Multipath Video Transport over Ad Hoc Networks

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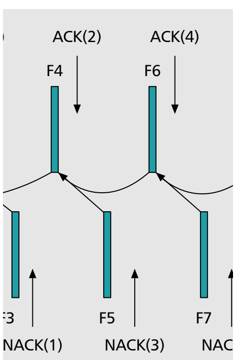

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Abstract

Real-time multimedia transport has stringent bandwidth, delay, and loss requirements. It is a great challenge to support such applications in wireless ad hoc networks, which are characterized by frequent link failures and congestion. Using multiple paths in parallel for a real-time multimedia session (called *multipath transport*) provides a new degree of freedom in designing robust multimedia transport systems. In this article, we describe a framework for multipath video transport over wireless ad hoc networks, and examine its essential components, including multistream video coding, multipath routing, and transport mechanisms. We illustrate by three representative examples how to extend existing video coding schemes in order to fully explore the potential of multipath transport. We also examine important mechanisms in different layers for supporting multipath video transport over ad hoc networks. Our experiments show that multipath transport is a promising technique for efficient video communications over ad hoc networks.

Introduction

With the recent advances in wireless technologies, wireless networks are becoming a significant part of today's access networks. Ad hoc networks are wireless mobile networks without an infrastructure, within which mobile nodes cooperate with each other to find routes and relay packets. Such networks can be deployed instantly in situations where infrastructure is unavailable or difficult to install, and are maturing as a means to provide ubiquitous untethered communication. There is a demonstrable need for providing video service for users of ad hoc networks, such as first responders, search and rescue teams, and military units. Such content-rich service is more substantial than simple data communications: it will add value to and catalyze the widespread deployment of ad hoc networks.

In video communications, a receiver usually displays the received video continuously. Such continuous display requires timely delivery of video data, which further translates to stringent quality of service (QoS) requirements (e.g., delay, bandwidth, and loss) on the underlying network. For the successful reconstruction of received video, the path used for the video session should be stable for most of the video session period. Furthermore, packet losses due to transmission errors and overdue delivery caused by congestion should be kept low, such that they can be handled by error control and error concealment techniques. However, this situation does not hold true in ad hoc networks, where wireless links are frequently broken and new ones reestablished due to mobility. Furthermore, a wireless link has a high transmission error rate because of shadowing, fading, path loss, and interference from other transmitting users. Consequently, for efficient video transport, traditional error control techniques, including forward error correction (FEC) and automatic repeat request (ARQ), should be adapted to take into consideration frequent link failures and high transmission errors. In addition, one should take a holistic approach in video transport system design, by jointly considering and optimizing mechanisms in various layers, including video coding, error control, transport mechanisms, and routing. This approach is often referred to as cross-layer optimization.

Among various mechanisms, *multipath transport*, by which multiple paths are used to transfer data for an end-to-end session, is highly suitable for ad hoc networks, where a mesh topology implies the existence of multiple paths for any pair of source and destination nodes. Multipath transport has been applied in various settings for data [1]. Recently, there has been

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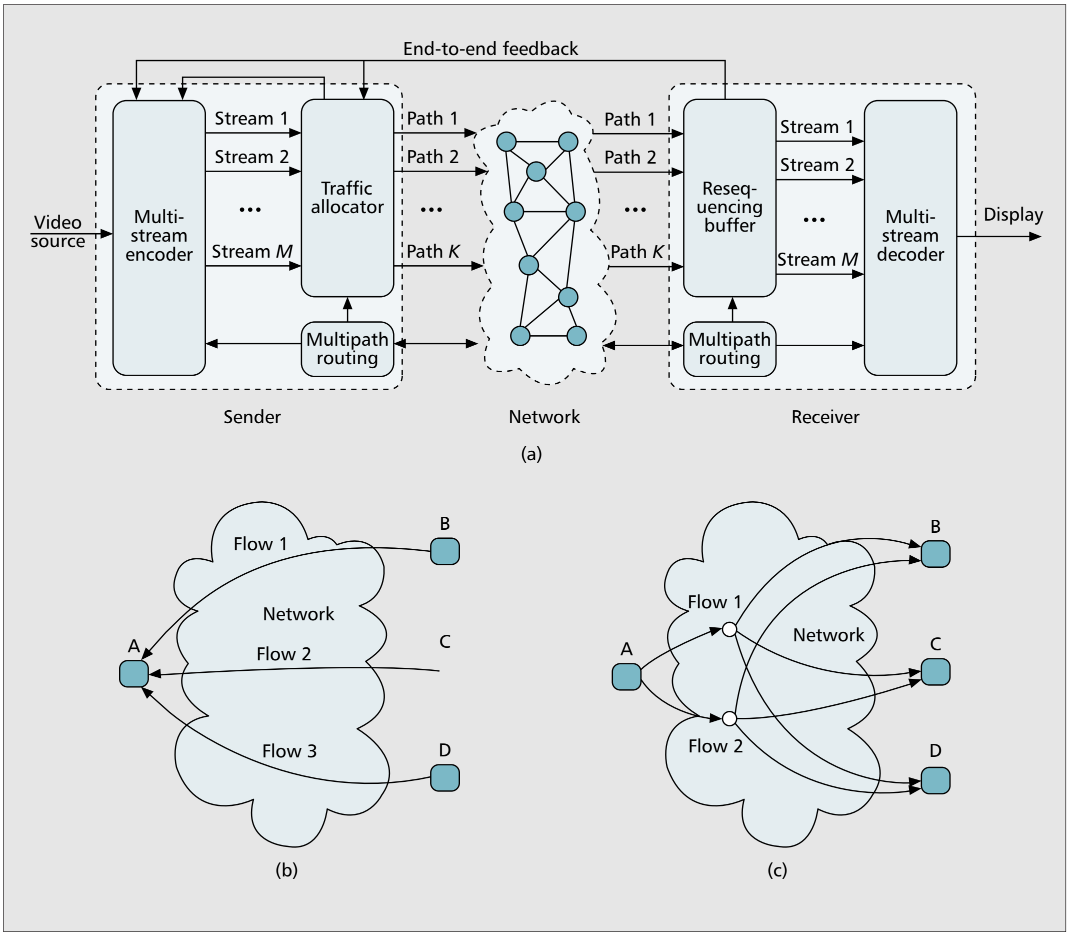

Figure 1. *The general architecture for the multipath transport of realtime multimedia applications: a) point-to-point video communications; b) concurrent streaming; c) multicast using two trees.*

In general, the quality of the paths may change over time. We assume that the system receives feedback about network QoS parameters. Although not necessary, such feedback can be used to adapt the coder and transport mechanisms to network conditions

considerable research on using multipath transport for real-time multimedia applications [2–13]. For example, multipath transport has been combined with multiple description coding (MDC) [2–6, 9–11], ARQ [6], and FEC [12] for video transport. It has been shown that, when combined with appropriate source and/or channel coding and error control schemes, multipath transport can significantly improve the media quality over traditional shortest-path-routing-based schemes. This also inspired previous and ongoing standardization efforts for multipath transport protocols in the Internet Engineering Task Force (IETF) [13, 14].

In this article we examine the problem of using multipath transport for video applications in ad hoc networks, and discuss related issues and techniques. We show that multipath transport provides a new dimension of freedom in designing a robust video transport system for ad hoc networks. We present the general application scenarios, as well as the benefits and design trade-offs of using multipath transport for video communications. We then discuss related issues in the following sections, including multistream video coding, and network and transport considerations. Performance studies are presented to demonstrate the efficacy of multipath transport techniques for video transport over ad hoc networks. We then conclude this article.

Multipath Multimedia Transport

Architecture Overview

The general architecture for multipath transport of video streams is depicted in Fig. 1a. At the sender, raw video is first compressed by a video encoder into M streams. When $M > 1$, we call the coder a *multistream coder*. Then the streams are partitioned and assigned to K paths by a *traffic allocator*. These paths are maintained by a *multipath routing protocol*. When the flows arrive at the receiver, they are first put into a *resequencing buffer* to restore the original order. Finally, the video data is extracted from the resequencing buffer to be decoded and displayed. The video decoder is expected to perform appropriate error concealment if any part of a substream is lost.

In general, the quality of the paths may change over time. We assume that the system receives feedback about network QoS parameters. Although not necessary, such feedback can be used to adapt the coder and transport mechanisms to network conditions (e.g., the encoder could perform rate control based on feedback information, in order to avoid congestion in the network). In practice, it is desirable for the sender to use a predesigned multistream coder that always produces a fixed number of streams (say, two to four). On the other hand, the num-

> An important advantage of using multipath transport is the inherent path diversity, i.e., the independence of loss processes of the paths. As a result, the receiver can always receive some data during any period, except when all the paths are down simultaneously.

ber of available paths, as well as their bandwidths, may vary over time due to network topology changes and congestion. Therefore, it is likely that $M \neq K$ in Fig. 1a, and the traffic allocator is responsible for distributing the video packets from the M streams to the K available paths [8].

The point-to-point architecture in Fig. 1a can be used for two-way conversational services as well as one-way streaming services. For the latter case, it can be extended to more general cases. For example, an architecture for the many-to-one type of application is shown in Fig. 1b, where a node streams a video clip from multiple servers concurrently. Every video server (e.g., node B, C, or D) is a mobile node in an ad hoc network that has the target video (or some portion of the target video) in its cache or public directory [4]. A multicast-based architecture is shown in Fig. 1c, where a source multicasts a video to a group of nodes using two multicast trees in parallel [15].

ADVANTAGES AND DESIGN TRADE-OFFS

The advantages of using multipath transport have been reported in many previous papers, for example, see [1–15] and the references therein. An important advantage of using multipath transport is the inherent *path diversity* (i.e., the independence of loss processes of the paths). As a result, the receiver can always receive some data during any period, except when all the paths are down simultaneously, which occurs much more rarely than single path failures. One may jointly design the source encoder, multipath routing algorithm, and traffic allocator to explore path diversity in order to optimize overall system performance. In addition, multipath transport provides a larger aggregate rate for a video session, thus reducing the distortion caused by lossy video coders. Finally, multipath transport distributes traffic load in the network more evenly, resulting in low congestion and delay in the network.

These advantages come at the cost of higher coding redundancy, higher computation complexity, and higher control traffic overhead in the network. In general, using more streams and paths will increase the robustness to packet losses and path failures, and reduce network congestion due to better load balancing. However, more streams may increase the video bit rate for the same video quality, as well as incur higher computation overhead and delay during traffic partitioning and resequencing. Maintaining multiple paths in a dynamic ad hoc network environment involves higher control traffic overhead and more complex routing/path selection algorithms. The study in [10] demonstrates that the most significant performance gain is achieved when K increases from 1 to 2, with lesser improvements achieved for further increases in K. As a result, a baseline system having $M = 2$ and $K = 2$ will provide significant performance gains at a moderate cost.

MULTISTREAM VIDEO CODING

For multipath transport to be helpful for sending compressed video, one must carefully design the video coder to generate streams so that the loss in one stream does not adversely affect the decoding of other streams. However, this relative independence between the streams should not be obtained at great expense in coding efficiency. Therefore, a multistream encoder should strive to achieve a good trade-off between coding efficiency and error resilience. In addition, one must consider what is feasible in terms of transport layer error control when designing the source coder. In this section we illustrate how to adapt a video coder to multipath transport for better performance. We use three representative coding schemes as examples, which differ in terms of their operations and network requirements.

FEEDBACK-BASED REFERENCE PICTURE SELECTION

One simple way to generate multiple video streams is to code a video into one stream in a standard way and then disperse that stream onto multiple paths (e.g., sending bits corresponding to the even frames on one path and those for the odd frames on the other). This simple method, however, has poor performance since the streams on the two paths are dependent on each other. That is, the even frames are predicted from the previous (odd) frame, and vice versa.

We improved this method by exploring the reference picture selection (RPS) technique [6]. Specifically, we still use the same time domain partitioning method (i.e., sending coded even and odd frames separately). However, a more network-aware coding method is used, which selects the reference picture based on feedback and estimated path status. Assume that the decoder sends a negative acknowledgment (NACK) for a frame if it is damaged or lost, and a positive one (ACK) otherwise. The encoder can then estimate the status of the paths and infer which of the previous frames are damaged. Based on the estimation, for a picture to be coded, the closest picture for which itself as well as its reference pictures have been transmitted on the better path is selected as the reference picture.

Figure 2a is an example of the proposed RPS scheme. When NACK(1) is received at the time frame 4 is being encoded, the encoder deduces that frames 2 and 3 cannot be decoded correctly due to error propagation. Therefore, frame 0 is chosen as the reference for frame 4, and path 2 is set to the “bad” state. When frame 6 is coded, the encoder uses frame 4, instead of frame 5 as the reference picture, because path 2 is still in the “bad” state. When ACK(7) is received, path 2 switches back to the “good” state. Frame 9 is then chosen as the reference picture for frame 10.

The RPS scheme offers a good trade-off between coding efficiency and error resilience. That is, when both paths are good, RPS uses the neighboring frame as reference, thereby achieving the highest possible prediction gain and coding efficiency; when one path is bad, the encoder avoids using any frames that are transmitted by that path, thereby minimizing the error propagation period. Note that the RPS scheme is only applicable for online coding, because it adapts the encoding operation based on channel feedback.

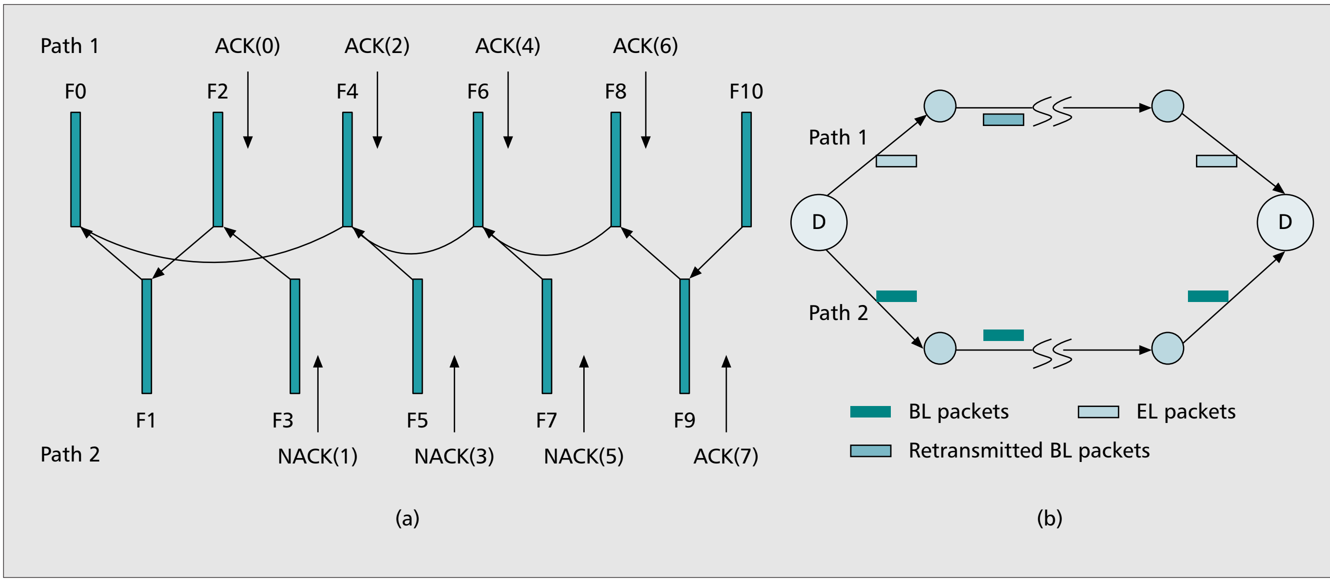

Figure 2. *a) The RPS scheme: the arrow from a frame indicates the reference picture used in coding that frame; b) the two-path layered video transmission model with selective ARQ for base layer packets.*

LAYERED CODING WITH SELECTIVE ARQ

A second option is based on the popular *layered coding* technique, where a video frame is coded into a base layer and one or more enhancement layers. Reception of the base layer can provide low but acceptable quality, while reception of the enhancement layer(s) can further improve the quality over the base layer alone, but the enhancement layers cannot be decoded without the base layer.

When the layered video is transmitted over multiple paths (e.g., two paths), the traffic allocator sends the base layer packets on one path and the enhancement layer packets on the other one. The path with a lower packet loss rate is used for the base layer if the two paths have different qualities. The receiver returns selective ARQ requests to the sender to report base layer packet losses. When the sender receives such a request, it retransmits the requested base layer packet on the enhancement layer path, as illustrated in Fig. 2b. The transmission bit rate for the enhancement layer will be reduced correspondingly according to the bandwidth reallocated for base layer retransmissions. We denote this scheme LC with ARQ [6].

Generally, a multihop wireless path is up or down for random periods of time, leading to bursty packet losses. If there is a base layer packet loss, the base layer path is likely to be experiencing a packet loss burst. Therefore, base layer retransmission using the same path is likely to be unsuccessful. Moreover, if the loss was caused by congestion at an intermediate node, using the base layer path for retransmission may intensify the congestion condition. When disjoint paths are used, the loss processes of the paths may not be totally correlated. Therefore, base layer packet retransmission using the enhancement layer path could have higher success probability and lower delay.

MULTIPLE DESCRIPTION CODING

The third option is to use multiple description coding. MDC is a technique that generates multiple *equally important* descriptions. The decoder reconstructs the video from *any* subset of received descriptions, yielding a quality commensurate with the number of received descriptions.

In [6] a multiple description (MD) coder known as multiple description motion compensation (MDMC) is employed. With this coder, two descriptions are generated by sending even pictures as one description and odd pictures as the other. When coding a picture, say picture n, the encoder uses two kinds of predictions:

- A prediction from a linear superposition of two previously coded frames, pictures $n - 1$ and $n - 2$, called the *central prediction*
- A prediction from the previously coded picture in the same description, picture $n - 2$, called the *side prediction.*

Then the encoder codes two signals for picture n; that is, the central prediction error (the difference between picture n and the central prediction) and the reference mismatch signal (essentially the difference between the central and side predictions). Description one includes central prediction errors and the reference mismatch signals for even pictures, and description two includes those for odd pictures. When both descriptions are received, the decoder can reproduce the central prediction and will reconstruct a picture by adding the central prediction error to the central prediction. When only one description is received, the decoder can only generate the side prediction, and a picture is decoded by using both the central prediction error and the mismatch signal. The redundancy of this coder can be adjusted by the predictor coefficient used for central prediction and the quantizer used for the reference mismatch.

Compared to layered coding, MDMC does not require the network or channel coder to pro-

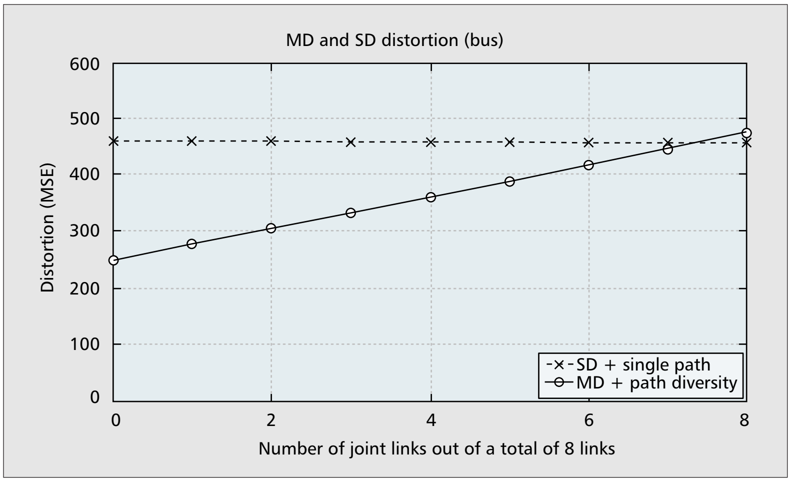

Figure 3 *MD vs. SD distortion as the number of joint links between two paths varies. Simulation parameters are given in [4].*

	RPS	LC with ARQ	MDMC
Feedback needed	Yes	Yes	No
Decoding delay	No	Yes	No
Offline encoding	No	Yes	Yes

Table 1. *Comparison of the three schemes.*

vide different levels of protection. Nor does it require any receiver feedback. Acceptable quality can be achieved even when both descriptions are subject to relatively frequent packet losses, as long as the losses on the two paths do not occur simultaneously and sufficient amount of redundancy is added by appropriately choosing the predictor coefficient and mismatch signal quantizer.

Comparison and Discussion

The three schemes have their respective advantages and disadvantages. A qualitative comparison of these three schemes is given in Table 1. In summary, if feedback is not available, only MDMC is applicable. If the decoding delay requirement is very stringent, RPS and MDMC are the possible choices. If the delay requirements are not stringent, LC with ARQ is an additional choice. In terms of the received quality for the same total bandwidth usage (in an application where all the three schemes are candidates), LC with ARQ has the best performance for medium and high loss rates. MDMC is well suited for low loss rates, and RPS outperforms the other two when the loss rate is very low [6]. However, because RPS adapts its encoder based on receiver feedback, it is not applicable for streaming of pre-encoded video.

Network and Transport Considerations

Path Correlation and Video-centric Multipath Routing

In multipath transport, correlation of the paths is an important factor that affects the performance achieved, and thus should be considered in path selection for the video session. In [4] Apostolopoulos *et al.* investigated the impact of shared links on the end-to-end video quality. Specifically, they examined the case when a user receives two descriptions from two servers through two symmetric paths, each consisting of eight links. In addition, these two paths have a different number of shared links at the receiver end for each experiment. Figure 3 shows that the improvement achieved by an MD video (using two paths) over single description (SD) video (using a single path) decreases as the number of shared links increases. The crossing point of the two systems depends on the video sequence content and link model parameters.

Generally, each of the paths consists of a number of links, and the shared links between the paths may be scattered over different portions of the two paths. Apostolopoulos *et al.* [4] show that the impact of packet loss on video quality can be captured by a simple model with only three subpaths: two disjoint subpaths, each consisting of disjoint links of a path, and one common subpath that consists of all shared links. The impact of the paths on video distortion can then be modeled by a four-state Markov chain, whose parameters can be computed from link statistics. A similar path model is derived in [5] using the technique of path segregation and link aggregation. In [5] the common subpath only includes the shared links before the two paths first split. This is motivated by the observation that once two paths split, packets that are initially sent back to back on the two paths will experience very different delays and rarely fall into the same burst interval again, even if they are carried on a shared link at a later time.

With such path models, the multipath routing for video traffic can be formulated as a crosslayer combinatorial optimization problem, where the objective is minimizing video distortion, and the constraints include connectivity, loop-free paths, and stable links. The solution space, which is exponential, consists of combinations of all feasible paths that provide a connection from the source to the destination. Since the formulated problem is NP-hard, efficient heuristic algorithms are highly desirable.

The recent work [7] presents an efficient genetic algorithm (GA)-based multipath routing approach. GA is a population-based metaheuristic that is inspired by the *survival of the fittest* principle. Starting with a set of solutions (i.e., a *population*), in each iteration a number of genetic operators are applied to the individuals of the current population in order to generate offspring. Individuals with a higher degree of fitness (in the form of an objective function value) are more likely to be chosen for the next generation. The survival of the fittest principle ensures that the overall quality of the population increases as the algorithm progresses from one generation to the next. It has been shown in [7] that significant gain in video quality can be achieved by this approach over traditional network-centric multipath routing schemes, mainly due to the fact that the application layer performance metric, video distortion, is directly optimized.

Establishing Multiple Routes

Once a set of paths are chosen, they may be established in several ways. If *source routing* is supported by the underlying network, the sender can store the entire route in the headers of video data packets. Each intermediate node simply examines the header of a received packet, and forwards it to the next node as indicated in the source route. The very popular ad hoc network routing protocol, Dynamic Source Routing (DSR), and its many extensions are based on source routing. Another way of using multiple paths is by using Stream Control Transmission Protocol (SCTP) [14], which has the built-in features of multistreaming, where a flow is partitioned and transmitted as multiple streams, and multihoming, where an SCTP endpoint can use multiple network interfaces. SCTP sockets can be used to set up multiple connections, after which a traffic allocator can assign real-time multimedia data to those connections.

If the underlying network does not support source routing and SCTP, multipath routing can be performed via an overlay approach, called *application-level multipath routing*. That is, all participating nodes will form an overlay network, within which each logical link may consist of one or more physical wireless links. Multipath routing and packet forwarding can be easily implemented in the application layer without changing the underlying network architecture and operation [5].

Multiflow Real-Time Transport Protocol

Multiflow Real-Time Transport Protocol (MRTP) was designed to support the general multipath transport architectures shown in Fig. 1 for realtime multimedia applications [13]. The protocol stack with MRTP is shown in Fig. 4. MRTP is a transport protocol implemented in the application layer. Given multiple paths maintained by an underlying multipath routing protocol, MRTP and its companion control protocol, the Multiflow Real-Time Transport Control Protocol (MRTCP), provide essential support for multiple path real-time transport, including session and flow management, data partitioning, traffic dispersion, timestamping, sequence numbering, resequencing, and QoS feedback.

MRTP/MRTCP is a natural extension of the Real-Time Transport Protocol/Real-Time Transport Control Protocol (RTP/RTCP). They are used to support real-time traffic, and facilitates multipath transport to combat frequent link failures and congestion in ad hoc networks. Although the functionality provided by MRTP can be implemented in the application layer by each application independently, it would be valuable to abstract and package the real-time multipath transport-related functions as a single generic protocol that can be used by many different applications, and thus relieve multimedia applications of such burdens.

Unlike RTP, MRTP is a session-oriented protocol. An MRTP session should be established first by MRTCP, where two end nodes exchange information such as available paths, session/flow identifiers, and initial sequence numbers. During data transmission, a new path may be added to the session when a better path is found, and a degraded path may be removed from the session based on QoS reports. The flows could then be switched from degraded paths to new paths.

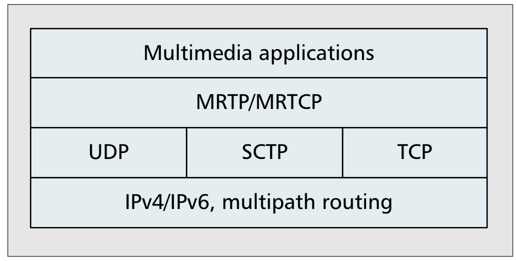

Figure 4. *The protocol stack using MRTP.*

With MRTP, a traffic allocator partitions and disperses the real-time multimedia traffic to multiple flows. As in RTP, MRTP generates QoS reports periodically. An MRTP sender report (SR) or receiver report (RR) carries both the per-flow statistics and session statistics. Unlike RTP, the MRTP SR and RR can be sent at an interval set by the application. Timely QoS reports enable the sender to quickly adapt to path characteristics. For example, the encoder can change the coding parameters or encoding mode for the next frame, introducing more (or less) redundancy for error resilience, or the traffic allocator can avoid the use of inferior paths and disperse packets to better paths.

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Performance Evaluation

The three alternatives discussed earlier were studied and compared using simulations based on theoretical channel models. The results are presented in [6]. We found that each of these three techniques is best suited for a particular environment, depending on the availability of feedback channels, the end-to-end delay constraint, and the error characteristics of the paths.

In the following we present a comparison study of a multipath transport (MPT) system (using two paths) with a single path transport (SPT) system for video streaming in an ad hoc network, using OPNET simulations and the MDMC codec. For MPT, a multipath routing extension of the DSR algorithm or MDSR [6], and MRTP/MRTCP were used. SPT consists of the DSR routing protocol and RTP/RTCP. For both systems, the MDMC codec was used to generate two descriptions, each with a rate of 59 kb/s. With the MPT system, the two descriptions are sent over two maximally disjoint paths maintained by MDSR. For the SPT system, the two descriptions are multiplexed onto a single path established by DSR.[1] Packets from the two descriptions are interleaved with a period of two frames before transmission in order to reduce the impact of bursty losses. A playout buffer is used in both simulations. The ad hoc network consists of 16 nodes moving in a 600 m × 600 m region at 5 m/s according to the random way-

[1] *Note that even with SPT, MDC can still be useful when the losses of two descriptions are not totally correlated. This is because the region lost in one description could be estimated based on the received bits for the corresponding region in the other description.*

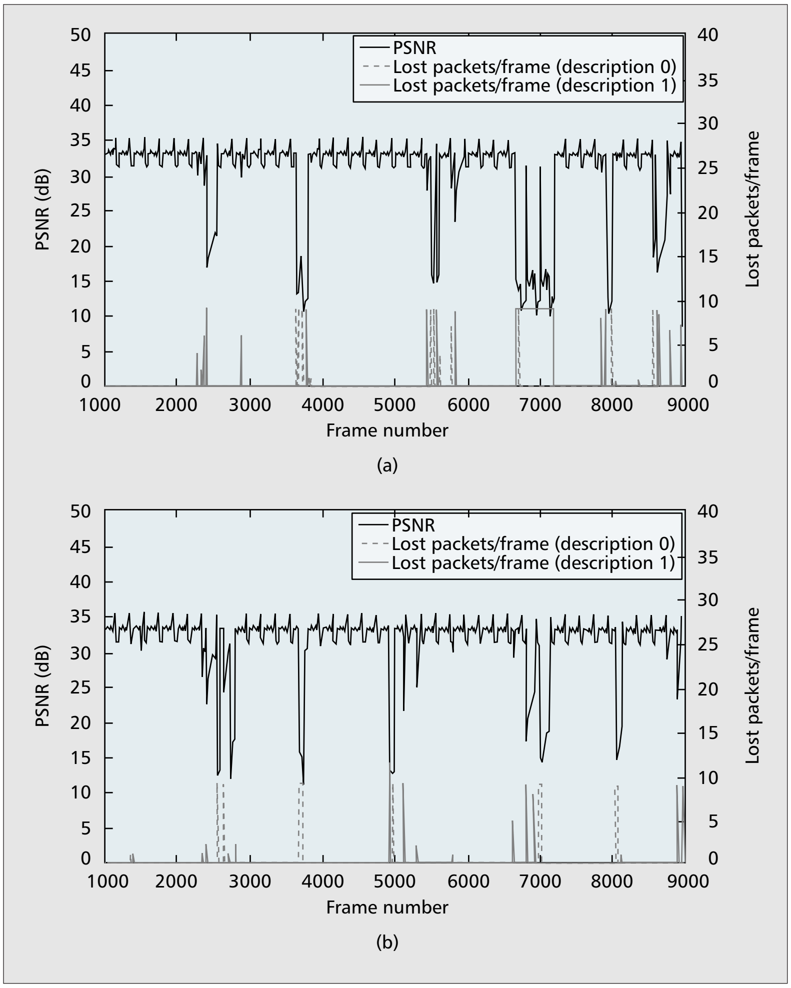

Figure 5. *Comparison of a) single path transport; b) multipath transport (two paths used). The top curve in each figure shows the PSNRs of received frames; the lower curves show the packet losses per frame experienced by the two streams. The average PSNR is a) 30.19 dB; b) 31.45 dB.*

point mobility model (2 s pause time).

We observe that the peak signal-to-noise ratio (PSNR) drops when there is loss in either description. Also, the deepest drop occurs when a large burst of losses in one description overlaps with a loss burst of the other flow. It can be seen that SPT has higher loss rates than MPT. Furthermore, a careful examination shows that the two loss traces of SPT are highly correlated. Therefore, the PSNR curve in Fig. 5a has more frequent and severe degradations than that in Fig. 5b. MPT achieves a significant 1.26 dB gain in average PSNR over SPT in this experiment.

In order to validate the feasibility of video transport over ad hoc networks and evaluate the achievable video quality with off-the-shelf technology, we implemented an ad hoc multipath video streaming testbed. The testbed consisted of four IBM Thinkpad laptops with 802.11b cards. One laptop was chosen as the video source, one as the video receiver, and the remaining two served as relay nodes. For this four-node ad hoc network, two static routes from the source node to the destination node were used, each going through a relay node. We implemented the key functions of MRTP, including traffic partitioning and reassembly, timestamping, sequence numbering, and QoS feedback functions. The LC with ARQ and MDMC codecs were implemented and used in this testbed.

Details on the testbed settings and experimental results can be found in [6]. Our experiments show that video streaming with an acceptable quality is achievable with both LC with ARQ and MDMC, for the range of video bit rates, background traffic, and motion speed examined. Our experiments with the testbed demonstrate the viability of video transport over ad hoc networks using multipath transport and multistream coding.

Conclusions

In this article we describe a framework for multipath video transport over wireless ad hoc networks, and examine its essential components, including multistream video coding, multipath routing, and transport mechanisms. We present example solutions for each component and the performance achievable with a system integrating these components. We demonstrate that multipath transport combined with appropriate video coding techniques can lead to substantial gain over a system using a single path.

It is worth noting that these schemes also apply to wired mesh networks (e.g., the Internet) where path diversity could be exploited by using multiple access routers. Furthermore, when retransmission is not available, one could generally consider applying FEC over the same stream to reduce the packet loss rate within the stream, as well as across packets delivered over separate paths [12]. However, these approaches cause additional delay and channel coding redundancy, and may not be efficient in combating bursty losses. Our studies show that MDMC and RPS without FEC are sufficient for the loss environment considered.

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Biographies

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Our experiments show that video streaming with an acceptable quality is achievable with both LC with ARQ and MDMC, for the range of video bit rates, background traffic, and motion speed examined.