MRTP: A Multi-Flow Realtime Transport Protocol for Ad Hoc Networks

Shiwen Mao†, Dennis Bushmitch‡, Sathya Narayanan‡, and Shivendra S. Panwar†

†Department of Electrical and Computer Engineering
Polytechnic University, Brooklyn, New York - 11201
Email: smao@ieee.org, panwar@catt.poly.edu

‡Panasonic Information and Networking Technologies Lab,
Panasonic Technologies Company, Princeton, New Jersey - 08540
Email: db@research.panasonic.com, sathya@research.panasonic.com

Abstract— **Transporting multimedia data over ad hoc networks is a challenging problem. However, the mesh topology of ad hoc networks implies the existence of multiple paths between two nodes. In our previous work, we have shown that path diversity provides an effective means of combating transmission errors and topology changes that are typical in ad hoc networks. Moreover, data partitioning techniques, such as striping and thinning, have been demonstrated to improve the queueing performance of realtime data. Recognizing the advantages of these techniques, as well as the increasing need of video services in ad hoc networks, we propose a new transport protocol to support multipath transport of realtime data. The new protocol, called Multi-flow Realtime Transport Protocol (MRTP), provides a convenient vehicle for realtime applications to partition and transmit data using multiple flows. Analysis results from a bottleneck mobile node and simulation results from multiple path video transport over a 16-node ad hoc network illustrate the benefits of MRTP.**

I. Introduction

Ad hoc networks are wireless mobile networks without an infrastructure. Since no pre-installed base stations are required, ad hoc networks can be deployed quickly in cases such as conventions, disaster recovery, and battle fields. When deployed, mobile nodes cooperate with each other to find routes and relay packets for each other. It is foreseeable that realtime service will soon be needed in ad hoc networks once they are widely available.

It is a great challenge to provide multimedia service in ad hoc networks. A wireless link usually has higher transmission error rate because of shadowing, fading, path loss, and interference from other transmitting users. An end-to-end path found in ad hoc networks has an even higher error rate since it is the concatenation of multiple wireless links. Moreover, user mobility makes the network topology constantly change. In addition to user mobility, ad hoc networks reconfigure also when users join and leave the network. An end-to-end route in ad hoc networks may only exist for a short period of time. Realtime services have stringent delay and bandwidth requirements. Even though some packet loss is generally tolerable, the quality of reconstructed video/audio will be impaired and errors will propagate in the following frames because of the dependency introduced among consecutive frames at the encoder [1].

In our previous work, we showed that, in addition to traditional error control schemes, e.g., Forward Error Correction (FEC) and Automatic Repeat Request (ARQ), path diversity provides a new dimension for video coding and transport design [2]-[4]. Using multiple paths can provide higher aggregate bandwidth, better error resilience, and load balancing for a multimedia session. Similar observations were made in wireline networks for audio streaming [5] and video streaming using multiple servers [6]. However, we believe that multipath transport has more potential in ad hoc networks, where link bandwidth may fluctuate and paths are unreliable. In addition, multipath routing is relatively easier since many ad hoc routing protocols can return multiple paths for a route query at only limited additional cost [7]. In addition to the above advantages, data partitioning techniques, such as striping [8] and thinning [9], have been demonstrated to improve the queueing performance of realtime data. Using multiple paths for realtime transport provides a novel means of traffic partitioning and shaping. It has been shown that traffic partitioning can reduce short term correlation in realtime traffic, thus improving the queueing performance of the underlying network [9][10].

In this paper, we present a new protocol, the Multi-flow Realtime Transport Protocol (MRTP), for realtime transport over ad hoc networks using multiple paths. Given multiple paths maintained by an underlying multipath routing protocol, MRTP and its companion control protocol, the Multi-flow Realtime Transport Control Protocol (MRTCP), provide essential support for multiple path realtime transport, including session and flow management, data partitioning, traffic dispersion, timestamping, sequence numbering, and Quality of Service (QoS) reports.

One natural question arises is that can any of the current existing protocols provides the same support, i.e., do we really need such a new protocol? There are two existing protocols that are closely related to our proposal. One is the Realtime Transport Protocol (RTP) [11]. RTP is a multicast-oriented protocol for Internet realtime applications. RTP does

0-7803-7954-3/03/$17.00 ©2003 IEEE.

not support the use of multiple flows. Usually a RTP session uses a multicast tree and a whole audio or video stream is sent on each edge of the tree. Compared with RTP, MRTP provides more flexible data partitioning support and uses multiple paths for better queueing performance and better error resilience. The use of multiple flows makes MRTP more suitable for ad hoc networks, where routes are ephemeral. When multiple disjoint paths are used for a realtime session, the probability that all the paths fail simultaneously is relatively low, making better error control possible by exploiting path diversity [4]. In addition, since a wireless link's bandwidth usually fluctuates with signal strength, using multiple flows makes the realtime traffic more evenly distributed, resulting in lower queueing delay, smaller jitter, and less buffer overflow at an intermediate node. Furthermore, RTP focuses on multicast applications, where feedback is suppressed to avoid feedback explosion [11]. For example, RTP Receiver Reports (RR) or Sender Reports (SR) are sent at least 5 seconds apart. Considering the typical lifetime of an ad hoc route, this is too coarse for the sender to react to path failures. With MRTP, since only a few routes are in use, it is possible to provide much timely feedback, enabling the source encoder and the traffic allocator to quickly adapt to the path changes, e.g., mode selection for each video frame or macroblock, retransmitting a lost packet, or dispersing packets to other better paths. In fact, MRTP is a natural extension of RTP exploiting path diversity in ad hoc networks.

The other closely related protocol is the Stream Control Transport Protocol (SCTP) [12]. SCTP is a message-based transport layer protocol initially designed for reliable signaling in the Internet (e.g., out-of-band control messages for Voice over IP (VoIP) call setup or teardown). One attractive feature of SCTP is that it supports multi-homing and multi-streaming, where multiple network interfaces or streams can be used for a single SCTP session. With SCTP, generally one primary path is used and other paths are used as backups or retransmission channels. But there are several recent papers propose to adapt SCTP to use multiple paths simultaneously for data transport [13][14]. SCTP cannot be applied directly for multimedia data because there is no timestamping and QoS feedback services. With MRTP, the design is focused on supporting realtime applications, with timestamping and QoS feedback as its essential modules. Moreover, since SCTP is a transport layer protocol and is implemented in the system kernel, it is hard, if not impossible, to make changes to it. A new multimedia application, with a new coding format, a new transport requirement, etc., could only with difficulty be supported by SCTP. For MRTP, it sits in the application layer and is implemented in the user space as an integral part of an application. New multimedia services can be easily supported by defining new profiles and new extension headers. Indeed, MRTP is complementary to SCTP in supporting multimedia services using multiple paths. MRTP can establish multiple paths by using SCTP sockets, taking advantage of the multi-homing and the multi-streaming features of SCTP. In this case, one or multiple MRTP flows can be mapped to a SCTP stream. MRTP is also flexible in working with other multipath routing protocols, e.g., the Multipath Dynamic Source Routing protocol in [4], when the implementation of these protocols are ready and the interfaces are given.

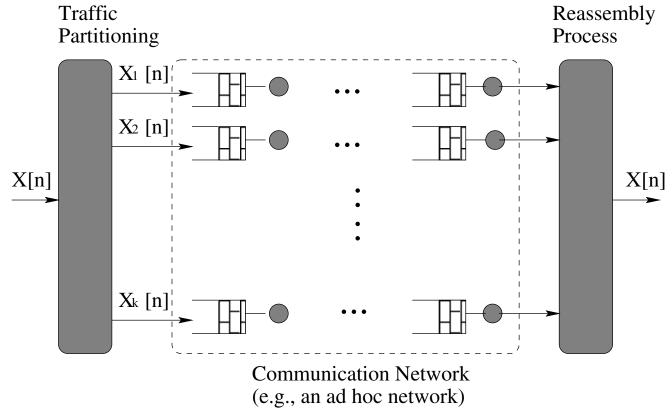

Fig. 1. Illustration of the multipath realtime transport scheme.

For the above reasons we believe a new protocol tailored to multimedia transport in ad hoc networks using multiple paths is needed. The new protocol, MRTP/MRTCP, is an extension of RTP/RTCP to exploit path diversity to combat high transmission errors and frequent topology changes found in ad hoc networks. It is also complementary to SCTP, providing the essential functionalities and the flexibility in supporting multimedia services.

The rest of the paper is organized as follows. In section II, we define MRTP and MRTCP and present their usage scenarios. Analysis and experimental results are presented in section III. Section IV outlines our future work and concludes the paper.

II. THE MULTIFLOW REALTIME TRANSPORT PROTOCOL

A. Overview

MRTP provides a framework for applications to transmit realtime data. The transport service provided by MRTP is end-to-end using the association of multiple flows. A companion control protocol, MRTCP, is also proposed for session/flow control and QoS feedback.

Figure 1 illustrates a MRTP session. After the MRTP session is set up by MRTCP, a video stream is first partitioned into several sub-streams. Each sub-stream is then assigned to one or multiple flows by a traffic allocator, and traverses a path, partially or fully disjoint, with other flows to the receiver. The receiver reassembles the multiple flows received using a resequencing buffer for each flow. Packets from the flows are put into the right order using the timestamps carried in their headers.

A possible protocol stack architecture is illustrated in Fig. 2. MRTP uses UDP datagram service or the multihoming/multistreaming transport service provided by SCTP for realtime data. We put MRTP above TCP also, because the session/flow management function can be performed using the Session Initiation Protocol (SIP) over TCP. An underlying

MRTP/MRTCP		
TCP	SCTP	UDP
IPv4 / IPv6, Multipath Routing		

Fig. 2. Positioning of MRTP/MRTCP in the TCP/IP protocol stack.

multipath ad hoc routing protocol maintains multiple paths from the source to the destination. When SCTP is used in the transport layer, SCTP sockets can be used to set up multiple flows.

The use of multiple flows results in the utilization of path diversity available in ad hoc networks, and provides fault tolerance and load balancing for realtime transmissions. Applications can make the choice of data partitioning method and its parameters based on application-specific requirements.

B. Definitions

The following terms are used in the description of the protocol:

- Flow: similar to an RTP flow [11].
- Session: defines an end-to-end realtime service to an application. It also defines the collection of MRTP flows over which end-to-end service is realized.
- Association: a collection of IP addresses and Port numbers, associated with flows of a MRTP/MRTCP session.

Note that when SCTP or an underlying multiple path routing protocol is not available, there will be only one flow in the MRTP session. In this case, MRTP degenerates to RTP.

The basic components of the proposed protocol are:

1) Traffic Partitioning: Assigns the realtime traffic to multiple paths. A basic traffic partitioning and dispersion scheme is provided, which can be overridden by applications.

2) Session and Flow Management: Unlike RTP, MRTP is a connection-oriented service. A MRTP session should be established first by MRTCP, where two end nodes exchange information on available paths, session/flow IDs, and initial sequence numbers. During the data transmission, a new flow may be added to the session, while stale flows may be removed (based on QoS reports). Each session and each flow in the session has a unique and randomly generated ID to identify them.

3) Timestamping and Sequence Numbering: Similar to RTP [11], but the sequence numbering is done for each flow. Note that the timestamps can be used to synchronize multiple flows at the receiver.

4) QoS Reports: Similar to RTP's Sender Reports (SR) and Receiver Reports (RR), but each has both per-flow statistics and session statistics included. Unlike RTP, the MRTP SR and RR can be sent for each frame. The frequency of MRTP SR and RR is set by the application.

5) Reassembly at the Receiver: The receiver uses a reassembly buffer for each MRTP flow to resequence the received packets, where sequence number of the packets are used. Timestamps in the MRTP data packet header can be used to synchronize the flows.

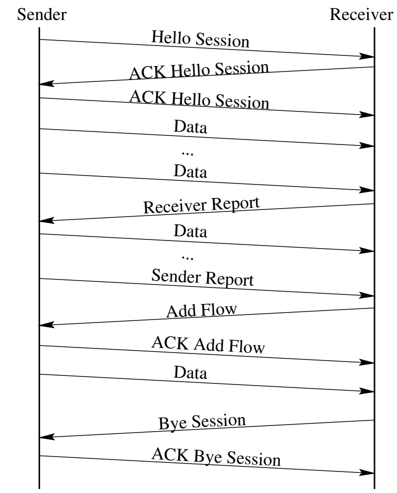

Fig. 3. The operation of MRTP/MRTCP.

The packet formats of MRTP and MRTCP are omitted due to the page limit. The MRTP data packet format is very similar to that of RTP, with additional fields of Flow ID (used to identify which flow it is sent on) and the sequence number in the flow. The MRTCP packets include packets used for session/flow control (Hello Session, ACK Hello Session, Add Flow, ACK Add Flow, Delete Flow, and ACK Delete Flow) and packets used for QoS feedback (RR and SR). The multimedia services supported by MRTP are defined by a set of companion MRTP profiles (RTP compatible). As in RTP, extension headers can be used to support additional functions. For example, authentication can be supported by defining an authentication extension header with encrypted user ID and password fields.

C. The Operation of MRTP/MRTCP

Figure 3 illustrates the typical operation of a MRTP session.

1) Connection Establishment and Termination: MRTP is connection oriented in the sense that a MRTP session needs to be set up before data transfer begins. Either the sender or the receiver can initiate a MRTP using a three-way handshake. Hello Session and ACK Hello Session packets are used for connection establishment. The three-way handshake gives both ends a chance to choose which flows to use for the session and to resolve possible collisions in randomly generated session/flow IDs.

During the data transmission, either end can terminate the MRTP session by sending a Bye Session packet to the other end. Once this Bye Session packet is acknowledged by an ACK Bye Session packet, the MRTP session is terminated. We did not use a four-way handshake because MRTP connections are simplex.

2) Data Transfer: When the MRTP session is established, packets carrying multimedia data is transmitted on the multiple flows associated with the session. Each packet carries a sequence number which is local to its flow and a timestamp that is used by the receiver to reassemble the flows.

3) QoS Reports: During the MRTP session, the receiver keeps monitoring the QoS performance of the flows, such as the accumulative packet loss, the highest sequence number received, and jitter for each flow. These statistics, as well as other information (such as the timestamps used for Round Trip Time (RTT) estimation, etc.) are put in a compound RR packet which is sent to the sender. The RR packets can be sent on a single flow, e.g., the best flow in terms of bandwidth, RTT, or loss probability, or some (or all) of the flows for better reliability. The frequency at which the RR is sent is set by the application.

4) Flow Management: During a MRTP session, some flows may be unavailable (e.g., a node the flow traverses may leave the network). In this case, either the sender or the receiver can delete the flow from the MRTP association by sending a Delete Flow packet carrying the ID of the broken flow to the other end. When a new path is found, a new flow can be added to the association by sending an Add Flow packet. These mechanisms enable MRTP to quickly react to topology changes in the ad hoc network.

D. Usage Scenarios

1) Unicast Video Streaming: This is a *point-to-point* scenario, where a wireless sensor network is deployed to monitor, e.g., wild life, in a remote region. There are two types of sensors in the network: Type One sensors carrying a video camera and with stronger computation capability, and Type Two sensors that are simple relays. Type One sensors capture, encode, and packetize the live video of the region, and Type Two sensors relay the video packets to the base. There could be a few Type One sensors and a large number of Type Two sensors which are relatively cheap. Source routing, or some other simple routing protocols can be used.

A Type One sensor initiates a MRTP session to the base, using multiple flows. The captured video is transmitted to the server in the base via multiple flows going through different Type Two sensors. Some sensors may be damaged or may run out of power. In this case, the underlying multipath routing protocol informs MRTP about the path changes. Either the sender or the receiver can delete a failed flow, or add a new flow to the session.

The server at the base maintains a resequencing buffer for each flow, as well as a deadline for each packet expected to arrive. If a packet arrives later than its deadline, it is regarded as lost.

2) Parallel Video Downloading: This is a *many-to-one* scenario. Consider an ad hoc network, where each node maintains a cache for recently downloaded files. When a node A wants to download a movie, it would be more efficient to search the caches of its neighbors first than going directly to the Internet. If the movie is found in the caches of nodes B, C, and D, A can initiate a MRTP session to these nodes, downloading a piece of the movie simultaneously from each of them. A pair of file pointers is used for each flow indicating the segment of the video assigned to the flow. There will be three flows, each with a unique flow ID. However, the flows have the same session ID since they belong to the same MRTP session. A resequencing buffer is used at A to put the packets into the right order. A similar application of video streaming using multiple servers is presented in [6].

During the transmission, Node D moves out of the network. Node A would delete the flow from D and adjust the file pointers in the other two flows. Now the part of the video initially chosen from D will be downloaded from B and C instead. Node A may broadcast probes periodically to find new neighbors with the video and replace the $stale$ flows in the session. Note MRTP provides the flexibility for applications to implement these schemes.

Combined with multistream video coding schemes, e.g., layered coding with unequal protection of the base layer packets [4] or multiple description coding [15], error resilience can be greatly improved. QoS feedback is used by the video encoder or the traffic allocator to adapt to transmission errors.

III. MRTP Performance Studies

A. The Effect of Data Partitioning in a Bottleneck Node

Consider a mobile node in the ad hoc network. There are N flows, belonging to different MRTP sessions, traversing this node. Also suppose the video is *thinned without time compression* [10]. For example, if a video stream is partitioned into S flows, then flow i consists of the h-th frame of the original video, where $(h \ mod \ S) = i$ and $i \in [0, ..., S-1]$. For an original video stream with a mean rate μ, a substream generated in this manner has a mean rate of μ/S. In the following, we present the impact of thinning on the queueing performance in this bottleneck node.

Previous work on large deviation technique using the Bahadur-Rao asymptotics shows that the buffer overflow probability (BOP) of a queue fed by N homogeneous sources and with total buffer size B and service capacity C is:

$$\Psi(c, b, N) \approx exp[-NI(c, b) + g(c, b, N)], \tag{1}$$

where $I(c, b) = \inf_{m \geq 1}\{[b + m(c - \mu)]^2/[2V(m)]\}$, $g(c, b, N) \approx -\frac{1}{2}\log[4\pi NI(c, b))]$, $c = C/N$, and $b = B/N$. $V(m)$ is the variance of a single source with aggregation level m [16]. It is well-known that video traffic is long range dependent (LRD). Therefore, for a video source with Hurst parameter H, we have $V(m) \approx \sigma^2 m^{2H}$. For the thinned video stream, we have $V(S, m) \approx S^{-2}\sigma^2 m^{2H}$ [10].

Let us also define the measure Γ of improvement in queueing performance of the queueing system fed with thinned video flows as compared to that of the queueing system fed by the original video streams as:

$$\begin{aligned} \Gamma(c, b, N, S) &\stackrel{def}{=} \frac{\Psi(c, b, N)}{\Psi^*(c, b, N, S)} && (2) \\ &= S^{1-H}\exp\{NI(c, b)(S^{2-2H} - 1)\}. && (3) \end{aligned}$$

Note that $\Gamma = 1$ when $S = 1$, and Γ is an increasing function of S.

Figure 4 plots the BOP of a queue fed by 100 thinned video sources with Hurst parameter $H = 0.88$, $\sigma^2 = 1$, $c = 1$,

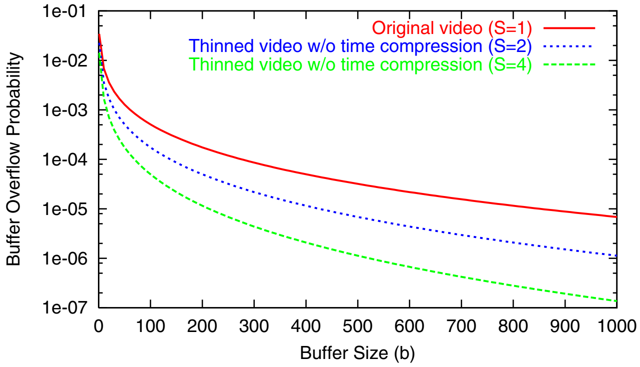

Fig. 4. Buffer overflow probability of a queue fed by 100 video flows.

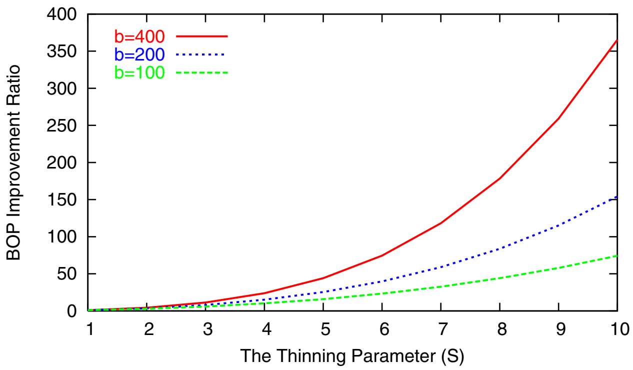

Fig. 5. The BOP improvement ratio, Γ, as a function of the number of flows for different buffer sizes.

$\mu = 0.9$, when 2 and 4 flows are used, respectively. The BOP of the queue fed by 100 original video sources is also plotted for comparison purposes. Note that the system load of the three curves are the same, i.e., the service rate of the thinned system $C^t = C/S$. It can be seen that with thinning, the queueing performance is greatly improved. The BOP decreases when more flows are used. The improved BOP results in smaller delay, lower packet lost rate, and smaller jitter.

To further illustrate the impact of the number of flows used in a MRTP session on the improvement achieved in BOP, we plot Γ (defined in (2)) in Fig. 5 when S increases from 1 to 10. It can be seen that Γ is 1 when a single flow is used and increases with S. Figure 5 also shows that larger improvement can be achieved for larger per-flow buffer assignments.

B. Experiments with Video Transport over Ad Hoc Networks

In this section, we present the performance study of MRTP using OPNET models [17]. We simulated an ad hoc network consists of 16 nodes in a 600m by 600m region. Each node is randomly placed in the region initially. The popular *Random Waypoint* mobility model is used [20], with a *constant* nodal speed of 10m/s and a constant pause time of 1 second. We used the IEEE 802.11 protocol in the MAC layer working in the DCF mode. The channel bandwidth is 1Mbps and the transmission range is 250 meters. MRTP is implemented in the application layer. UDP is used in the transport layer and

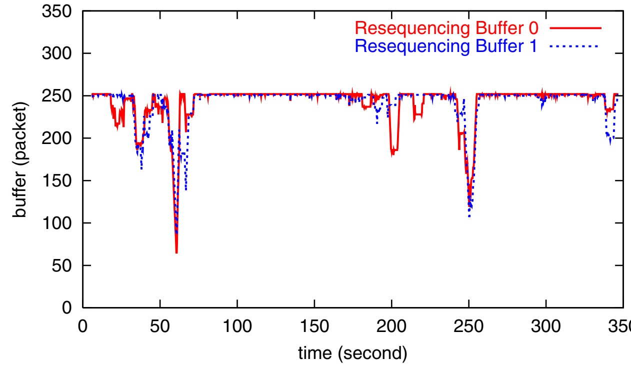

Fig. 6. The occupancies of the resequencing buffers of two flows at the receiver.

source routing [18] is used.

We used the Quarter Common Intermediate Format (QCIF, 176×144 Y pixels/frame, 88×72 Cb/Cr pixels/frame) sequence "Foreman" (the first 200 frames from the original 30 fps sequence) encoded at 10 fps. The MDMC video codec was used [15], generating two video flows with a bit rate of 59Kbps each. In MDMC, 5% macroblock level intra-refreshments are used, which has been found to be effective in suppressing error propagation for the range of the packet loss rates considered. Each group of blocks (GOB) is packetized into a single packet, to make each packet independently decodable.

Among the 16 nodes, one is randomly chosen as the video source and another as the video receiver, where a 5 second playout buffer is used to absorb the jitter in received packets. The MRTP session uses two routes. All other nodes generate background traffic to send to a randomly chosen destination. The inter-arrival time of the background packets is exponentially distributed with a mean of 0.2 second. The background packets have a constant length of 512 bits.

Figure 6 plots the resequencing buffer occupancies at the receiver. It can be seen that the variations in the buffer occupancies are relatively independent to each other. This demonstrates the benefit of path diversity. Note that at the 30th, the 60th, and the 250th seconds, both buffers drops simultaneously, which implies that both paths are down at these time instances. When more than two paths are used, it is expected that this will occur less often.

Next we compare the performance of MRTP with RTP in Fig.7, where the same MDMC codec was used. For RTP, we used the NIST DSR model [19] which maintains a single path to a destination. For MRTP, we used the MDSR model [4], which is a multipath routing extension of [19]. We transmit both the substreams on a single path in the RTP simulations, while for MRTP, each substream is assigned to a path found by MDSR. The PSNR traces (using the left y axis) and loss traces (using the right y axis) using MRTP and RTP are plotted in the upper and lower plot of Fig.7, respectively. It can be seen that PSNR drops when there is loss in either flow. Also the deepest drop occurs when large bursts on the two flows overlap. RTP has higher loss rates than MRTP, and therefore

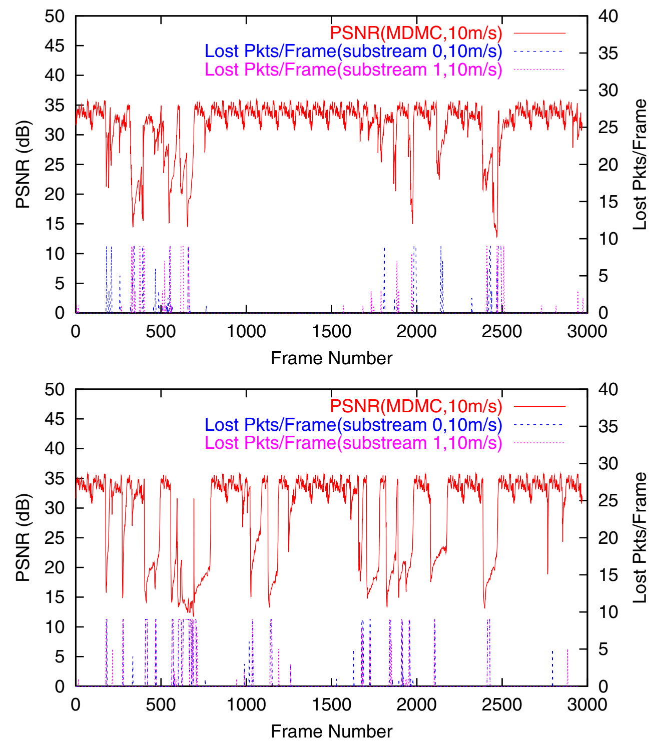

Fig. 7. The lost packets per frame plot and the PSNRs of the received video frames. The upper plot: two flows were used. The lower plot: a single flow was used.

its PSNR curve has more frequent and severe drops than that of MRTP. It is obvious that RTP has poorer performance than MRTP.

IV. Conclusions and Future Work

In this paper, we propose a new protocol, MRTP/MRTCP, for realtime transport over ad hoc networks using multiple flows. Our proposal is motivated by the observations that (1) path diversity is effective in combating transmission errors in ad hoc networks, and (2) data partitioning techniques are effective in improving the queueing performance of realtime traffic. The new protocol is an extension of RTP/RTCP, exploiting multiple paths existing in mesh networks. The new protocol works closely and effectively with SCTP for realtime transport.

For the future work, a working implementation of the proposed protocol, e.g., a MRTP/MRTCP testbed, would be useful in validating its pros and cons. Furthermore, although multipath transport has an inherent security strength (since it would be difficult for an attacker to track all the paths in use and to guess how the traffic is partitioned), security considerations are yet not the focus of our design. Besides the randomly generated session/flow IDs and initial sequence numbers, security can be strengthened by introducing some randomness in data partitioning. These are interesting research problems worth investigating. In addition, we are working on an Inernet Draft on MRTP/MRTCP for the IETF. An extended version of this paper can be found at [21].

Acknowledgment

The authors would like to thank Professor Yao Wang and Mr Shunan Lin of Polytechnic University for the MDMC codec, and the anonymous reviewers for their helpful comments.

This work is supported by the National Science Foundation under Grant ANI 0081375 and the New York State Center for Advanced Technology in Telecommunications (CATT) at Polytechnic University, Brooklyn, NY, USA.

References

[1] Y. Wang and Q.-F. Zhu, “Error control and concealment for video communication: a review,” in *Proc. IEEE*, vol.86, issue 5, pp.974-997, May 1998.

[2] N. Gogate, D. Chung, S. S. Panwar, Y. Wang, “Supporting image/video applications in a multihop radio environment using route diversity and multiple description coding,” *IEEE Trans. Circuit Syst. Video Technol.*, vol.12, no.9, pp.777-792, Sept. 2002.

[3] S. Mao, S. Lin, D. Bushmitch, S. Narayanan, S. S. Panwar, Y. Wang, and R. Izmailov, “Real time transport with path diversity,” the 2^{nd} *NY Metro Area Networking Workshop*, New York, September 2002.

[4] S. Mao, S. Lin, S. S. Panwar, and Y. Wang, “Video transport over ad hoc networks: Multistream coding with multipath transport,” to appear, *IEEE JSAC Special Issue on Recent Advances in Wireless Multimedia*.

[5] Y. J. Liang, E. G. Steinbach, and B. Girod, “Multi-stream voice over IP using packet path diversity,” in *Proc. IEEE Multimedia Siganl Processing Workshop*, pp.555-560, Sept. 2001.

[6] J. G. Apostolopoulos, T. Wong, W. Tan, and S. Wee, “On Multiple Description Streaming in Content Delivery Networks,” in *Proc. IEEE INFOCOM*, pp.1736-1745, June 2002.

[7] E. M. Royer and C.-K. Toh, “A review of current routing protocols for ad hoc mobile wireless networks,” *IEEE Personal Communications*, vol.6 issue.2, pp.46-55, April 1999.

[8] P. J. Shenoy and H. M. Vin, “Efficient striping techniques for multimedia file servers,” *Performance Evaluation*, vol.38, pp.175-199, 1999.

[9] D. Bushmitch, R. Izmailov, S. Panwar, A. Pal, “Thinning, Striping and Shuffling: Traffic Shaping and Transport Techniques for Variable Bit Rate Video,” in *Proceedings of IEEE GLOBECOM’02*, Taipei, 2002.

[10] D. Bushmitch, “Thinning, striping and shuffling: Traffic shaping and transport techniques for VBR video,” *PhD Dissertation*, Electrical and Computer Engineering Department, Polytechnic University, 2003.

[11] H. Schulzrinne, S. Casner, R. Frederick, and V. Jacobson, “RTP: A transport protocol for realtime applications,” IETF Request For Comments 1889. [Online]. Available at: `http://www.ietf.org`.

[12] R. R. Stewart and Q. Xie, *Stream Control Transmission Protocol: A Reference Guide*. Reading, MA: Addison-Wesley, 2001.

[13] D. S. Phatak and T. Goff, “A novel mechanism for data streaming across multiple IP links for improving throughput and reliabilit in mobile environments,” in *Proc. IEEE INFOCOM*, pp.773-782, June 2002.

[14] H.-Y. Hsieh and R. Sivakumar, “A transport layer approach for achieving aggregate bandwidths on multi-homed mobile hosts,” in *Proc. ACM Inter. Conf. Mob. Comp. Networking*, pp.83-95, September 2002.

[15] Y. Wang and S. Lin, “Error resilient video coding using multiple description motion compensation,” *IEEE Transaction on Circuits and Systems for Video Technology*, vol.12, no.6, pp.438-452,September 2002.

[16] M. Montgomery and G. De Veciana, “On the relevance of time scales in performance oriented traffic characterization,” in *Proceedings of the IEEE INFOCOM*, 1996.

[17] OPNET Tech., Inc. *OPNET Modeler*. [Online]. Available: `http://www.mil3.com`.

[18] D. B. Johnson, D. A. Maltz, Y.-C. Hu, and J. G. Jetcheva, *The Dynamic Source Routing Protocol for Mobile Ad Hoc Networks* , IETF Internet Draft (draft-ietf-manet-dsr-03.txt), Oct., 1999 (work in progress).

[19] The National Institute of Standards and Technology, *OPNET DSR Model*. [Online]. Available: `http://w3.antd.nist.gov/wctg/prd\_dsrfiles.html`.

[20] 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 *Proc. ACM/IEEE Inter. Conf. Mobile Comp. and Networking*, pp.85-97, 1998.

[21] The MRTP homepage. [Online]. Available: http://theater.poly.edu/mrtp.