

Control Based Mobile Ad-hoc Networks for Video Communications

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Abstract — *This paper presents a feedback control scheme for transmission of video signals over mobile ad-hoc channels. The scheme is a combination of cross-layer (local) feedback and receiver feedback. The receiver feedback is based on the real time transport control protocol (RTCP), which is designed to provide an end-to-end feedback assessment of multihop transmission channels by tracking packet receptions in synchronization with the video frame rate. The control packet carries an overlapping bit-pattern in order to cope with losses on the reverse link. Assisting the receiver feedback is a local feedback, which aims at controlling the packet transmission flow with respect to the ad-hoc routing characteristics. The combined feedback scheme, together with a bitrate control and packet-loss compensation strategy, have been shown to be very effective in improving the ad-hoc network reliability for transmission of H.264/RTP/UDP/IP packets over multihop fading channels¹.*

Index Terms — **ad-hoc video, MANET, multihop, RTP, feedback control, cross-layer, CSMA/CA, H.264, IEEE 802.11**

I. INTRODUCTION

The popularity of IEEE 802.11 [1] and recent advances in ad-hoc routing protocols, have created a strong union in developing ad-hoc networks for many applications which may require transmission of real-time voice and video information [2], [3], [4]. The main difficulty is that the IEEE 802.11 uses the Carrier Sense Multiple Access/Collision Avoidance (CSMA/CA) protocol, which is not well suited for real-time traffic support. This is, to some extent, due to the CSMA/CA backoff scheme that can cause significant delay and jitter. More importantly, in a single session multihop environment where packets are continually competing to access the media in their shared collision domains, the interference caused by nodes more than one hop away could seriously impact the end-to-end throughput performance [4]. In addition, with respect to routing protocols, these networks can also suffer from an unprecedented long delay due to dynamically changing network topologies. Such latency may result in a loss of large numbers of packets. Despite the support of IEEE 802.11 MAC retransmissions, the network performance remains unsatisfactory for delay sensitive real-time applications.

Under these circumstances additional means of packet transmission control such as feedback techniques would be essential in order to improve the reliability of these networks. Feedback schemes have been widely considered for a single

hop wireless multimedia communications [5], [6], [7], [8], and can be classified into two categories: i) cross-layer feedback and ii) receiver feedback. The general concept of the former approach, in the context of the cross layer design and optimization [9], [10], [11], [12], [13], [14] is based on exploiting the inter-dependencies between different layers such as application, routing, media access control (MAC), and the physical layer. The cross-layer approach is especially attractive for mobile ad-hoc networks, where the node mobility and network topology may change rapidly [15], [16], [17], [18], [19]. For instance, Goldsmith & Wicker studied various cross-layer scenarios for interactions amongst different layers of network stacks [15]. The interaction between the physical layer and the MAC layer for small and large wireless networks has been investigated by L. Tang et-al [16]. In addition, the issue of power control and energy-efficient cross-layer optimization has been investigated in [21], [22], [23], and continues to be a dominating research issue. However, as far as deficiencies associated with the CSMA/CA protocol is concerned, not much attention has been given as to how to handle real-time transmissions in multihop environments. Therefore, our main objective is to develop a feedback control scheme which is designed to help the application layer opt to the changes in the network topology. Our approach, in the context of the cross-layer feedback, is based on the one-way interaction between the application layer and the underplaying routing layer and will be referred to as a local feedback. In addition, the application layer also uses a receiver feedback for combating fading, shadowing, and abstractions in a situation where packets are transmitted hop-by-hop over wireless links. The receiver feedback is specifically developed to improve the encoder resiliency against packet loss.

The paper is organized as follows. Section II presents our feedback scheme, which consists of two parts: Local (cross-layer) feedback and the receiver feedback. Section III is mainly concerned with video coding strategies and how the application layer reacts to the joint feedback information. In the case of the receiver feedback, our strategy is based on identifying the exact location of the missing data on a transmitted video frame in a timely manner in order to prevent the propagation of distortion. Section IV presents the performance of the integrated feedback scheme using our real-time simulation testbed. In our experiments the H.264 video coding standard [23], and the AODV routing protocol [24] have been used to evaluate the relative performance of a joint feedback scheme under various conditions using our real-time multihop testbed.

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II. PROPOSED FEEDBACK CONTROL SCHEME

A. Local Feedback

A major obstacle for supporting real-time services ad-hoc network environments is the long delay associated with a frequent route change due to dynamically changing network topology. This includes latency for detecting a link loss as well as the time that is needed for discovering a new route. In addition, CSMA/CA access protocol, which has been widely adopted for distributed peer-to-peer multihop ad-hoc networks lacks a sustainable channel capacity as packets traverse over more hops [25]. For instance, let's consider a single session, multihop route consisting of N nodes. There would be $N-2$ intermediate nodes that function as relay nodes (i.e., node 2 to node $N-1$) with the hop-count of $N-1$.

Assume that $G_{i,j}$ denotes the power (propagation) gain on the link between node i and node j . Thus the received power P_j at node j when node i transmits with power P_i is $P_j = G_{i,j}P_i$ (1)

For wireless links, a signal transmitted with power P_t over a link with distance D gets attenuated and is received with power

$$P_r = \frac{P_t}{D^\alpha} \quad (2)$$

where α is a constant that depends on the propagation medium and antenna characteristics. Generally α satisfies $2 \leq \alpha \leq 4$. Let $\gamma_{i,i+1}$ denotes the Signal to Noise plus Interference Ratio (SINR) for the link $(i, i+1)$, which is from node i to node $i+1$. Thus,

$$\gamma_{i,i+1} = \frac{P_i/D_{i,i+1}^\alpha}{P_{Noise} + \sum_{k \neq i,i+1} X_k P_k / D_{k,i+1}^\alpha} = \frac{P_i}{D_{i,i+1}^\alpha (P_{Noise} + \sum_{k \neq i,i+1} X_k P_k / D_{k,i+1}^\alpha)} \quad (3)$$

where P_{Noise} is the background noise at node $i+1$, P_i is the transmitting power at node i , $D_{i,i+1}$ is the distance between node i and node $i+1$, and X_k is a binary variable:

$$X_k = \begin{cases} 1 & P_k / D_{k,i}^\alpha \leq S_i \\ 0 & P_k / D_{k,i}^\alpha > S_i \end{cases}, \quad 1 \leq k \leq N, k \neq i, i+1 \quad (4)$$

where S_i is the receiver sensitivity threshold for node i . For convenience, let's assume that all the nodes are equally distributed along a straight line and the distance between the two closest nodes is d . Then, $D_{k,i+1} = |i+1-k|d$.

Thus from (3), $\gamma_{i,i+1}$ can be:

$$\gamma_{i,i+1} = \frac{P_i}{(d^\alpha P_{Noise} + \sum_{k \neq i,i+1} X_k P_k / |i+1-k|^\alpha)} \quad (5)$$

where

$$X_k = \begin{cases} 1 & P_k \leq |i-k|^\alpha d^\alpha S_i \\ 0 & P_k > |i-k|^\alpha d^\alpha S_i \end{cases}, \quad 1 \leq k \leq N, k \neq i, i+1 \quad (6)$$

As we can see from (5), the amount of interference is decided by the value of X_k . For instance; in a single-hop communication (as indicated in (6)) X_k will always be 0 and consequently there will be no interference. However, if the hop count $(N-1)$ increases, $|i-k|^\alpha d^\alpha$ would also increase. In this case, the probability of $X_k = 1$ would increase with higher transmission rates which simply indicates that nodes located more than one hop away may interfere with the receiver node. This could significantly degrade the multihop link performance as the hop-count (i.e., $N-1$) becomes larger. At the same time nodes which are too many hops away will be too far to interfere with any receiving nodes along an active path (see equation 2). Therefore, the link performance is not expected to degrade notably while increasing the hop count

This behavior has been verified experimentally through simulation where the input data at a constant bitrate (CBR), is encapsulated into fixed 612 bytes UDP packets. Figure 1 shows the throughput results as a function of hop-count using a link consisting of 802.11b nodes with a bandwidth of 2 Mb/s and a bit-rate of 1.4 Mb/s. These results were obtained under the Ricean fading model with differing Ricean factors ($K=0$, $K=5$, $K=10$, $K=100$).

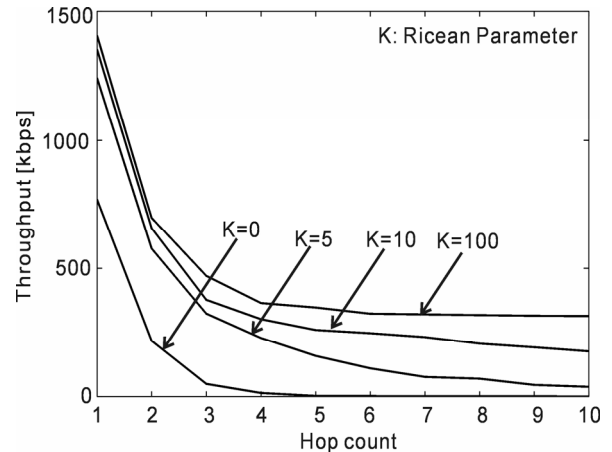


Fig. 1. Hop count versus maximum throughput performance (UDP packet size is 612-byte).

As can be observed, the throughput performance deteriorates rapidly as the hop-count increases. It should be noted that a rapid degradation of throughput going from single hop to two hops is mainly due to the intermediate node which cannot receive and transmit packets at the same time (half duplex). According to Fig. 1, in order to transmit video at the permissible rate, each time a new link is established the route information, such as hop-count, should be transferred to the application layer in order to adjust its bitrate value of the quantization parameter (QP). This parameter has been specifically defined in the syntax structure by all video-coding

standards including H.264 [23], as a means to control the video transmission rate.

Indeed, the QP value can have a direct bearing on the video quality and hence, is selected as a two-way compromise between the average transmission rate and the video quality. For a near-fixed packet-size Fig. 2 shows the average number of packets per frame (excluding I-frames) versus the QP value which is obtained for a set of motion video sequences.

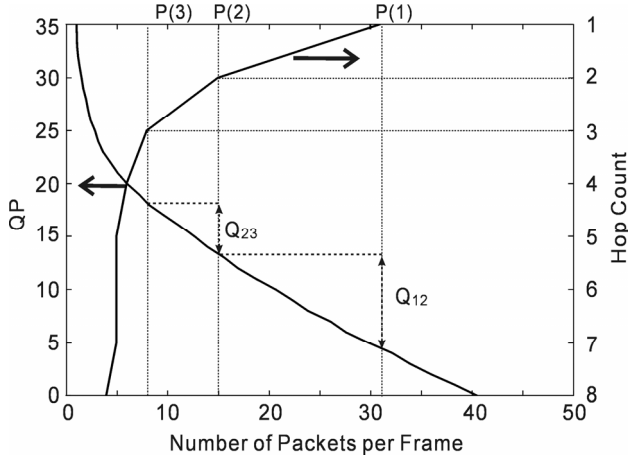


Fig. 2 Selecting of the quantization range with respect to the transitional hop-count. Q_{12} = Range of QP values for a change of hop-count from 1 to 2 (or vice versa). Q_{23} = Range of QP values for a change of hop-count from 2 to 3 (or vice versa).

As can be observed, since the QP value has a non-linear relationship with the video coding bitrate, its value can be recursively estimated to meet the permissible packet rate as [25], accordingly. For video communications, the bitrate can be adjusted by changing the

$$QP_n = QP_{n-1} + \delta QP_n(h) \quad (7)$$

Where $\delta QP_n(h)$ is the QP update at the given hop-count= h . QP_{n-1} and QP_n are the quantization values for frame $n-1$ (previous coded frame) and n (the current frame) respectively.

Now let's define,

$P(h)$ = permissible number of packets/frames at the hop-count = h (i.e., for frame n : $P_n(h) = P(h)$)

P_{n-1} : Measured number of packets on the previously coded frame

$Q_{i,j}$ = Range of QP values that can change the number of packets/frames with respect to the change of hop-count from $h = i$ to $h = j$

$\Phi_{i,j}$ = A multiplication factor whose value is determined by the change of hop-counts from i to j , where

$$\Phi_{i=j} = 1 \quad (8)$$

and

$$\Phi_{i \neq j} = \left| \frac{Q_{i,j}}{P(h=i) - P(h=j)} \right| \quad (9)$$

Based on the above definitions, at the beginning of each coding frame (i.e., frame n), the δQP is estimated as,

$$\delta QP_n(h) = \text{Integer part of } \{ \Phi_{ij} \cdot \{ P_{n-1} - P(h) \} \} \quad (10)$$

Note that the $Q_{i,j}$ is obtained according to the transitional hop-counts from i to j with respect to the permissible number of packets/frame at each hop-count. For better clarity, in Fig. 2 we have also shown an example of how the $Q_{i,j}$ (e.g., $Q_{1,2}$ or $Q_{2,3}$ in this figure) are selected (see also Fig. 1 for $K = 100$).

A.1 Link-Loss Notification

Discovering a new route each time a link failure occurs and the long delay associated with it, is another major obstacle in supporting delay sensitive video communications. In this situation, our solution is based on passing the link-loss notification to the application layer as soon as the source node receives a route error message (RERR) from the network layer. In the case of the AODV protocol, however, a route change is not always detectable at the source node. For instance, we have identified two distinct cases:

- i) when the destination node moves toward the upstream (with the hello message option),
- ii) when a local repair option is utilized (i.e., a node initiating the local repair finds a new route with equal or smaller hop-count).

To tackle this problem we have slightly modified the AODV routing protocol where the RERR message with the N-flag [24] is utilized to report a link breakage. Note that the RERR message with N-flag is originally intended in conjunction with the local repair option. With this modification, a link breakage will be reported with or without a local repair option. This has been subject to the condition that the source node receiving the RERR message with N-flag should not delete the route table entry to the destination. For example, in the first case where a node detects a route change it will send the RERR message with the N-flag (a reserved bit in the RERR message may also be used for this purpose) and a hop-count of one. For the second case, the node initiating the local repair will also send the RERR message with the N-flag, regardless of the hop-count of the newly discovered route.

It should be noted that the time in which the link information can reach the source node depends on how quickly the intermediate node can detect the link failure. In the case of the AODV protocol with the hello message option, the delay depends on the hello message interval (HELLO_INTERVAL), as well as the number of allowed losses at the reception of hello messages (ALLOWED_HELLO_LOSS). In the case of IEEE 802.11, the AODV protocol can use a fixed number of unsuccessfully MAC-layer retransmitted packets to detect a link-failure. For example, if the source node does not receive any acknowledgements after n transmission attempts, a link breakage will be triggered. In this case, the link-loss delay

depends on the maximum number of retransmission settings (i.e., Retry-limit).

B. Receiver Feedback

Under the best effort UDP, which is a preferred transport protocol for delay sensitive real-time applications, deploying a receiver feedback would be crucially important in order to assess the channel performance in a multihop transmission. In the case of Real-time Transport Protocol (RTP) [26], a receiver feedback is readily provided by its control packet protocol known as RTCP. RTCP packets are periodically transmitted (e.g., every 5 seconds) to each participant during an RTP session. Its main function is to provide a detailed representation of the exchanged packets such as end-to-end feedback information about delay jitter and packet-loss performance [26]. In addition, if the objective is to track and identify missing, packets the RTCP extended report (RTCP-XR) [27] may have to be considered.

The RTCP-XR report block (Loss RLE Report Block) is defined for network monitoring or for applications that can make use of massive raw data. The RTCP-XR packet consists of two parts: the header and the report block. The important information contained in the report block consists of the RTP sequence numbers of the first packet (begin-sequence) and the last packet (end-sequence) of the string of packets in which their status is reported by the X-R packet. The packet transmission status is reported in the form of delivered/undelivered packets (i.e., 1/0), and is placed at the bottom of the RTCP-XR in a stack of 32-bit chunks.

An XR report as defined in [27], is generated after a predefined fixed number of transmitted packets. For video applications however, the main problem is the lack of synchronization between generating the XR report and the video frame rate. This is mainly due to the variable bit-rate nature of compressed video, which could result in a different number packets/frame. Obviously, a lack of synchronization between releasing the XR packet with video sampling frequency can prolong the worst-case delay bounds. This can impact the performance of the receiver feedback in its ability to reduce the effect of distortion propagation. In the following we present our proposed frame-synchronized approach, which aims at timely detection of packet losses at the transmitting node as well as the overlapping bit-pattern to recover from the loss of an RTCP-XR packet transmitted via the reverse link.

B.1 Report Block & Overlapping Bit-pattern

In our feedback strategy, the receiver releases the XR packet after receiving the last packet in the video frame. Bear in mind that the last packet in the frame can be easily identified by either checking the M flag bit (last packet in the current frame) or the P flag bit (first packet in the next frame) in the RTP header. However, in situations where both packets are lost, it would be impossible to detect the end of frame. To overcome this, we have also considered the change of timestamp in order to detect the end of the frame (see Section III). Fig. 3 shows an

example of how the report block is generated which includes a scenario where the last packet in frame $n + 2$ and the first packet in frame $n + 3$ are lost.

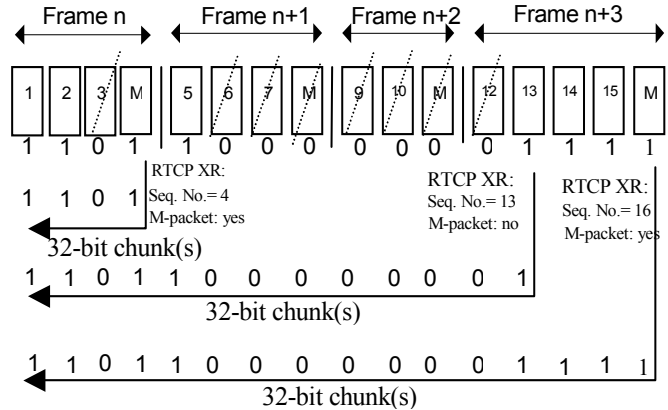


Fig. 3. Identification of delivered/undelivered packets and the generation of the bit pattern, M: M flag packet.

In this Figure, we also show the tracking information reported by the neighboring XR packet in the form of an overlapping bit-pattern. The motivation behind the proposed overlapping bit-pattern strategy is to aid in recovering losses from the previously transmitted tracking information. This arrangement can be very effective when the reverse link also suffers from poor and unreliable channels.

The inter-dependencies between the string of delivered and undelivered packets (as consecutive 0's and 1's), can be exploited using runlength coding [27]. Here, we have considered the Golomb code [23] for runlength encoding of the bit-pattern. The packaging arrangement of the runlength-coded information within a single chunk is depicted in Fig. 4. It should be noted that the beginning and end of the sequence are included in the header of the block report.

In the overlapping bit-pattern strategy it is important to accommodate many possible codewords into a chunk. Obviously, a higher runlength compression rate would contribute to a larger overlap between the successive bit-pattern. However, in some cases runlength coding might have an adverse effect on compressing the bit-pattern. This situation normally occurs when transmission between the two neighboring nodes begins to deteriorate as they moving away from each other (e.g., path-loss effect). This could results in many isolated packet drops, and tends to affect the runlength coding efficiency. Therefore, before transmitting an XR packet, if the runlength-coded bit-pattern (in a frame) is longer than 31-bit, the bit-pattern would be transmitted uncoded. Thus, the first bit in the 32-bit chunk is used to signify whether the bit-pattern is runlength coded or not.

Obviously, allocating more chunks could provide longer overlaps but this would at the expense of an increased traffic load (considering that XR packets will be transmitted within a short interval). At the same time, increasing the report interval could prolong the propagation of distortion. The impact of allocating more chunks and changing the interval for releasing each RTCP-XR packet will be evaluated in Section IV.

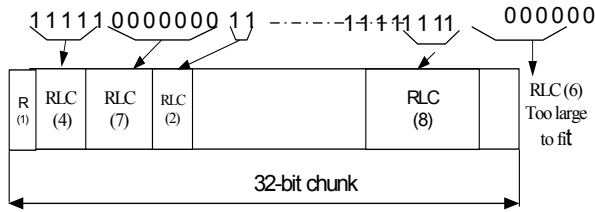


Fig. 4. RunLength Coding arrangement of the bit-pattern in a 32-bit chunk, RLC: RunLength Codeword, R: 1-bit runlength coding flag.

Finally, we should point out that it is possible to transmit XR packets only when there are packet losses to report (negative report feedback). This would certainly help reduce the traffic load, particularly under reliable and more stable transmission conditions. Its main drawback, however, is when both RTP data packets and their corresponding XR packets are lost, but the RTP data packets for the next frame are transmitted successfully. In this situation, as no XR packets will be sent, the encoder won't be able to detect any loss on the previous frame.

III. FEEDBACK CONTROL STRATEGY

The source node at the beginning of each video frame, via its application layer, evaluates the feedback information received from: i) the network layer and ii) the destination node through the XR packet. In each case, the video encoder is designed to choose different policies. In the former case, the encoder strategy, as discussed earlier, is based on proactively controlling the video transmission rate in accordance with the permissible bandwidth, which is decided by the hop-count (see Section II). For instance, in the case of the AODV (Ad-hoc On Demand Distance Vector) protocol [24], each node maintains a routing table for an entry (destination) with the hop-count (number of hops from source to destination). The source node also uses the reception of an RERR (route error) message as an indication of a link breakage. The routing information and notification about a link failure are passed to the application layer from the routing table using shared memory. In addition, as the link-failure can result in a burst of packet-drops, by anticipating that some of its transmitted packets won't reach their destination the encoder switches to the intraframe mode (I-Frame) [23] to avoid distortion propagation.

Similarly, in the case of the receiver feedback, the encoder responsibility would be to improve its resiliency knowing that some of the previously transmitted packets could not reached their destination. Since these packets carry the compressed video data, the source node should be able to identify the exact regions on the pel domain in order for the encoder to compensate the effect of missing data.

In our approach, this is accomplished by constructing a transmitter table in order to locate the exact positions of pixels on the previously transmitted video frame. Obviously, the design of the transmitter table depends on the video compression technique as well as encapsulation of the

compressed video into RTP packets. For the H.264 standard, the operation of packet loss identification via the RTCP-XR feedback is best described in the example depicted in Fig. 5. As shown, at the sender node (NODE-A) H.264 encodes each video frame at the slice-level (using a near-fixed packet size). Note that the slice output of the video encoder layer (VCL) consists of header information which includes parameter set, picture structure (progressive frame picture, top field picture, bottom field picture, etc.), slice type (Intra, Inter, B, etc.), address of the first macroblock (MB) in the slice, and so on. The coded slice is then encapsulated into an RTP packet with the NAL header [23].

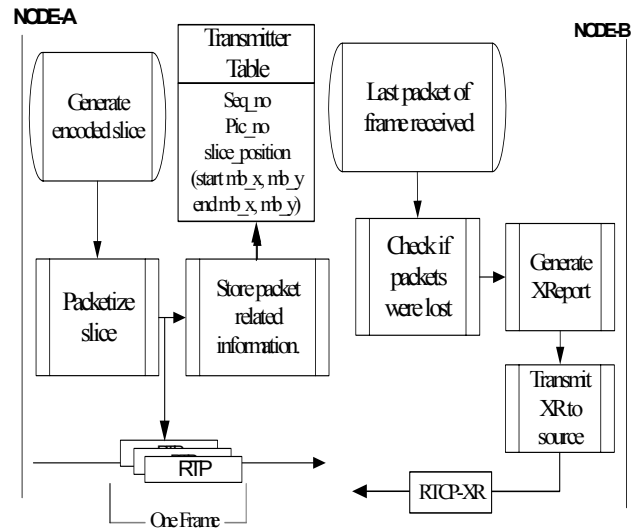


Fig. 5. Video frame synchronized RTCP-based feedback for transmission of RTP packets from Node-A to Node-B.

Prior to the packet's transmission, specific information is stored at the transmitter table with respect to the RTP packet sequence number (see transmitter table in Fig. 5). This information consists of: a) the picture number of the packetized slice, and b) the position of the slice within that picture (i.e., the first MB and the last MB of the slice). At the destination node (NODE-B), the RTCP-XR packet is generated as soon as all the successfully received packets belonging to the same frame are detected. This includes a bit pattern, which is packaged into one or more 32-bit chunks. As mentioned previously, the criteria by which the receiver can detect the end of the frame is 1) checking both the M flag in the RTP header and 2) checking RTP timestamp. For example, if the last packet in the frame (M flag packet) has also been lost, the receiver, via a change of RTP time-stamp, can easily detect that the newly arrived packet belongs to the next frame. Note that timestamp for video is derived from a 90 KHz clock reference can yield the same integer timestamp for a video signal with respect to its generic frame rate [26]. In other words, all the multiple packets from the same frame will carry the same timestamp.

We should also point out that under hostile channel environments, loss of packets might extend beyond a single

frame. In this situation, and with the aid of the stored sequence number at the transmitter table, the encoder can still identify the missing frames (including their corresponding packets).

IV. EVALUATIONS

The Qualnet network simulator has been used to evaluate the performance of the joint -feedback control scheme. The testbed is capable of grabbing live video and then transmitting H.264/RTP/UDP/IP packets from source to destination in real-time. In order to carry out these tests under the same conditions, a pre-captured video sequence with the QCIF format of 10 frame/s has been used. In addition, the compressed video was encapsulated at a near fixed packet size of 612-byte RTP packets (including the RTP header). In our experiments the bit rate for all the IEEE 802.11b devices was set to 2 Mb/s. In addition, the maximum number of retransmissions was set to 2 (Retry-Limit = 3) with a buffer size of 5000-byte for every node. To assess the performance of the feedback scheme in terms of quality of the received video we used the scenario depicted in Fig. 6.

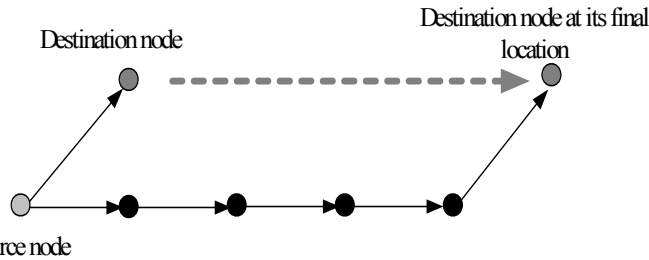


Figure 6: A change of hop-count scenario where the destination node moves from left to right undergoing one to five hops

In this scenario, the destination node moves from left to right in such a way that the hop-count changes from 1-to-5 hops. The average channel SNR between the neighboring hops was 25 dB and the Ricean fading model with the Ricean factors of $K = 10$ and $K = 5$ was used. With respect to the routing protocol, as mentioned earlier, we modified the AODV protocol in such a way that a route change is always reported to the source node.

With the above conditions, we first evaluate the performance of the local feedback using the rate adaptive control scheme. Fig. 7 compares the packet loss performance versus the number of transmission hops with and without the adaptive rate control scheme. As can be observed from this figure, without the rate control the number of missing packets goes up rapidly as video packets traverse over more hops. At the same time, with control, the packet-loss performance remains relatively undeterred by the change of hop-count but can be affected by the severity of multipath fading channel conditions ($K = 5$ compared with $K = 10$).

Indeed, in these situations, the application layer may have to rely on the receiver feedback (XR report) in order to minimize the effect of packet losses on the quality of the

received video signal. Although there are a number of options that can be considered to improve the packet-loss resiliency, we have used a low-delay packet-loss concealment scheme mainly to reduce the effect of distortion propagation of interframe coded video signals data.

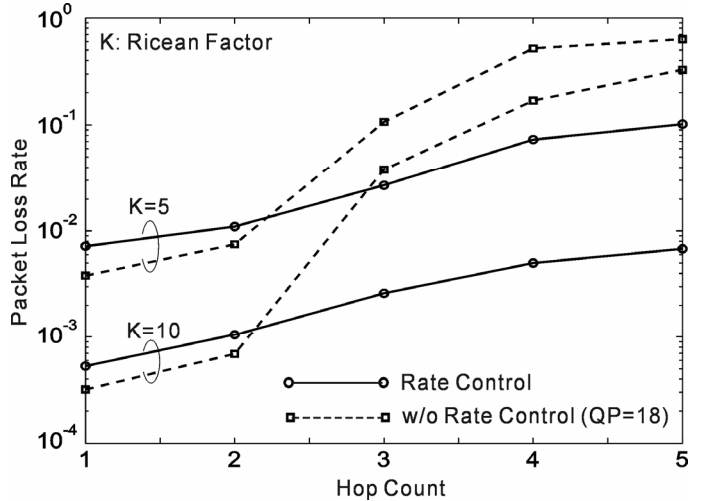


Fig. 7. Packet loss rate with rate control for $K = 10$ and $K = 5$.

Figure 8 shows a simple example of the error concealment process via the XR receiver feedback. In this example, packets have been lost during transmission of the first frame shown in Fig. 8 (i.e., frame: n) and we have assumed that the XR packet will reach the transmitting node before encoding the frame $n + 2$. According to this figure, as soon as the transmitter receives the XR packet the encoder, via its transmitter table (see Fig. 3), can identify the frame and its specific regions, which have been affected by the missing packets. Having identified the frame, the encoder first replaces the pixel values covering these regions (from previously decoded frames) in the locally decoded frame (i.e., frame n in Fig. 8). This modified frame is then used as a new reference frame to locally reconstruct the next frame (frame $n+1$ in Fig. 8). This process continues until reaching the frame $n+2$, which has not yet been encoded. Assuming this frame reaches its destination without any loss of packets, the local decoder at the transmitter and the remote decoder at the receiver are now in full coordination and the distortion can no longer affect this frame. It should be noted that the main objective behind this arrangement is to allow the distortion to propagate through the locally reconstructed frames (reference frames) in the same way as in the remote decoder.

In this example, since the XR packet does not reach the transmitting node before the frame $n+1$, this frame could not be rescued from the distortion propagation. Therefore, the number of infected frames depends entirely on the time interval in which the transmitter can receive its feedback report. It should be noted that an important factor which may affect a timely reception of the XR report is the loss of the RTCP-XR packet itself. Fortunately, with the bit-pattern overlapping strategy, on receiving the next XR packet the encoder can identify the status of the transmitted packets reported in the previous XR

packets. Nevertheless, the loss of an XR packet would extend the effect of distortion propagation.

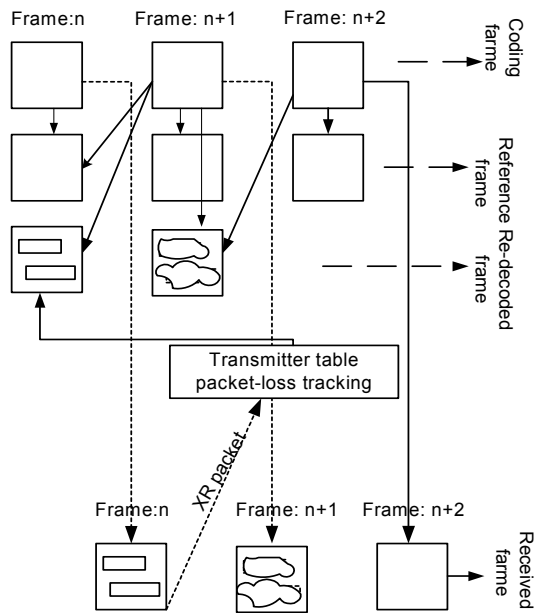


Fig. 8. An example of distortion propagation and error compensation by correcting the reference frame at the encoder

Based on the XR report, the local reference frame correction scheme has been applied to compensate the effect of missing packets. Fig. 9 and Fig. 10 show the average PSNR performance with and without incorporating the local reference frame correction. In both cases, the transmission rates are adaptively controlled by changing the quantization parameter (QP) in accordance with the hop-count. In addition, to assess the effect of excessive delay on the PSNR performance, these figures also include the results when the RTCP-XR packets are generated after every N frame (i.e., $N = 1, 3, 5, 10$). A 32-bit chunk is used to transmit the bit-pattern. Note that in these Figures the PSNR degradation at the higher hop-counts is mostly the result of the higher quantization noise in order to meet the permissible rate. In addition, worsening a fading condition could affect distortion as the packet-loss rate increases with the hop-count. With local correction, such a distortion depends on the time difference in which the status of the transmitted packets is reported back to the source node. In particular, looking at these results (see Fig. 9 and Fig. 10), we can observe a difference in the PSNR quality when XR packets are generated at differing intervals (i.e., $N = 1, 3, 5, 7, 9$).

The impact of the distortion on the temporal basis (frame-by-frame) can be more distinctly observed in Fig. 11. This Figure shows a typical PSNR performance where local correction is accomplished after receiving an XR packet within a differing frame delay period. In this experiment, a packet in the fifth frame has been deliberately removed at the destination node. As can be observed, a larger N would cause more frames to be infected before recovery.

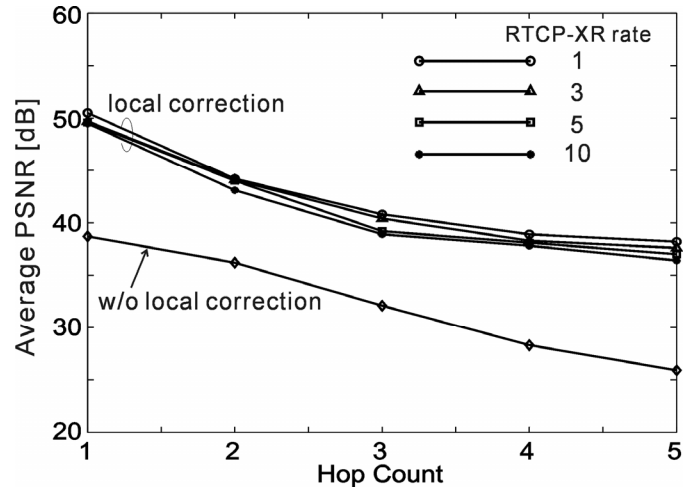


Fig. 9. Hop count versus average PSNR for different RTCP-XR rates ($K=6$).

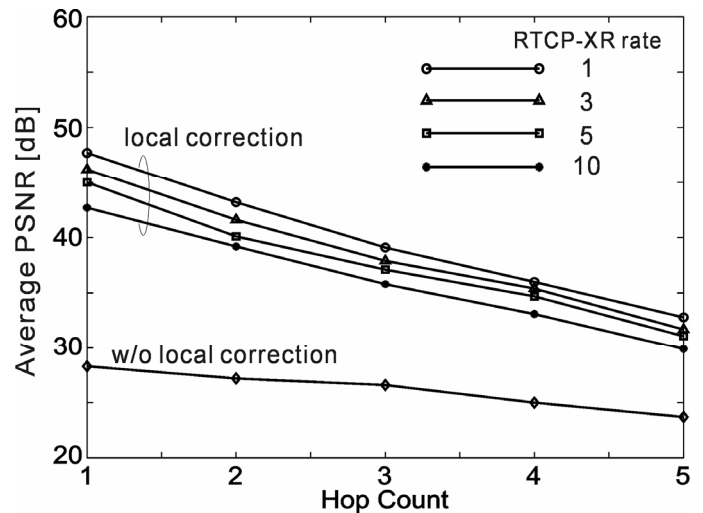


Fig. 10. Hop count versus average PSNR for different RTCP-XR rates ($K=6$).

The number of frames that can be affected by the distortion propagation also depends on the loss of XR packets and the encoder's ability to recover the tracking information from the neighboring XR packets. As explained earlier, such a recovery would be possible via the overlapping bit-pattern. In particular, under severe fading channel conditions, or in the case of a route change, a more substantial overlap would enhance the chance for recovery.

Fig. 12 shows the PSNR performance (with the help of local correction) in a typical route change situation where the receiver sends an XR packet every frame with a 32-bit chunk or 64-bit chunk. Because of a link loss, both RTP and RTCP packets are not expected to reach their destinations after the 104-th frame. As indicated in Fig. 12, the XR packet with the 64-bit chunk has successfully managed to recover the tracking information reported on the previous XR packet. On the other hand, since the 32-bit chunk did not provide a sufficient

overlap, the local correction scheme was unable to prevent the distortion propagation.

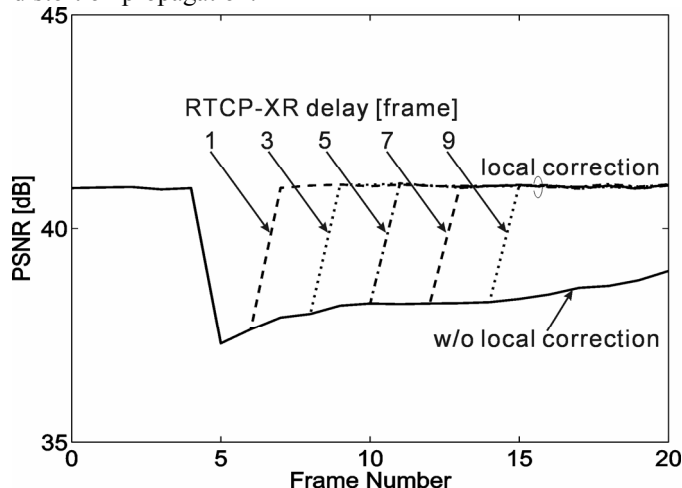


Fig. 11. Effect of the RTCP-XR delay on the distortion propagation (QP=20).

To further examine the contribution of the overlapping bit-pattern, Fig. 13 shows the effect of RTCP-XR losses on the local correction performance. In these experiments, we compared the performance of the RTCP-XR packet with a single chunk (32-bit) and the RTCP-XR packet with double chunks (64-bit), which are generated every frame ($N=1$). In order to investigate the effect of XR packet losses, we intentionally eliminated some of the RTCP-XR packets. From this figure, we can clearly observe that the chunk size can have a significant impact on the performance when we encounter the loss of RTCP-XR packets (e.g., route change). At the same time, allocating more chunks would increase the traffic load, particularly if the XR packet is transmitted at every frame. For this reason the allocation of chunks can be selected adaptively according to the cross-layer information received from the routing layer.

V. CONCLUSION

Our main objective was to develop a comprehensive feedback control scheme to improve ad-hoc networks reliability for real-time multimedia communications. A combined local feedback and a receiver feedback approach has been designed for video communications. For the receiver feedback, a frame synchronized RTCP-XR packet generation scheme, which reports the status of the received packet on a frame-by-frame basis, is developed to reduce the effect of packet loss caused by fading channel conditions in a multihop transmission. The contribution of the local feedback is mainly proactive in nature, and aims at controlling the packet transmission flow as well as reacting to the expected delay caused by the route discovery process. For transmission of RTP/UDP/IP packets, we have shown that, based on the integrated feedback approach, the quality of the received video can be significantly improved.

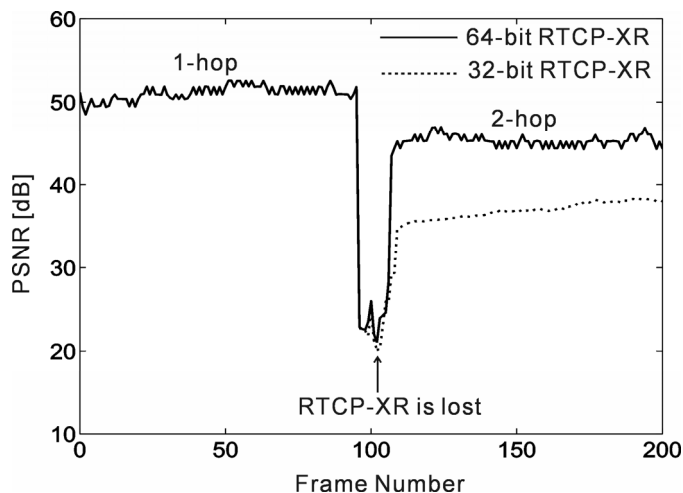


Fig. 12. Effect of RTCP-XR loss on the route change

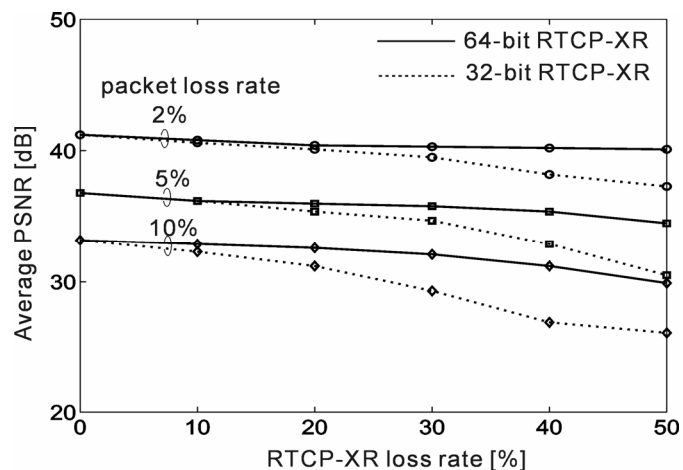


Fig. 13. Effect of RTCP-XR loss on the average PSNR performance (QP=15).

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