附件1（外文文献原文，范例）：

PS：建议直接打印外文文献，并装订与相应位置，论文对外文文献原文格式不做要求，坚持直接用外文文献原格式直接打印

MULTIPATH VIDEO TRANSPORT OVER AD HOC NETWORKS

**SHIWEN MAO, VIRGINIA POLYTECHNIC INSTITUTE AND STATE UNIVERSITY**

**SHUNAN LIN, HARMONIC INC.**

**YAO WANG, SHIVENDRA S. PANWAR, AND YIHAN LI, POLYTECHNIC UNIVERSITY**

**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**

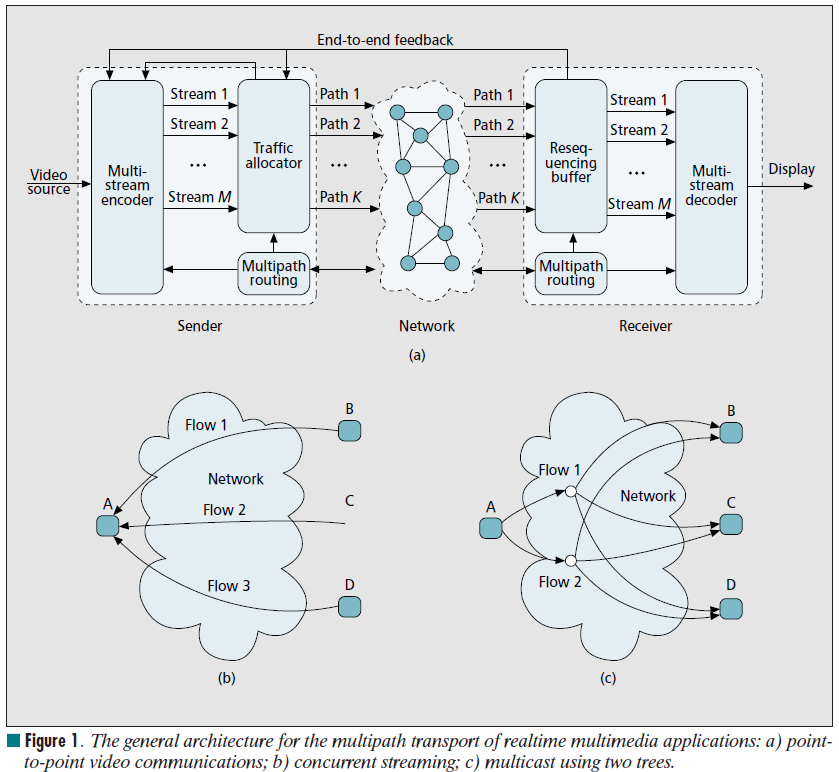
With the recent advances in wireless technologies, wireless networks are becoming a significant part of today’s access networks. Ad hoc networks are wireless mobile networks without an infrastructure, within which mobile nodes cooperate with each other to find routes and relay packets. Such networks 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 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-pathrouting-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 number 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* *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-toone 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].

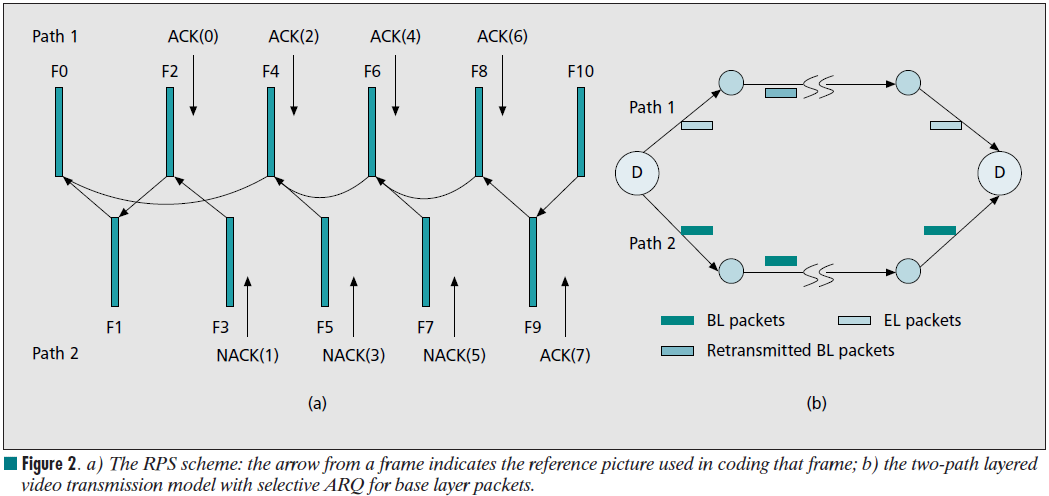
**ADVANTAGES AND DESIGN TRADE-OFFS**

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.

**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-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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附件2（译文，范例）：

通过自组网的多径视频传输

SHIWEN MAO，弗吉尼亚理工学院暨州立大学

SHUNAN LIN, 美国哈雷公司

YAO WANG, SHIVENDRA S. PANWAR, AND YIHAN LI, 纽约大学理工分校

**摘要**

实时多媒体传输有严格的带宽、延迟、和丢包率要求。由于无线自组网频繁链路故障和阻塞的特性，在无线自组网中支持这些应用是一个巨大的挑战。为一个实时的多媒体会话使用多条并行路径（称作多径传输）可以为设计健壮的多媒体传输系统时提供一点新的自由选择。在本文中，我们描述了一个通过无线自组网的多径视频传输的框架，并研究了它的基本组成部分，包括多数据流视频编码，多径路由和传输机制。我们以三个代表性的例子来说明如何扩展现有的视频编码方案以充分发掘多径传输的潜力。我们还研究了在不同层次支持通过自组网进行多径视频传输的重要机制。我们的实验表明在自组网上进行高效视频通信，多径传输是一个很有前景的技术。

**简介**

随着无线技术的最新进展，无线网络正成为当今的网络接入的重要组成部分。自组网是没有基础设施的无线移动网络，其中的移动节点相互合作以寻找路由并转发数据包。这种网络可以在基础设施不可用或难以设置的地方快速部署，并日趋完善地成为一种提供无所不在的无缆通信的手段。对自组网的用户，比如现场急救员，搜索救援队，军事单位等，显然需要其能够提供视频服务。这些具有丰富内容的服务比简单的数据通信更有价值：它会提高自组网的效能并促进自组网的广泛部署。

在视频通信中，接收器通常会显示连续接收到的视频。这种连续显示要求视频数据的及时交付，可以进一步解释为在下层网络上严格的服务质量（QoS, quality of service）要求（例如，延迟，带宽和丢包率）。为保证接收到的视频成功重建，视频会话使用的路径必须在视频会话的大部分时间内保持稳定。而且，由于传输错误造成的分组丢失和拥塞引起的逾期交付应当被保持在较低水平，这样他们就可以通过差错控制和错误隐藏技术处理。然而，这种情况在自组网中并不成立，在这里无线链路频繁中断并且依靠其灵活性重建。而且，由于遮蔽、衰减、路径丢失以及其他发送用户的干扰，一条无线链路有着很高的传输错误率。因此，为了高效的视频传输，传统的差错控制技术，包括前向纠错（FEC, forward error correction）和自动重传请求（ARQ, automatic repeat request），应当考虑到适应频繁的链路中断和高传输错误。此外，应通过在各个层次中的共同考虑和优化机制，包括视频编码，差错控制，传输机制和路由，在视频传输系统设计时采用一个全面的方法。这种方法通常被称为跨层优化。

在各种机制中，使用多条路径进行端对端会话传送数据的多径传输，十分适用于自组网，这里的网络拓扑表明任何一对源节点和目的节点之间都存在多条路径。多径传输数据已经被应用在不同场合[1]。最近，出现了对实时多媒体应用使用多径传输的大量研究[2-13]。例如，多径传输已经结合多描述编码（MDC, multiple description coding）[2-6, 9-11]，ARQ[6]和FEC[12]进行视频传输。它已经表明，当与适合的源和/或信道编码以及差错控制手段结合，多径传输可以显著提高通过传统基于最短路径路由手段的媒体质量。这也开启了早先和目前在互联网工程任务组（IETF, the Internet Engineering Task Force）[13, 14]的多径传输协议的标准化工作。

在这篇文章中，我们研究了在自组网中使用多径传输进行视频应用的问题，并讨论相关事项和技术。我们证明，多径传输在为自组网设计健壮的视频传输系统时提供了一个新的自由层面。我们展现了一般应用的前景，以及为视频通信使用多径传输所带来的好处和设计权衡。然后，我们在后面的章节讨论了相关问题，包括多数据流视频编码和网络和传输的讨论。性能研究用来证明多径传输技术在通过自组网进行视频传输时的效果。然后我们结束了这篇文章。

**多径多媒体传输体系结构概述**

视频流的多径传输的一般体系如图Fig.1a所示。在发送端，原始视频首先被视频编码器压缩成M条数据流。当M>1，我们称此编码器为多数据流编码器。然后，数据流被流量分配器分割并分配到K条路径。这些路径是由多径路由协议维护的。当数据流到达接收端时，他们首先被放进一个重新排序缓冲区以重建原始顺序。最后，视频数据被从重新排序缓冲区提取出来被解码并显示出来。若其中任何一个子串流丢失，视频解码器将会执行适当的错误隐藏。通常情况下，路径的质量会随时间而改变。我们假定系统收到有关网络QoS参数的反馈。尽管并非必要，但是这种反馈可以被用来令编码器和传输机制适应于网络条件（例如，编码器可以执行基于反馈信息的速率控制，以避免网络拥塞）。在实践中，可以采取发送端使用预先设计的多数据流编码器，它总是产生一个固定数量的数据流（比如，2到4）。另一方面，可用路径的数量，以及他们的带宽，会因网络拓扑变化和拥塞而随时间改变。因此，就像图Fig.1a中M≠K时，流量分配器负责将视频数据包从M条数据流分发到K条可用路径[8]。

图Fig.1a中的点对点架构可用于双向对话服务，以及单向流服务。对于后者而言，它可以扩展到更一般的情形。例如，如图Fig.1b所示的多对一类型的应用的架构，其中一个节点从多个服务器并发地获取视频剪辑。每个视频服务器（例如，节点B、C或D）都是自组网内的移动节点，在其缓存或公共目录中存在目标视频（或目标视频的某些部分）[4]。一个基于多播的体系结构如图Fig.1c所示，其中一个源将一个视频多播给并行使用两个多播树的一组节点[15]。

**优点和设计折衷**

使用多径传输的优势已经被许多以往的文献报道，例子参见文献[1-15]。使用多径传输的一个重要优势是其固有的路径多样性（即路径失效处理的独立性）。因此，接收端在任何时候总能接受到一些数据，除非所有的路径同时中断，这种情况的发生远远比单路径故障罕见得多。人们可以联合设计源编码器，多路径路由算法，流量分配器，探讨路径的多样性以优化系统的整体性能。此外，多径传输为视频会话提供了一个更大的聚合率，从而减小由有损视频编码器造成的失真。最后，多径传输更均衡地分配网络中的流量负载，降低网络的拥塞和延迟。

这些优势也带来了较高的编码冗余度，较高的计算复杂性和较高的控制流量成本开销的代价。一般而言，使用更多的数据流和路径会增加丢包和路径故障的健壮性，并且通过更好的负载平衡减少网络拥塞。然而，更多的数据流可能会增加相同视频质量的视频比特率，并导致在流量分割和重新排序时更高的计算开销和延迟。在动态自组网环境下维护多条路径涉及更高的控制流量开销和更复杂的路由/路径选择算法。文献[10]的研究表明，最显著的性能增益在K从1增加到2时获得，进一步对K的增加只能取得较小的改善效果。因此，一个具有M=2和K=2的基准系统用适度的代价提供显著的性能收益。

**多数据流视频编码**

为使多径传输可用于发送压缩视频，必须仔细设计视频编码器的生成数据流，这样一个数据流的丢包不会对其他数据流的解码产生不利影响。然而，这种相对独立的数据流之间不应该靠在编码效率上的巨大的代价来获得。因此，多数据流编码器应力求在编码效率和错误恢复能力之间得到良好的平衡。此外，在设计源编码器时必须考虑在传输层差错控制方面什么是可行的。 在本节中，我们说明了为获得更好的性能如何令视频编码器以适应多径传输。我们使用三种代表性的编码手段作为例子，在其业务和网络需求方面各不相同。

**基于反馈的参考帧选择**

一个简单的生成多径视频数据流的方法是将一个视频用标准方法编码，然后将这些数据流分散到多条路径（例如，发送偶数帧对应的位数据到一条路径上，奇数帧对应的到另一条）。但是，由于两条路径相互依存，这种简单的方法性能不高。也就是说，偶数帧由先前（奇数）帧推测而来，反之亦然。

我们通过探讨参考帧选择法（RPS, reference picture selection）技术[6]改进了这个方法。具体来说，我们仍然使用一样的时域分割方法（即分别发送已编码的偶数和奇数帧）。但是，我们使用了一个更具有网络感知的编码方法，它基于反馈选择参考帧并建立路径状态。假定如果一个帧损坏或丢失，解码器就对其发送一个否定确认（NACK, negative acknowledgment），否则就发送确认（ACK）。然后编码器就可以估计出路径状态并推断先前的哪些帧被损坏了。基于以上估计，对一个已编码的帧，最接近它本身并在更好的路径上像参考帧一样传送的帧被选择作为参考帧。

图2a是一个提出的RPS方案的例子。当正在编码4号帧时接收到NACK(1)，编码器推断由于错误的传播，2和3号帧不能被正确地解码。因此，0号帧被选作4号帧的参考帧，路径2被置为“坏”状态。当6号帧被编码时，编码器使用4号帧代替5号帧作为参考帧，因为路径2仍然为“坏”状态。当接收到ACK(7)时，路径2切换到“好”状态。然后，9号帧被选作10号帧的参考帧。

RPS方案提供了一个在编码效率和错误恢复之间的良好平衡。也就是说，当两条路径都是有效的，RPS使用临近帧作为参考帧，从而达到尽可能高的预期增益和编码效率；当一个路径失效时，编码器避免使用任何在那条路径上传送的帧，从而最大限度的减小错误传播时间。注意到RPS方案仅仅应用在在线编码，因为它适用于基于信道反馈的编码操作。

**性能评估**

使用基于理论信道模型的仿真对前面讨论的三个方案进行研究和比较。结果显示在文献[6]。我们发现，这三种技术所适用的特定的环境，依其反馈信道可用性，端对端时延约束和路径错误性质而定。

接下来我们使用OPNET仿真和MDMC编解码器对比研究了一个多径传输（MPT, multipath transport）系统（使用两条路径）和一个单一路径（SPT, single path transport）系统在自组网中的视频流传输。对MPT，使用一个DSR算法的多径路由扩展或MDSR[6]，以及MRTP/MRTCP。SPT包括DSR路由协议和RTP/RTCP。对于这两个系统，MDMC编解码器用于生成两个描述，每个速率为59 kb/s。对MPT系统，两个描述被发送到MDSR维护的两条最大不相交路径上。对SPT系统，这两个描述复用DSR建立的单一路径。两个描述的数据包在发送前交错到两个帧里以减小突发丢失的影响。在两个仿真中都使用播放缓冲区。自组网包括16个节点，在600m×600m的区域内以5m/s的速度的随机基准点移动模型（2s暂停时间）。

我们观察到在某个描述中丢失时峰值信噪比降低。此外，最大的降低发生在一个描述中的很大的突发丢失与其他流的丢失突发重叠时。可以看出，SPT比MPT有更高的丢失率。此外，仔细研究表明，SPT的这两种丢失痕迹有高度相关性。因此，图Fig.5a中的PSNR曲线比图Fig.5b中更频繁和剧烈地降低。在实验中，MPT在平均PSNR上比SPT得到显著的1.26dB增益。

为验证通过自组网进行视频传输的可行性并评估以现行技术可以达到的视频质量，我们使用了一个自组网多径视频流平台。该平台包括四台带有802.11b网卡的IBM Thinkpad笔记本电脑。一台笔记本电脑被选为视频源，一台作为视频接收端，剩下的两台作为中继节点。在这个四个节点的自组网中，源节点到目的节点之间使用两条静态路由，每一条都经过一个中继节点。我们实现了MRTP的关键功能，包括流量分割与重组、时间标记、顺序编号和QoS反馈功能。带有ARP和MDMC的LC编解码器也被实现并用在此实验平台上。

实验平台设置的详情和实验结果可以参见文献[6]。我们的实验表明，在考察视频比特率范围、背景流量、移动速度之后，用带有ARP和MDMC的LC实现的视频流的质量是可以接受的。我们在此平台上的实验证实使用多径传输和多数据流编码在自组网上传输视频的可行性。

**结语**

在本文中，我们描述了一个通过无线自组网进行多径视频传输的框架，并研究了他的基本组成部分，包括多数据流视频编码，多径路由和传输机制。我们提出了每个组成部分的例子的解决方案以及集成这些组件的系统的性能改善。我们证明，多径传输结合适当的视频编码技术可以带来超过使用单一路径的系统的显著收益。

值得注意的是，这些方案也适用于可以通过多径接入路由器产生路径多样性的有线网络（例如，互联网）。而且，当无法使用重传时，人们通常会考虑在同一个数据流中应用FEC以降低在数据流内以及跨越不同路径交付的包的丢包率[12]。但是，这些措施会引起额外的延迟和信道编码冗余，而且可能在应对突发丢失时并不有效。我们的研究表明，在考虑丢失的环境中，无FEC的MDMC和RPS已经足够了。

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