Real Time Transport With Path Diversity[1]

Shiwen Mao[†], Shunan Lin[†], Dennis Bushmitch[‡], Sathya Narayanan[‡],

Shivendra S. Panwar[†], Yao Wang[†], and Rauf Izmailov[*]

[†]Electrical and Computer Engineering Department,

Polytechnic University, 6 MetroTech Center Brooklyn, NY 11201

[‡]Panasonic Information Networking Technologies Lab (PINTL)

2 Research Way, Princeton, NJ 08540

[*]C&C Research Laboratories, NEC USA Inc.

4 Independence Way, Princeton, NJ 08540

ABSTRACT - Realtime transport poses great challenges in network research. In this paper, we propose realtime transport using multiple paths. We show the advantages of using multiple paths for streaming services, namely: (1) better error resilience performance, (2) better traffic characteristics and hence better queueing performance, and (3) inherent security advantages. We are working on developing a new protocol to support realtime transport with multiple paths. We are also working on a testbed to implement the ideas and demonstrate the performance of our proposal.

KEY WORDS - real time, long-range dependence, video streaming, multipath transport, Multiple Description Coding

I. INTRODUCTION

There has been a call for the development of new applications in the network research community lately [1]. Among the emerging and existing applications, video streaming is a good candidate for such "killer" applications. According to a TIME article, there are more than 4000 web sites providing free downloads of *The Simpsons* TV show [2]. Emerging peer-to-peer file sharing service providers such as Napster and Morpheus also show the strong market drive for video and high quality music streaming services.

Real time traffic transport, with its stringent delay, jitter, and bandwidth requirements, poses great challenges to network research. First, in a wireless network consisting of heterogeneous devices such as handheld devices and notebook computers, the bandwidth of the wireless links is usually limited. High rate video streams may cause congestion on the links, and retransmission is often not feasible because of the packets carrying the video frames arrive too late. Second, in a mobile ad hoc wireless network, interference among the transmitting nodes causes high bit error rates. Wireless links can come up and go down in an unpredictable manner because of the movement of the nodes. For video streams encoded with the motion prediction and compensation technique, errored or lost packets in a frame not only degrade the display quality of the current frame, but also cause error propagation along the following frames. This is because that the decoding of the following frames depends on the correct reception and reconstruction of the current frame. The popular videoconference tool *vic* avoids using the motion prediction/compensation techniques and encodes all video blocks in intra-mode to cope with this error propagation problem [3]. Third, the Long-Range Dependence (LRD) phenomenon in variable bit rate (VBR) video traffic causes the *buffer ineffectiveness* problem, where buffering is no longer an effective means to reduce loss for such traffic [4]. Considering the stringent delay requirement of real time traffic, one has to provide extra bandwidth to achieve a low loss rate, which causes low bandwidth utilization. Fourth, security of video/audio transport is also an important issue. It is desirable to prevent hackers from intercepting a video stream.

On the other hand, there are some nice features of the wireline and wireless networks that can be exploited to cope with these challenges. The cost of wireless/wireline network interface cards (NIC) is dropping, making it possible to install multiple cards in one node. Hence more bandwidth is available. In a mobile ad hoc network, the meshed nature of radio connections means that it is likely that multiple disjoint paths exists between two mobile nodes. Furthermore, most of the ad hoc routing protocols proposed, e.g. DSR [5], ZRP [6], can return multiple paths between a source and a destination node in response to a route discovery query. In wireline networks, the Stream Control Transmission Protocol (SCTP) [7] also provides multiple disjoint paths for a source and destination pair. With SCTP, a primary path is chosen for data transmission and all other paths are used as backups. A primary link failure can be quickly recovered by switching to a backup path, and fast retransmission of lost packets can be done on backup paths [7].

We propose the transport of real time traffic with path diversity in this paper. Real time traffic can be traffic from an interactive videoconference, downloading of streaming video, or high quality music streaming. The network can be a wireline

[1] This work is support in part by NSF Grant ANI 0081357 and by the New York State Center for Advanced Technology in Telecommunications (CATT).

network or a mobile wireless ad hoc network. In a wireless setting where the link bandwidth is low and the links are more unreliable, the load balancing and path redundancy of multipath routing is more appealing. We assume the routing protocol returns multiple maximally disjoint paths between the source and destination node [8]. The source splits the video traffic and sends each piece to a different path. Note that our proposal is different from previous research work, such as a bundle of Real-time Transport Protocol (RTP) [9] sessions or a SCTP session [7]. In the former case, each RTP session in the bundle hold its own integrity, i.e., each RTP session is an audio stream or a video stream. A single stream is never split and sent to multiple paths. In the latter case, the SCTP protocol uses only *one* primary path for data transmission, while all other paths are maintained as backups for failure recovery and retransmissions.

As we mentioned before, it is possible to perform multipath transport in today's wireline or wireless networks. There are several advantages in traffic splitting among multiple paths. First, each piece of the real time traffic requires less bandwidth on each path. Hence the link congestion probability is reduced. Second, since the real time traffic is sent through multiple disjoint paths, the loss of the packets on different paths will be independent to each other. It is less likely that all the paths being used are down at the same time. With appropriate error concealment schemes or joint source channel coding techniques, the end-to-end transport can be more robust to wireless link failures. Thirdly, there are many ways to split a real time traffic stream into multiple sub-flows, which provides opportunity for traffic shaping to get flows with better queueing performance [4,15]. As we will show later, *stripping*, *thinning*, and *shuffling* of video traffic break up short-term correlation in video traffic and thus improve its queueing performance. Fourth, our multipath transport scheme has inherent security advantages. With multiple path transport, an eavesdropper not only has to intercept all the sub-flows, but also has the knowledge of how the data partitioning is performed, before he can reconstruct the original real time flow. The disadvantage of multiple path transport is the additional complexity in data partitioning, establishing and maintaining of multiple paths between the source and destination, the additional resequencing buffer required to reassemble multiple flows into a single stream, and the associated delays. Our previous work shows that resequencing delay and buffer requirement are moderate if multi-path routing is used in conjunction with enhanced versions of current transport and network layer protocols [14].

We are working on a new protocol called Multi-flow Realtime Transport Protocol (MRTP), incorporating all these ideas and features. Our research group is also developing a mobile wireless ad hoc network testbed built on the IEEE 802.11b MAC.

The rest of this paper is organized as follows. In Section II we present our simulation study of three new video transport schemes using path diversity in mobile ad hoc networks. These preliminary results show that path diversity makes video streaming more robust to link failures and transmission error, and results in higher replay quality. In section III, we show that thinning, stripping, and shuffling of video traffic improve its queueing performance. We also observe multipath transport of real time traffic is more secure to eavesdropping or interception in Section IV. Ongoing work at Polytechnic University is reported in Section V. Section VI concludes the paper.

II. Error Resilience Performance

In this section, we present our simulation study of three new video transport schemes using path diversity in mobile ad hoc networks.

A. Video Transport Schemes Using Multipath Transport

Multipath transport schemes have been proposed in the past for wireline networks for increased connection capacity, as well as for reliability. We feel that multipath transport has more potential in wireless networks where individual physical links may not have adequate capacity to support a high bandwidth service, and the links are more unreliable. The idea of utilizing path diversity in wireless ad hoc networks for multimedia data transmission was proposed in [10], which mainly considered image transmission. Recently, several error resilient video coding and transport control techniques have been proposed for video transmission using path diversity, especially in a wireless ad-hoc network environment. In [11], a feedback-based reference picture selection (RPS) scheme for video transmission over multiple paths was proposed. First, a video stream is partitioned into a number of subflows on the frame bases, e.g., if two paths are available, all the even frames are sent on one path and all the odd frames are sent on the other. The receiver notifies the sender of the latest correctly received frame. The video encoder always selects the latest correctly received frame as the reference picture for the frame it is encoding. The scheme can achieve high error resilience at moderate cost in coding efficiency. Layered coding combined with a selective ARQ transport scheme was proposed in [12]. With this scheme, base and enhancement layer packets are transmitted over different paths. Only a base layer packet is allowed to be retransmitted, and the retransmitted base layer packet is sent on the path used by the enhancement layer. This scheme can significantly reduce error propagation in the re-constructed frames at the cost of retransmission delay. Both of the above schemes are workable only when feedback is available within a few frame times.

If feedback is unavailable, multiple description coding (MDC) is a natural option for multiple path transmission. MDC refers to a coding method that generates two or more correlated bit-streams so that a high-quality reconstruction can be obtained from all the bit streams put together, while a lower, but still acceptable, quality reconstruction is guaranteed if only one bit stream is received. A multiple description video coding technique, dubbed multiple description motion compensation (MDMC), was proposed in [13]. MDMC predicts current frame from two previously encoded frames and transmits different descriptions over different paths. By varying the coding parameters, it can achieve the desired trade-off between redundancy and distortion. Interested read can refer to [13] for details.

We also simulate two other schemes for video transmission over the two-path environment for comparison purpose, namely: the video redundancy coding scheme (VRC) [16] and the alternative Group of Blocks (Alt-GOB) transmission scheme. VRC is an error resilient video coding technique that generates several independent bit streams by using independent prediction loops. In the special case of two descriptions, an even frame is predicted from the previous even frame, and an odd frame is predicted from the previous odd frame. Encoded even frames are sent on one path and encoded odd frames on the other path. In the Alt-GOB transmission scheme, even GOBs and odd GOBs are sent to two paths alternately. In the decoder, the missing GOBs are concealed using the motion information from above GOBs.

B. Simulation study

To evaluate the performance of these three multipath transport schemes, the Quarter Common Intermediate Format (QCIF) sequence "Foreman" (frame 1 to 200, QCIF) are encoded at 10 frame per second frame rate. We assume the allocated bandwidth on each path for source coding is 57kbps. For both the RPS and the Layered Coding with ARQ techniques, the feedback time is assumed to be less than 300ms. In MDMC, please refer to [13] for the parameter settings. Note that for the MDMC method, its optimal coding parameters are determined by the characteristics of the source and the channels. It is likely that some other choices of the coding parameters may yield better results for MDMC. In all the cases, 5% macroblock level intra refreshments are used to combat error propagation. One group of blocks (GOB) is packetized into a packet. In the layered coding with ARQ scheme, the base layer packets are transmitted on the better channel if the two channels have different error characteristics. In the VRC simulation, the 2-5 mode is used when the two channel packet loss rates are (3%, 3%) and the 2-3 mode is used for the loss rates of (10%, 10%) and (5%, 10%), based on the recommendation given in [15].

To simulate video transmission over ad-hoc networks, a multi-hop channel model was used to generate bursty packet loss patterns. We assume that multiple paths can typically be set up for two end users and each path consists of multiple links. A three-state Markov model is used for each link with the three states representing the "good", "bad" and "down" status of the link, respectively. The "down" state means the link is totally unavailable (loss rate is 1). The "good" state has a relatively low packet loss rate as compared to the "bad" state. The packet losses are assumed to consist of packets lost due to link failures or FEC failures. In our simulation, two paths were set up for each connection, and each path was continuously updated as follows: After every two seconds, four links were chosen randomly from a link pool to construct a new path. Each link had its own state transition parameters and packet loss rates at the states. The parameters are chosen to match the characteristics of a typical wireless link [17]. A video packet can go through a path correctly only when it goes through every link successfully. For each pair of specified average loss rates, ten packet loss traces were generated according to the above multi-hop channel model. Because of the lack of experimentation and measurement data in wireless ad hoc networks, it is not clear that if the approach used here models a wireless multihop path very well. But we believe these simulations can provide initial insights on the performance of the multipath transport proposal. We are working on a more realistic simulation model using OPNET Modeler.

The average *PSNR*s of decoded video sequences are given in Table 1. From this table, we can see that: (1) All three proposed schemes outperform VRC and Alt-GOB. (2) The layered coding with ARQ scheme has the highest decoding quality when packet loss rate is high, especially for unbalanced channels. (3) For channels with low error rates, MDMC and RPS outperform layered coding with ARQ.

Table 1. Average PSNR of Decoded Images

Packet Loss Rate	**3%,3%**	**10%,10%**	**5%,10%**
RPS	31.3	27.5	28.8
Layered Coding+ARQ	31.1	29.4	30.6
MDMC	31.3	26.8	27.9
VRC	30.1	24.8	25.3
Alt-GOB	27.73	23.26	24.20

III. TRAFFIC PARTITIONING OVER MULTIPLE PATHS

In this section, we report our experiments with traffic partitioning for multipath transport. It is shown that better load balancing and queueing performance are obtained.

A. LRD in VBR Video Tranffic

It is well known that VBR video exhibits the LRD phenomenon. The autocorrelation function of a VBR video trace decays much slower than that of the Short Range Dependent (SRD) traffic. On queueing performance, LRD causes the *buffer ineffectiveness* problem [4] in queues, where buffering is no longer an effective means to reduce loss for such traffic. Extra bandwidth has to be provided to achieve a low loss rate, which results in low bandwidth utilization.

LRD traffic usually can't be regulated by a piece-wise linear regulator, e.g. a leaky bucket type regulator. Using such regulator, either a much higher long-term average rate is required, or a large portion of the incoming traffic is marked or dropped. In [18], a fractal leaky bucket regulator is proposed with an envelope process of the form of a linear term plus a t^H term, where H is the Hurst parameter of the video trace. Although no excess dropping of incoming traffic occurs, the output process of the regulator is as bursty as the incoming traffic. So there will be no improvement on the queueing performance.

B. Traffic Partitioning For Better Queueing Performance

We propose to split a highly bursty real time stream into a number of sub-flows for multipath transport. There are many ways to perform traffic splitting, hence providing opportunity for traffic shaping to get flows with better characteristics.

Assume the real time traffic is divided into equal sized blocks, e.g. a frame per block. We suggest and evaluate the following traffic splitting techniques that improve the characteristics in the short range region [19]:

- Thinning: The process of creating a new media stream by means of picking the blocks of the original traffic process in their increasing order and periodically skipping every S blocks in between.
- Striping: The process is defined as follows: (1) Each data stream is partitioned into blocks of equal size B. (2) For each block the destination striped substream is determined by using a round robin order. (3) Blocks from multiple streams are interleaved for each path.
- Shuffling: the shuffled data stream is constructed from the original sample sequence by means of a shuffling permutation rule. This technique may or may not be used in conjunction with multipath transport.

Although these techniques have been proposed in previous work, application of these techniques in a wireless setting and to real time traffic is still very interesting. Also we investigate their impact on queueing performance from the resulting correlation structure.

We concentrate on a single MPEG4 traffic source without aggregation and study its queueing performance on the time scales characteristic of the multimedia client-server video streaming environment (about 100 to 300 ms buffer size). We show that while both LRD and SRD influence the queueing performance for some time scale of the queueing system, it is the SRD properties within the critical time scale (CTS) that are dominant in determining the buffer efficiency of the queue. Although none of the three traffic shaping techniques are shown to decrease the degree of LRD in data, all the schemes break up short-term correlation in video traffic and improve its queueing performance.

Part of the experiment results with shuffling is presented here. Fig.1 shows the autocorrelation of the original and shuffled "Jurassic" video traces. It can be seen that shuffling greatly reduces the short-term correlations in the video trace within CTS of the system. Fig.2 plots the tail distributions of the single-frame shuffled Jurassic traces as well as that of the original trace. Lower short-term correlation within the CTS gives better queueing performance.

IV. Security Considerations

Security is an important issue in transport of real time traffic in networks. Usually, security functionality is classified into a number of security services [20]. While our Multipath Transport proposal can provide all other security services, it has the inherent strength in providing *confidentiality*.

Confidentiality is a security service that provides resistance to the security attack known as interception or eavesdropping. With our Multipath Transport scheme, each sub-flow only contains partial information of the original flow. An eavesdropper has to intercept all the sub-flows before he can reconstruct the original real time stream. In some cases, the more important part of the real time stream can be encrypted, or sent through a secure path, while other sub-flows are sent as they are. Interception of one or two sub-flows without the critical part would be of little use to the attacker. Furthermore, even if an eavesdropper can intercept all the sub-flows, he has to know how the traffic partitioning is performed in order to reassemble the original stream. Hence multipath transport is more robust to such attacks.

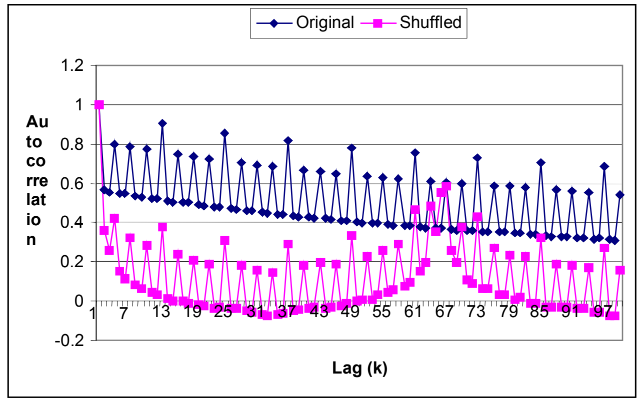

Fig.1 Autocorrelation of the original and shuffled "Jurassic" Trace (S=64, R=64, B=1).

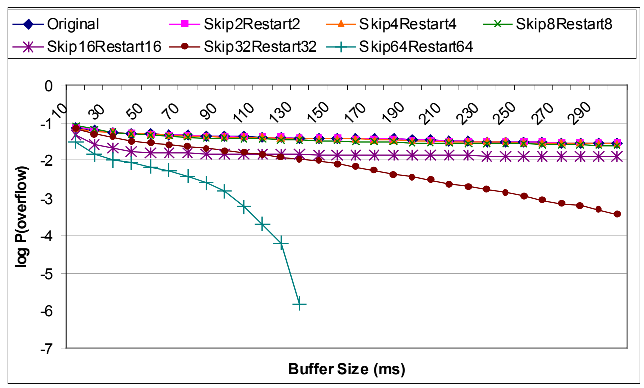

Fig.2 Queueing performance of shuffled video traces, ρ=0.5, B=1.

V. Ongoing Work

This section presents a couple of research projects we are working on.

A. A Multi-Flow Realtime Transport Protocol

We are working on a new protocol, Multi-flow Realtime Transport Protocol (MRTP), which incorporates the ideas and provides basic support for realtime transport using multiple paths. The transport service provided is an end-to-end service using an association of multiple flows. A companion control protocol, Multi-flow Realtime Transport Control Protocol (MRTCP), is also proposed to provide session and flow management and feedback control. MRTP/MRTCP can be viewed as an extension of the RTP/RTCP protocol [9], with the difference that MRTP/MRTCP supports multi-homed hosts, multiple flows in a single session, and that it doesn't require integrity within individual flows.

Fig.3 shows the layer positioning of MRTP/MRTCP. Similar to RTP, MRTP/MRTCP is intended to be malleable to provide the information required by a particular application and will often be integrated into the application rather than being implemented as a separate layer [9]. The basic services provided by MRTP/MRTCP are:

- Session management: maintaining multiple flows
- Participant identification
- Payload type identification
- Sequence numbering
- Timestamping
- Statistics for the session and for each flow
- Optional retransmission support

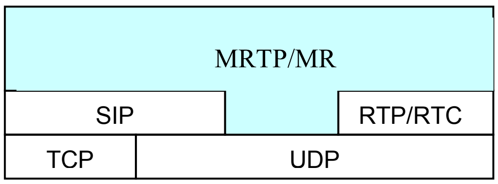

Fig.3 The positioning of the new protocol

B. Testbed Development

We are developing a testbed to verify and demonstrate the performance of our real time multipath transport proposal.

In the testbed, each mobile node is a notebook computer equipped with multiple IEEE 802.11b PCMCIA cards. The cards are set to work in ad hoc mode. Fig.4 shows the main components of the testbed.

VI. Conclusion

Realtime transport poses great challenges in network research. In this paper, we propose realtime transport using multiple paths. We show the advantages of using multiple paths for streaming services, namely: (1) better error resilience performance, (2) better traffic characteristics and hence better queueing performance, and (3) inherent security advantages. We are working on developing a new protocol to support realtime transport with multiple paths. We are also working on a testbed to implement the ideas and demonstrate the performance of our proposal.

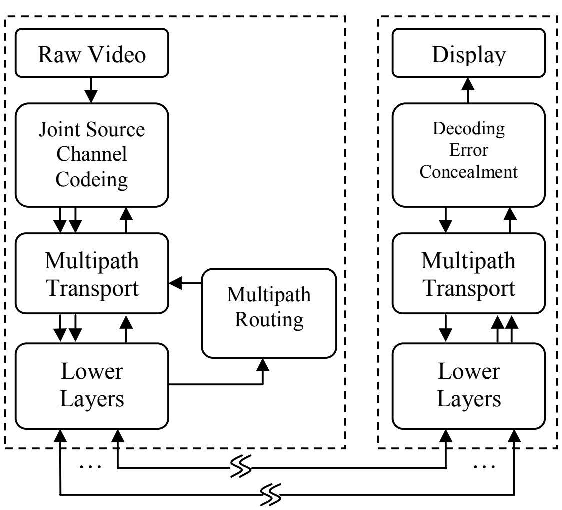

Fig.4 Main components of the testbed

References

[1] C. Diot, M. Nagshineh, D. Towsley, J. Turner, and L. Zhang, *Panel: Defining The Next Generation of Challenges in Networking Research*, Moderated by J. Kurose, *IEEE INFOCOM 2002*, June 2002.

[2] A. Hamilton, “The pirates of prime time,” *TIME* magazine, pp.54-56, Feburary 25, 2002.

[3] S. McCanne, V.Jacobson, "vic: a flexible framework for packet video," in *Proceedings of the ACM Multimedia'95*, 1995.

[4] A. Erramilli, O. Narayan, and W.Willinger, “Experimental queueing analysis wit Long-Range Dependent packet traffic,” *IEEE/ACM Transactions on Networking*, vol.4, no.2, pp.209-223, April 1998.

[5] D. B. Johnson, D. A. Maltz, Y.-C. Hu, and J. G. Jetcheva, "The dynamic source routing protocol for mobile ad hoc networks (DSR)," *IETF Internet Draft*, November 2001.

[6] Z. J. Haas, M. R. Pearlman, and P. Samar, "The zone routing protocol (ZRP) for ad hoc networks," *IETF Internet Draft*, July 2002.

[7] R. R. Stewart and Q. Xie, *Stream Control Transmission Protocol (SCTP): A Reference Guide*, Adison-Wesley, 2001.

[8] M. R. Pearlman, Z. J. Hass, P. Scholander, and S. S. Tabrizi, "On the impact of alternate path routing for load balancing in mobile ad hoc networks," in Proceedings of *ACM MobiHOC'2000*, August 2000.

[9] H. Schulzrinne, S. Casner, R. Frederick, and V. Jacobson, "RTP: A transport protocol for real-time applications," *IETF RFC 1889*, available at: http://www.ietf.org.

[10] N. Gogate, D. Chung, S. S. Panwar, and Y. Wang “Supporting image/video applications in a mobile multihop radio environment using route diversity and multiple description coding”, to appear in the *IEEE Transaction on Circuits and Systems for Video Technology*.

[11] S. Lin, S. Mao, Y. Wang, and S. S. Panwar, "A reference picture selection scheme for video transmission over ad-hoc networks using multiple paths," in Proceedings of the *IEEE International Conference on Multimedia and Expo 2001*, Tokyo Japan, August 2001.

[12] S. Mao, S. Lin, S. S. Panwar, and Y. Wang, "Reliable transmission of video over ad-hoc networks using Automatic Repeat Request and Multi-path Transport," in *Proceedings of the IEEE Fall Vehicular Technology Conference 2001*, vol.2, pp.615-619, Atlantic City, October 2001.

[13] 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, issue.6, pp.438-452, June 2002.

[14] N. Gogate and S. S. Panwar, "On a resequencing model for high speed networks," in *Proceedings of INFOCOM'94*, vol.1, pp.40-47, 1994.

[15] University of British Columbia, "H.263+ codec," avaliable at ftp://dspftp.ece.ubc.ca.

[16] S. Wenger, "Video redundancy coding in H.263+,", in *Proceedings of AVSPN*, September 1997.

[17] M. Veeraraghavan, N. Cocker, and T. Moors, "Support of voice services in IEEE 802.11 wireless LAN," *in Proceedings of INFOCOM 2001*, April 2001, Anchorage, Alaska.

[18] N. L. S. Fonseca, G. S. Mayor, and C. A. V. Neto, "On the equivalent bandwidth of self-similar sourceses," *ACM Transactions on Modeling and Computer Simulation*, vol.10, no.2, pp:104-124, April 2000.

[19] D. Bushmitch, R. Izmailov, S. Panwar, A. Pal, "Thinning, striping and shuffling: Traffic shaping and transport techniques for variable bit rate video", to appear in *GLOBECOM'02*.

[20] A. Rubin, End-to-end web security and e-commerce, Tutorial of INFOCOM 2002, June 2002, New York, New York.