11-1999)*, pp. C1–1184, 2007.

[26] Y. J. Liang, J. G. Apostolopoulos, and B. Girod, "Analysis of packet loss for compressed video: Effect of burst losses and correlation between error frames," *IEEE Transactions on Circuits and Systems for Video Technology*, vol. 18, no. 7, pp. 861 –874, July 2008.

[27] J. Broch, D. A. Maltz, D. B. Johnson, Y. C. Hu, and J. Jetcheva, "A performance comparison of multi-hop wireless ad hoc network routing protocols," in *Proceedings of the 4th annual ACM/IEEE international conference on Mobile computing and networking*, 1998, pp. 85–97.

[28] *QualNet 4.5.1 Programmers Guide*, Scalable Network Technologies, July 2008.

[29] Z. Wang, H. R. Sheikh, and A. C. Bovik, "Objective video quality assessment," *The Handbook of Video Databases: Design and Applications*, pp. 1041–1078, 2003.

[30] R. Yates and D. Goodman, *Probability and Stochastic Processes: A Friendly Introduction for Electrical and Computer Engineers.* John Wiley & Sons, 2004.

Yiting Liao is a Ph.D. candidate in the Department of Electrical and Computer Engineering at University of California, Santa Barbara, under the supervision of Prof. Jerry D. Gibson. She received the B.E. and M.S. degrees from the Department of Electronic Engineering at Tsinghua University, China in 2004 and 2006, respectively. Her research interests are in the area of video compression and communications, with a focus on error resilient video coding and multiple description coding over wireless networks.

Jerry D. Gibson is Professor and Department Chair of Electrical and Computer Engineering at the University of California, Santa Barbara. He is co-author of the books Digital Compression for Multimedia (Morgan-Kaufmann, 1998) and Introduction to Nonparametric Detection with Applications (Academic Press, 1975 and IEEE Press, 1995) and author of the textbook, Principles of Digital and Analog Communications (Prentice-Hall, second ed., 1993). He is Editor-in-Chief of The Mobile Communications Handbook (CRC Press, 2nd ed., 1999), Editor-in-Chief of The Communications Handbook (CRC Press, 2nd ed., 2002), and Editor of the book, Multimedia Communications: Directions and Innovations (Academic Press, 2000).

Dr. Gibson was Associate Editor for Speech Processing for the IEEE Transactions on Communications from 1981 to 1985 and Associate Editor for Communications for the IEEE Transactions on Information Theory from 1988-1991. He was President of the IEEE Information Theory Society in 1996, and served on the Board of Governors of the IT Society for 10 years. He was a member of the Speech Technical Committee of the IEEE Signal Processing Society from 1992-1994. Dr. Gibson served as Technical Program Chair and Founder of the 1999 IEEE Wireless Communications and Networking Conference, Technical Program Chair of the 1997 Asilomar Conference on Signals, Systems, and Computers, and General Co-Chair of the 1993 IEEE International Symposium on Information Theory. He was an elected Member-at-Large on the Communications Society Board of Governors for 2005-2007. Currently, he serves on the Steering Committee for the Wireless Communications and Networking Conference and on the Editorial Board of the Hindawi Open Access journal Advances in Multimedia. He was an IEEE Communications Society Distinguished Lecturer for 2007-2008, and he is a member of the IEEE Awards Committee and the IEEE Medal of Honor Committee.

In 1990, Dr. Gibson received The Fredrick Emmons Terman Award from the American Society for Engineering Education, and in 1992, was elected Fellow of the IEEE for contributions to the theory and practice of adaptive prediction and speech waveform coding. He was co-recipient of the 1993 IEEE Signal Processing Society Senior Paper Award for the Speech Processing area. Dr. Gibson received the IEEE Technical Committee on Wireless Communications Recognition Award in 2009 for contributions in the area of "Wireless Communications Systems and Networks", and he and his students received the IEEE Transactions on Multimedia Best Paper Award in 2010.

His research interests include data, speech, image, and video compression, multimedia over networks, wireless communications, information theory, and digital signal processing.