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OPTIMIZING AIRBORNE NETWORKING PERFORMANCE WITH CROSS-LAYER DESIGN APPROACH

City University of New York

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We analyzed the impact of the physical propagation environment on the performance of mobile ad hoc network (MANET) protocols. Dynamic Source Routing (DSR) protocol is chosen as typical routing protocol for the experiment. We used the network simulator OPNET to design our airborne network (AN) environment model. Our approach utilized the experimental results that we have obtained at the Air Force Lab in our design to model a cross layer design to reduce the impact of the physical propagation parameters in AN environment. We evaluated the performance of MANET-based protocol under Air Force environment. We designed a delay tolerant network (DTN)-based probabilistic routing protocol to work in AN environment that is associated with intermittent connectivity.

15. SUBJECT TERMS

Airborne Networking, Cross-Layer, Mobile Ad-Hoc Network

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1.0 Summary Executive

In this research report, we focus on enhancing the airborne networking (AN) architecture and protocols by the inclusion of cross layer design (CLD) approach to optimize the performance of AN. Our study aims to improve end-to-end message delivery ratio in a multi-hop scenario where links are very dynamic and resources are limited. Furthermore, it can provide ubiquitous and guaranteed network connectivity to all Air Force platforms across dynamic heterogeneous sub-networking, with Quality of Service (QoS) assurance such as minimizing bit error rate (BER), delay and delay variation.

CLD is very essential approach for AN where unpredictable variables such as aircrafts mobility, density, and tasks intended make the diverse and stringent wireless QoS requirements very critical to satisfy. Mobility affects various layers of the protocol stack; at the Data Link Layer (DLL) mobility governs how rapidly the link characteristics vary over time. On the other hand, at the medium access control (MAC) layer mobility governs how long the measurements regarding channel state and interference remain valid. Also mobility leads to rapid changes in network topology and thus governs the performance of the routing protocols.

Since reliable data transmission rate is affected by deterioration of the signal due to physical characteristics of the channel; noise, path loss, multi-path effect, interference, mobility, limited bandwidth and limited transmission power, Open System Interconnect (OSI) and Transport Control Protocol/Internet Protocol (TCP/IP) stack becomes inflexible in design. Also since certain QoS guarantee is required. An alternate and optimal solution of CLD is needed. There is strong dependency between the different layers, and changes in one layer affect other layers performance metrics. This interdependency is stronger between the physical layer, MAC and network layer.

In AN environment, mobile wireless networks episodically connected because of terrain, weather, jamming, and access schedules; resulting in rapid topology changes, therefore, routing protocol will drop intermittent end-to-end connections. In this proposed architecture, we will overcome this issue by implementing delay-tolerant network (DTN) architecture by moving messages towards destination via store and forward technique that supports multi-routing algorithms to acquire best path towards destination.

We analyzed the impact of the physical propagation environment on the performance of mobile ad hoc network (MANET) protocols. Dynamic Source Routing (DSR) protocol is chosen as typical routing protocol for the experiment. We used the network simulator OPNET to design our AN environment model. Our approach utilized the experimental results that we have obtained at the AirForce Lab in our design to model a Cross Layer Design to reduce the impact of the physical propagation parameters in An environment, see our published work reference [7] and attached in the Appendix. We evaluated the performance of MANET-based protocol under Air Force environment. We designed a DTN-based probabilistic routing protocol to work in AN environment that is associated with intermittent connectivity.

2.0 Technical Approach and Contributions

Airborne Networking (AN) is designed to connect both space using Communications Satellite System and the surface networks, making the AN coherent part of the Global Information Grid (GIG). AN can also operate without connectivity with the ground. AN vary in size from a single aircraft to hundreds of aircraft communicating from anywhere to anywhere. The most essential purpose of AN is to provide ubiquitous and guaranteed network connectivity to all Air Force platforms across dynamic heterogeneous sub-networking, providing access to all needed GIG services and sustaining all needed services with Quality of Service (QoS) assurance in mind for all types of traffic (e.g., voice, video, data, or any combination), references [1] & [2].

Cross layer design (CLD) between the physical layer and the MAC and/or the network layer has been proven to exploit networking in Ad hoc networks which could be implemented to optimize the performance in AN such as but not limited to throughput, latency, bit error rate (BER) and packet loss ratio as we will discuss below. It is critical for AN to meet high QoS to accomplish the intended task, for example resource allocation and prioritization of traffic according to its Class of Service (CoS) (i.e. level of precedence in accordance with the commanders' operational objectives) is vital especially when resources are limited, references [1] and [3].

Delay Tolerant Networking (DTN) is an end-to-end network architecture designed to provide communication in a highly stressed networking environment such as AN where an instantaneous end-to-end path between source and destination may not exist, and the links between nodes may be opportunistic, predictably connectable, or periodically-(dis)connected. Stressed networking environments such as AN characterized by intermittent connectivity, long delays, delay variation, and high BER. DTN architecture has been implemented through the Bundle Protocol (BP). The key capabilities of the BP include custody-based reliability, ability to cope with intermittent connectivity, ability to take advantage of scheduled and opportunistic connectivity [5] and [6].

Current routing protocols favor routing traffic based on shortest path, thus causing a bottleneck. Routing in multi-hop wireless networks (such as AN) using the shortest-path metric is not a sufficient condition to construct good quality paths, because minimum hop count routing often chooses routes that have significantly less capacity than the best paths that exist in the network. Thus, it is desirable to select the routes with minimum cost based on some other metrics which are aware of the nature of the wireless underlying physical channel. In a self-organized network like airborne networking, there are many other metrics to be considered: power, packet loss, maximum available bandwidth etc., these metrics should come from CLD approach in which the network layer is aware of the state of the physical layers.

Air Force environment imposes different constraints occurring in different layers of the airborne networking protocol stack. One CLD approach is to propagate physical layer parameters that reflects its state to the network layer, in particular SNR that will enhance the performance of the DTN protocol in AN. The network throughput will greatly improve and average packet delay will significantly decrease. SNR experienced by mobile terminal (aircraft) is complex mobility-dependent stochastic process resulting in a fading components each of which significantly influence the performance of the wireless channel.

Mobility of the user affects both multi-path fading and shadowing. Multi-path fading is caused by multiple path propagation of the wave between transmitter and receiver. Shadowing is caused by loss of line of sight between transmitter and receiver due to shadowing of the propagating wave by large obstacles. While moving between shadowers, the received signal power varies in accordance with alternating interruptions and release of the line of sight between the transmitter and receiver. Therefore, with increase of mobility, the coherence time of the channel degrades due to the increase in Doppler speed.

Variable link quality effects lead to unpredictable packet errors causing loss of the packet that could be vital in the battle field. So, when the quality of the link degrades the link layer must adapt to the changes, by increasing the transmit power or using a better coding scheme. This would temporarily solve the problem if the change in SNR is due to a random fluctuation. This will cause large number of routing updates thus, increasing the routing overhead, at the transport layer the packet loss could be attributed to congestion leading to a decrease in the throughput of the network.

In this research study, we have focused on enhancing the airborne networking architecture and protocols by the inclusion of Cross Layer Design (CLD) approach to optimize the performance of AN, see our published work in reference [7] and included in the Appendix. This new architecture approach will improve end-to-end message delivery ratio in a multi-hop scenario where links are very dynamic and resources are limited. Furthermore, it will provide ubiquitous and guaranteed network connectivity to all Air Force platforms across dynamic heterogeneous sub-networking, with Quality of Service (QoS) assurance in mind such as minimizing BER, delay and delay variation among the aircrafts in the AN.

We have performed detailed experimentation on single physical link at the Air Force lab in Rome New York to study the physical impact between two nodes in terms of node's speed, Doppler Effect and Fading. We have studied the impact of the physical parameters such as Doppler Effect, multi-path fading, mobility, limited bandwidth and limited transmission power on the performance of the Mobile ad hoc network (MANET) protocols. We designed OPNET simulator to analyze the impact of physical propagation environment on the performance of MANET protocols. Dynamic Source Routing (DSR) protocol is chosen as typical routing protocol for the study.

We used in our model OPNET simulator to build up our scenario. The network consists of 8 nodes distributed in a 1000 meter by 1000 meter area. Each node, transmitting at 30 mW, generates traffic of 1 kbps with an average packet size of 1024 bits. Random Waypoint Mobility Model (RWMM) is used to model the node's mobility with minimum speed of 0 m/s and maximum speed of 1, 3, 50, 150 mile/hr. See figure 1.

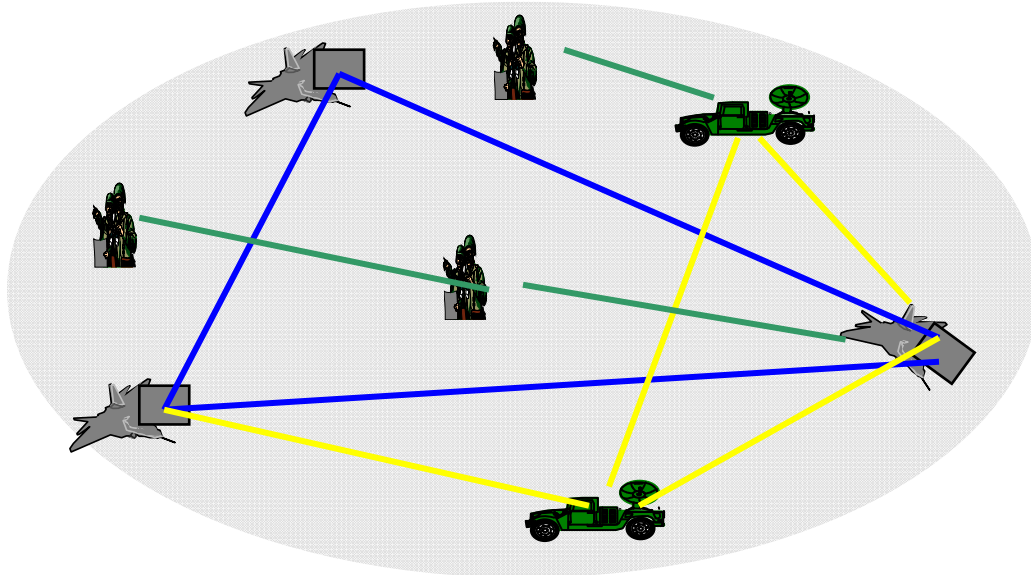


Figure 1. our scenario shows typical AN model.

The simulation parameters used in this simulation are listed in table 1.

Table 1- Simulation parameters

Simulator	OPNET
MAC Protocol	IEEE 802.11
Network Area	1Km x 1Km
Number of Nodes	8
Simulation Time (sec)	1800
Data Streams	Exponential
Inter arrival Packet time (sec)	1
Packet Size (bits)	1024
Transmitting Power (mW)	30
Reception Power Threshold (dBm)	- 95.0
Mobility Model	Random Waypoint Mobility Model (RWMM)
Maximum speed of nodes (mph)	1,3, 50, 150
Minimum speed of nodes (m/s)	0
Fading Type	Rician

In our Cross Layer Design, we used the MANET based routing protocols to route the generated traffic as phase I of this research. We have chosen DSR as a routing protocol in our model. In phase II, we will build our model using DTN based routing protocols so we can compare the performances between the two approaches. Figure 2 illustrates the obtained results in our scenario using the DSR routing protocol using OPNET simulator.

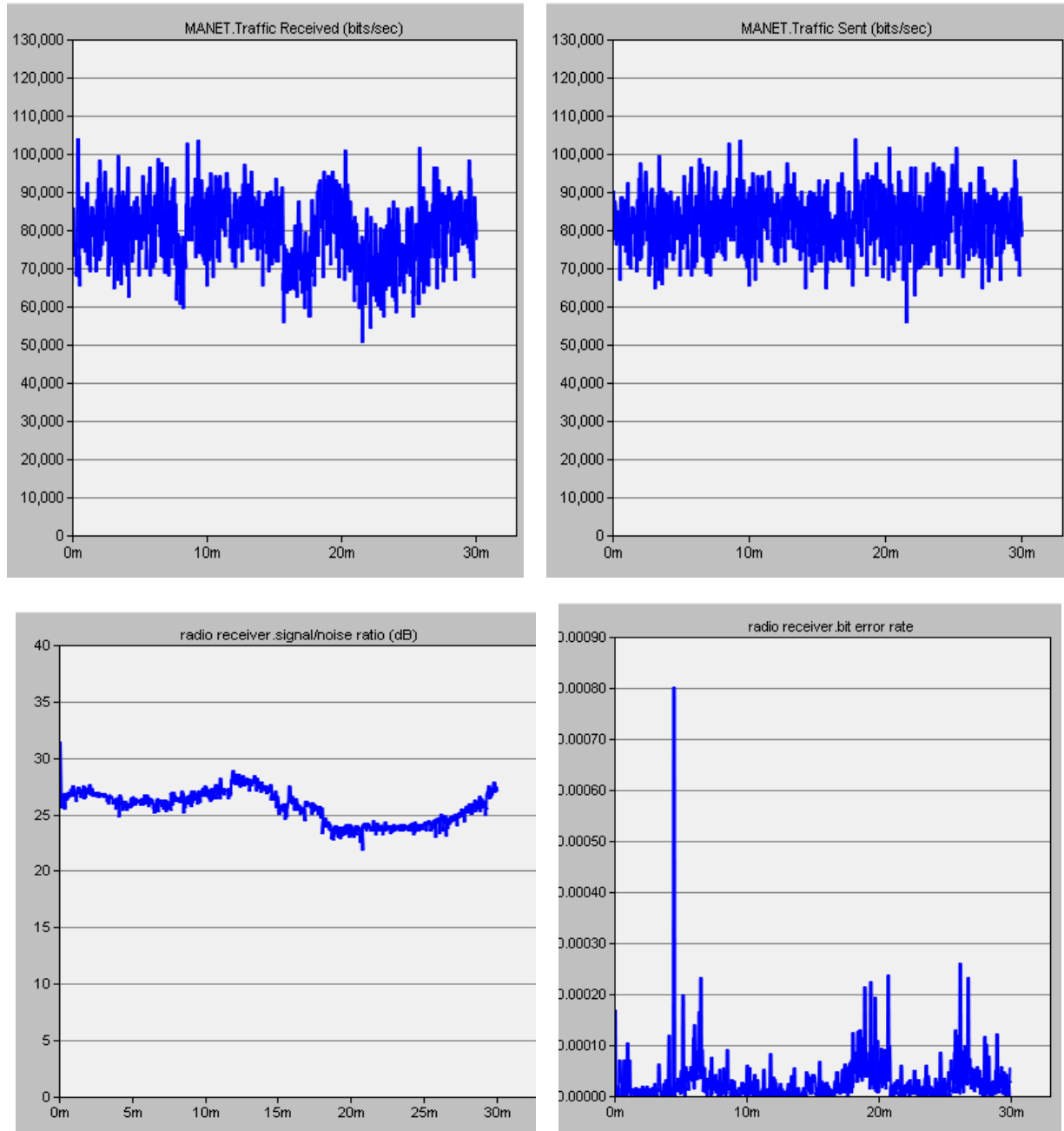


Figure 2. Simulation results using DSR routing protocol.

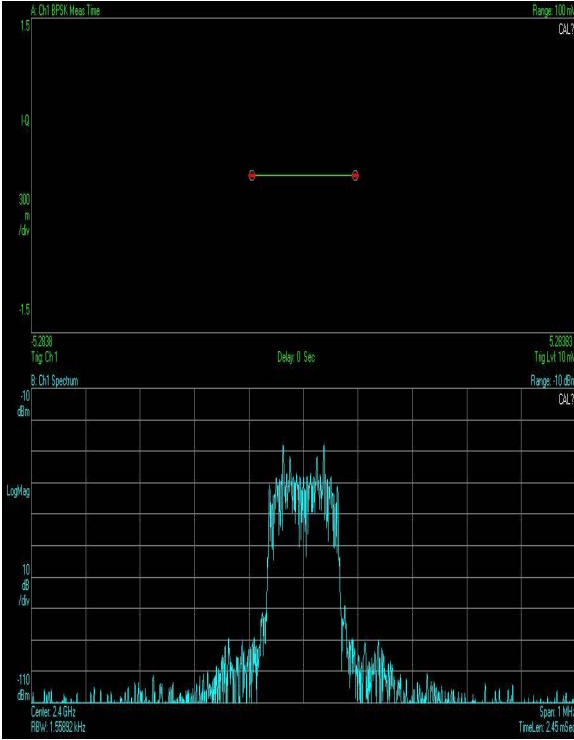
Figure 2 shows the traffic sent/ received (bps), S/N ratio and Bit Error Rate using Rician fading with speed of nodes (0-150 mph). By increasing nodes mobility the BER is increasing.

We designed our model to contain the actual Air Force environment including airplanes' speed, encounter time window between planes, Doppler Effect, multi-path fading, etc. In our experiment at the Air Force lab (Rome, New York) we studied the impact of the physical propagation parameters on single physical link between two nodes in terms of node's speed, Doppler Effect and multi-path fading. We used in the experiment the below parameters in Table2.

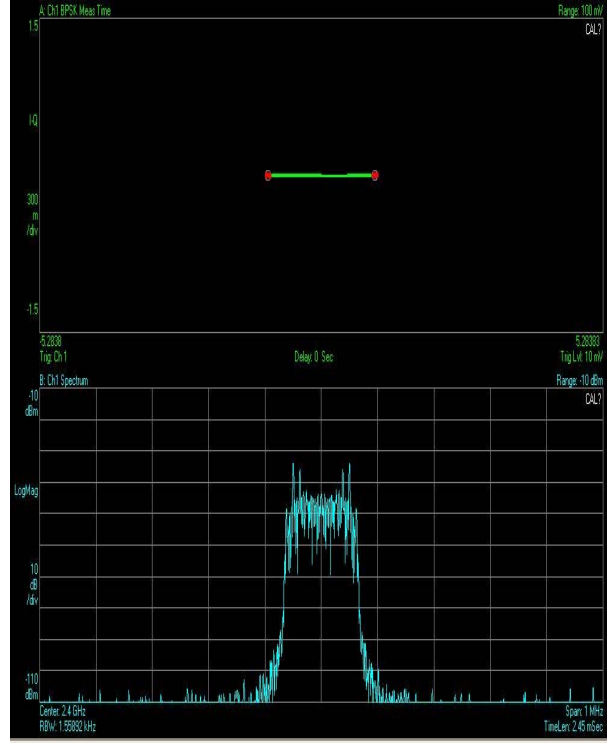
Table 2- experiment parameters

Parameter	
Node speeds (mph)	0, 3, 50, 150
Carrier Frequencies (GHz)	2.4
Fading Type	Rician
Modulation	BPSK
In Band RF Power (dBm)	- 20

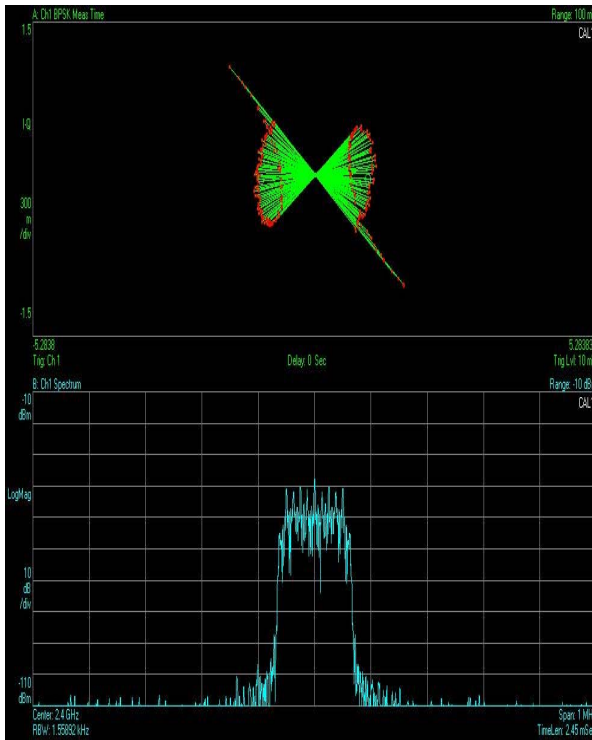
Figure 3 shows the experimental results. In this experiment we changed the node's mobility from 0 to 150 mph to obtain different values of Doppler Effect. We used one path Rician fading. The results show that the communication between the two nodes destroyed after reaching 20 mph considering one path Rician fading.



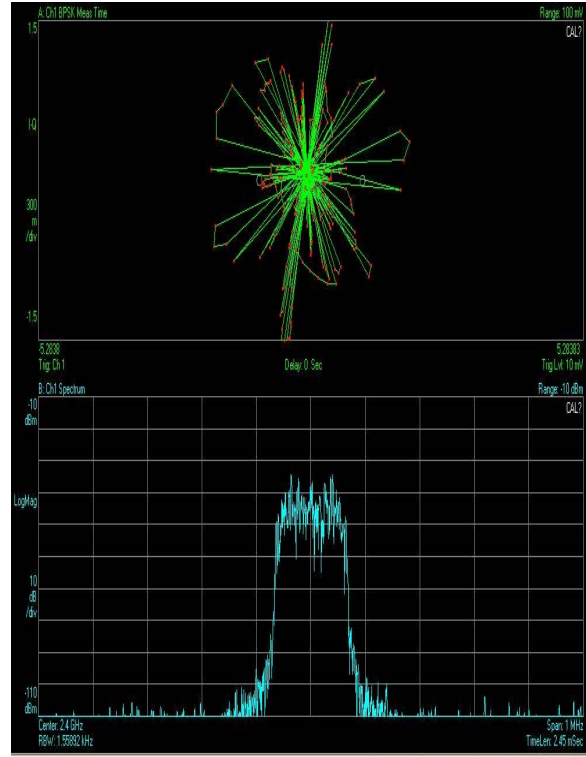
a) Speed: 0 mph (fixed node), No fading



b) Speed: 3 mph, Rician Fading



c) Speed: 50 mph, Rician Fading



d) Speed: 1500 mph, Rician Fading

Figure 3: Top figure: I/Q graph (Amplitude voltage vs. Quadrature voltage)
Bottom figure: power – frequency graph

3.0 Conclusion

We analyzed the impact of the physical propagation environment on the performance of ad hoc network protocols. Dynamic Source Routing (DSR) protocol is chosen as typical routing protocol for the experiment. We used OPNET simulator to design our AN environment model. Our goal is utilized the experimental results that obtained at the AirForce Lab in our design to model a Cross Layer Design to reduce the impact of the physical propagation parameters in An environment, see our published work reference [7] and attached in the Appendix. We evaluated the performance of the MANET-based protocol under Air Force environment. We designed a DTN-based probabilistic routing protocol to work in AN environment that is associated with intermittent connectivity. The performance of the MANET-based protocols under Air Force environment will be compared with the DTN-based routing protocols to show the advantage of DTN architecture.

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- [6] DTNRG website (www.dtnrg.org).
- [7] A. Abdelal, T.Saadawi, “High Throughput Path Selection Mechanism for Multi-Path Video Streaming in Ad-Hoc Networks,” IEEE Symposium on Computers and Communications, ISCC 2008, Morocco, July 2008.

5.0 List of Acronyms

AN – Airborne Networking
CLD – Cross Layer Design
QoS – Quality of Service
DLL – Data Link Layer
MAC – Medium Access Control
OSI – Open System Interconnect
TCP/IP – Transport Control Protocol / Internet Protocol
DTN – Delay-Tolerant Network
MANET – Mobile Ad Hoc Network
GIG – Global Information Grid
CoS – Class of Service
BP – Bundle Protocol
DSR – Dynamic Source Routing

APPENDIX

Published Papers

- A. Abdelal, T.Saadawi, "High Throughput Path Selection Mechanism for Multi-Path Video Streaming in Ad-Hoc Networks," IEEE Symposium on Computers and Communications, ISCC 2008, Morocco, July 2008.

High Throughput Path Selection For Multi-Path Video Streaming in Ad-Hoc Networks

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Abstract— *Owing to the absence of any static structure, Ad-hoc networks are prone to packet losses and link failures. Selecting the shortest path may not lead to high quality routes. The consequent high loss rates can lead to the degradation of throughput and increase in the end-to-end delay on the selected path, which can lead to devastating effects for the transport of compressed video. This paper suggests a selection mechanism for multi-path routing protocols that selects routes on the basis of the expected throughput of the candidate routes. We have studied the performance of the proposed mechanism by incorporating it into a multi-path routing protocol, and examining the quality of compressed video streams in this network. Simulation results show that the proposed selection mechanism maintains the video quality under different loss rates and mobility speeds. Also we have found that our mechanism significantly outperform shortest path selection.*

I. INTRODUCTION

With the ever increasing demand for connectivity, the need for mobile wireless communication is inevitable. The use of portable laptops and hand held devices is increasing rapidly. Most of the portable communication devices have the support of fixed base stations or access points and predesignated routers for communication purposes. However, such support is not available in settings where access to a wired infrastructure is not possible. Situations like natural disasters, conferences and military settings are noteworthy in this regard. This has led to the need for *Mobile Ad-hoc Networks* [1]. A mobile ad-hoc network is a dynamically changing network of mobile devices that communicate without the support of a fixed structure. There is a direct communication among the neighboring devices, but non-neighboring devices require a robust and intelligent routing strategy to ensure reliable and efficient communication. In the same time, the increase in the

bandwidth of wireless channels and the computing power of mobile devices increase the interest in video communications over ad-hoc wireless networks. However, in such networks there is no end-to-end guaranteed quality of service (QoS) and packets may be discarded due to bit errors. Wireless channels provide error rates that are typically around 10^{-2} , which range from single bit errors to burst errors or even intermittent loss of the connection. The high error rates are due to multi-path fading, which characterizes mobile radio channels, while the loss of the connection can be due to the mobility in such networks. In addition, designing the wireless communication system to mitigate these effects can be complicated by the rapidly changing quality of the radio channel.

The effect of the high error rates in ad-hoc networks can be devastating for the transport of compressed video. Video standards, such as MPEG [2] and H.263 [3], use motion-compensated predictive (MCP) coding to reduce the temporal and statistical redundancy between the video frames. Although MCP coding can achieve high compression efficiency, it is not designed for transmission over lossy channels [4].

Much of the recent work in video streaming in ad-hoc networks have focused on coping with the rapidly changing topologies, and the high packet loss rates by utilizing multiple paths between the sender and receiver [5, 6, 7]. Less attention has been paid in these mechanisms on the path selection criteria for selecting the paths for video streaming from the set of candidate paths returned by the multi-path routing protocols. The criteria most commonly used by these mechanisms are based on minimum hop-count. This paper presents selection criteria which accounts the expected throughput of the available paths by estimating the *Medium Transmission Time (MTT)* of these paths.

Minimizing the hop-count maximizes the distance traveled by each hop, which is likely to minimize signal strength and maximize the loss ratio. Even if the best route is a minimum hop-count route, in a dense network there may be many routes of the same minimum length, with widely varying qualities; the arbitrary choice made by most minimum hop-count path selection is not likely to select the best.

One approach to cope with wireless characteristics is to mask transmission errors. For example, the 802.11b ACK mechanism resends lost packets, making all but the worst

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802.11b links appear loss-free. However, retransmission does not make lossy links desirable for use in paths: the retransmissions reduce path throughput and interfere with other traffic. Another approach might be to augment minimum hop-count routing with a threshold that ignores lossy links, but a lossy link may be the only way to reach a certain node, and there might be significant loss ratio differences even among the above threshold selected links. The solution proposed and evaluated in this paper is the *MTT* metric. *MTT* finds paths with the lowest expected transmissions delays, due to transmissions and retransmissions, required to deliver a packet all the way to its destination. The metric predicts the time required for transmitting a packet using per-link measurements of packet loss ratios in both directions of each wireless link, as well as the bandwidth of the channel. The primary goal of the *MTT* design is to find paths with lowest transmission delays, thus the highest throughput, despite losses.

In order to demonstrate that *MTT* is effective, we implemented the routes selection criteria in a simulation model for a multi-path routing protocol, namely Split Multipath Routing (SMR) [8], and we compared the performance of using *MTT* to select routes for video streaming to that provided with the minimum hop-count metric. This paper makes the following main contributions. First, it presents the design, implementation, and evaluation of the *MTT* path selection criteria. Second, it describes a set of detailed design changes to multi-path routing protocols (e.g., SMR) to which *MTT* is an extension, that enable them to more accurately choose routes with the best metric.

This paper is organized as follows. Section II provides a review for related works. The proposed mechanism is presented in detail in Section III. We present experimental results and performance evaluation in Section IV. Finally, conclusions and future work are outlined in Section V.

II. RELATED WORK

The problem of devising a link-quality metric for static 802.11 ad-hoc networks has been studied previously. Most notably, De Couto et al. [9] propose a mechanism that selects paths which minimize the expected number of retransmissions and compare its performance to number of hops using DSDV and DSR with a small datagram workload. Their study differs from ours in many aspects. They only study the throughput of single, short (30 seconds) data transfers using small datagrams. Also, their experiments include no mobility. In contrast, we study video streaming where throughput and latency are important aspects for ensuring high video quality. Finally, our work includes scenarios with mobility. Woo et al. [10] examines the interaction of link quality and ad-hoc routing for sensor networks. Their scheme is based on passive observation of packet reception probability. Using this probability, they compare several routing protocols including shortest-path routing with thresholding to eliminate links with poor quality. Signal-to-noise ratio (SNR), has been used as a link quality metric in several routing schemes for mobile ad-hoc

networks. For example, in [11] the authors use an SNR threshold value to filter links discovered by DSR Route Discovery. The main problem with these schemes is that they may end up excluding links that are necessary to maintain connectivity. Another approach is used in [12], where links are still classified as good and bad based on a threshold value, but a path is permitted to use poor-quality links to maintain connectivity. Punnose et al. [13] also use signal strength as a link quality metric. They convert the predicted signal strength into a link quality factor, which is used to assign weights to the links. Awerbuch et al. [14] study the impact of automatic rate selection on performance of ad-hoc networks. They propose a routing algorithm that selects a path with minimum transmission time. Their metric does not take into account the packet loss.

In [15] Draves et al. conduct a detailed, empirical evaluation of the performance of different link-quality metrics and compare them against minimum hop count. Draves et al. [16] present a routing metric for multi-channel multi-hop static wireless networks.

Several researchers have utilized the path diversity for video streaming in ad-hoc networks. Apostolopoulos [7] propose to code a video source into multiple descriptions, using temporal frame sub-sampling, and to transmit them over multiple paths through either IP source routing or relay service. Gogate et al. [17] examine the effectiveness of combining Multiple Description Coding (MDC) and multi-path transport (MPT) for video and image transmission in a multi-hop mobile radio network. Shunan et al. [5] propose a scheme for reliable transmission of video over bandwidth limited ad-hoc networks. A raw video stream is layer coded and the base-layer and the enhancement-layer packets are transmitted separately on two disjoint paths. The base layer packets are given higher protection than enhancement layer packets. Mao et al. [6] compare the error-resilience capabilities of MDC and Layered Coding (LC) in case of multi-path transport under different path environments. Again their approach depends on the quality of the path selected to transport the base layer. Abdelal et al. [18] propose unequal error protection scheme for wireless video through multiple paths redundant retransmissions. Begen et al. [19] study how to select multiple path so as to maximize the average video quality at clients of an Internet overlay networks. Wei and Zakhori [20] propose a multi-path selection framework for streaming over wireless ad-hoc networks, while Mao et al. [21] propose a metaheuristic approach based on genetic algorithms to solve the above path selection problem in wireless ad-hoc networks. However these approaches are too complex to be performed in real-time.

III. PROPOSED SOLUTION

A. Computing Path Selection Criteria

Much prior research has recognized the shortcomings of shortest-path routing in multi-hop wireless networks. In this section, we will focus on the *MTT* (Medium Transmission

Time) route selection criteria. The *MTT* measures the total transmission time. The inverse of the *MTT* of a link is proportional to the capacity of that link. Similarly, the inverse for the path *MTT* approximately equals the end-to-end path capacity. Therefore, by selecting paths based on using *MTT* we are selecting paths that minimizes the usage of the shared medium and maximizes its end-to-end path capacity.

The *MTT* metric measures the medium transmission time, including retransmissions, needed to send a unicast packet across a link. The derivation of *MTT* starts with measurements of the underlying packet loss probability in both the forward and reverse directions; denoted by *pf* and *pr*, respectively; and then calculates the expected number of transmissions. We begin by calculating the probability that a packet transmission is *not* successful. The 802.11 protocol requires that for a transmission to be successful, the packet must be successfully acknowledged. Let *p* denote the probability that the packet transmission from *x* to *y* is *not* successful:

$$p = 1 - (1 - pf) * (1 - pr) \quad (1)$$

The 802.11 MAC will retransmit a packet whose transmission was not successful. Let the probability that the packet will be successfully delivered from *x* to *y* after *k* attempts be denoted by *s(k)*. Then:

$$s(k) = p^{k-1} * (1 - p) \quad (2)$$

Finally, the expected number of transmissions required to successfully deliver a packet from *x* to *y* (link *l*) is denoted by *ETX(l)*:

$$ETX(l) = \sum_{k=1}^{\infty} k * s(k) = \frac{1}{1 - p} \quad (3)$$

We define the *MTT* of a link as a “bandwidth-adjusted *ETX*”. In other words, we start with the *ETX* (number of expected transmissions) and multiply by the link bandwidth to obtain the time spent in transmitting the packet. We can formalize this as follows. Let *N* denote the size of the packet (for example, 400 Bytes) and *B* the bandwidth (raw data rate) of the link. Then:

$$MTT(l, N) = overhead(l) + ETX(l) * \frac{N}{B} \quad (4)$$

The fixed overhead term *overhead(l)* is calculated according to the specifications in the wireless standard and specification of optional manufacturer provided features such as fast framing and packet bursting. It may include RTS, CTS, ACK, preamble, contention time, and any other sources of fixed overhead. A routing protocol should be able to query its wireless card's configuration parameters programmatically.

Note that this definition of *MTT* does not incorporate backoff time spent waiting for the radio channel; it only reflects the time spent actually using the channel. Thus it represents the lower bound of the expected transmission delay. More complex techniques can be used to capture the

backoff time [16], however we choose to use an approximation model to simplify computation at intermediate nodes.

The path metric is the sum of the *MTT* values for each link along the path. To compute the *MTT* for a complete path Π_{ij} between the source *i* and destination *j*, the *MTT* is defined as:

$$MTT(\Pi_{ij}, N) = \sum_{\forall l \in \Pi_{ij}} MTT(l, N) \quad (5)$$

The routing protocol selects the path with the minimum path *MTT*.

B. Multi-path Routing Implementation

We have implemented *MTT* path selection criteria in Split Multi-path Routing (SMR) protocol [8]. SMR is an on-demand routing protocol that builds multiple disjoint routes using request/reply cycles.

When the source has data packets to send but does not have the route information to the destination, it transmits a RREQ packet. The packet contains the source ID and a sequence number that uniquely identify the packet. When a node other than the destination receives a RREQ that is not a duplicate, it appends its ID and re-broadcasts the packet. In order to avoid this overlapped route problem, SMR introduces a different packet forwarding approach. Instead of dropping every duplicate RREQs, intermediate nodes forward the duplicate packets that traversed through a different incoming link than the link from which the first RREQ is received, and whose hop count is not larger than that of the first received RREQ. The destination node selects multiple disjoint routes and sends ROUTE REPLY (RREP) packets back to the source via the chosen routes.

In SMR, when a node fails to deliver the data packet to the next hop of the route, it considers the link to be disconnected and sends a ROUTE ERROR (RERR) packet to the upstream direction of the route. The RERR message contains the route to the source, and the immediate upstream and downstream nodes of the broken link. Upon receiving this RERR packet, the source removes every entry in its route table that uses the broken link (regardless of the destination). If only one of the two routes of the session is invalidated, the source uses the remaining valid route to deliver data packets. The source may select to initiate route recovery when either one of the paths or both paths fail. SMR selects the shortest paths from the returned disjoint paths.

To use the *MTT* selection criteria in SMR, the implementation was modified in a few simple ways. First, link probes are sent periodically to measure delivery ratios. Second, when a node forwards a request, it appends not only its own address, but also the metric for the link over which it received the request. These metrics are included in the route replies sent back to the sender. Third, we have added to the sources a path refreshment mechanism that periodically sends RREFRESH messages across all the known paths, to update their *MTTs*. The source can change the transmission path based on the updated *MTTs*.

To measure the link bandwidth, we depend on getting the bandwidth information from the 802.11 card auto-rate algorithm. Another ways to measure the bandwidth is through packet dispersion [22], however this will require regular exchange of two packets between the nodes, which will increase the protocol transmission overhead.

IV. PERFORMANCE ANALYSIS

To examine the *MTT* and compare it to minimum hop path selection, we have built a detailed OPNET simulations [23] model, to provide a more realistic view of the impact lower layers, including mobility, multi-path routing, multi-hop routes, and the MAC layer on the system performance. To represent link losses we have used a three-state Markov model [24] for each link with the states representing a “good,” “bad” or “down” status for the link. The “down” state means the link is totally unavailable. The “good” state has a lower packet loss rate than the “bad” state. The packet loss rates we used are, $p_0 = 1.0$, $p_1 \in [0.1\%, 25\%]$, and $p_2 = 0$, and, for the “down,” the “bad,” and the “good” states, respectively. The transition parameters are chosen to generate loss traces with desired loss rates and mean burst lengths.

Delay for link i is modeled by an exponential distribution with the mean delay $D_i = 15$ ms. We set the path maximum transfer unit (MTU) of 400 bytes for all the paths.

We have simulated an ad-hoc network with 16 nodes in a 600 m x 600 m region. Each node is randomly placed in the region initially. We used a version of the popular *random waypoint* mobility model, where each node first chooses a random destination in the region, then moves toward it at a *constant* speed. When it reaches the destination, it pauses for a constant time interval, chooses another destination randomly, and then moves toward the new destination. We used a pause time of 1.0 s for all the experiments reported in this paper. The speed of the nodes varies from 0 m/s to 10 m/s, which models movement of pedestrians or vehicles in city streets.

We use the IEEE 802.11 protocol in the MAC layer working in the DCF mode. Its physical layer features, e.g., frequency hopping (FH), are not modeled. The channel has a bandwidth of 1 Mb/s. The transmission range is 250 m. If the sender of a packet is within this range of the receiver, and the sender has successfully accessed the channel during the transmission period, the packet is regarded as correctly received. The maximum number of link layer retransmissions is seven, after which the packet is dropped.

Among the 16 nodes, one is randomly chosen as the video source and another node is chosen as the video sink, where a 250 ms playback buffer is used to absorb the jitter in received packets. The video source starts a session using two routes, sending encoded video at 200 kb/s to the sink. All other nodes generate background traffic to send to a randomly chosen destination. The interarrival time of the background packets is exponentially distributed with a

mean of 0.2 s. The background packet has a constant length of 512 bits.

To generate the video sequence used in our simulation, we used open source XviD MPEG-4 compliant video codec [25]. Sixty seconds of a high motion video sequence (football match) are encoded at 15 frames per second (fps), which results in a sequence of 900 frames. The frame resolution is Quarter Common Intermediate Format (QCIF, 176 x 144 pixels/frame), which is the most common format at low bit rates, and the coding rate is 200 Kb/s. We repeated our experiments with limited motion video sequence and we got similar results to that shown here. We limited the playout delay at the receiver to 100 ms, to represent an interactive video application.

The average *peak-signal-to-noise ratio* (PSNR) is used as a distortion measure of objective quality. PSNR is an indicator of picture quality that is derived from the *root mean squared error* (RMSE). The PSNR for a degraded $N_1 \times N_2$ image f' with respect to the original image f is computed as follows:

$$PSNR = 20 \log_{10} \frac{255}{\left(\frac{1}{N_1 N_2} \sum_{x=0}^{N_1-1} \sum_{y=0}^{N_2-1} [f(x, y) - f'(x, y)]^2 \right)^{1/2}} \quad (6)$$

Without transmission losses, the average PSNR of the decoded frames for the video sequence used in our performance study is 27 dB. After obtaining a transmission trace of a video sequence, we run the decoder on the trace to measure the image distortion due to packet losses, using the PSNR. In order to generate statistically meaningful quality measures, for each simulation scenario we repeated the experiment three times with different random seeds. The presented PSNR values are the average of the three experiments.

In all experiments, we selected the packet size, N in (4), equal 400 Bytes.

A. Comparison between *MTT* and Shortest Path Route Selection

In this section, we compare *MTT* against shortest path selection scheme. Fig. 1 shows the PSNR for each frame in the video sequence. We have set the nodes speed to 2 m/sec. As can be shown from the figure that *MTT* is able to maintain the video quality, than the shortest path selection. As explained before, shortest path selection which minimizes the hop-count maximizes the distance traveled by each hop, which is likely to minimize signal strength and maximize the loss ratio. This packet losses of video packets is magnified by the error propagation in compressed

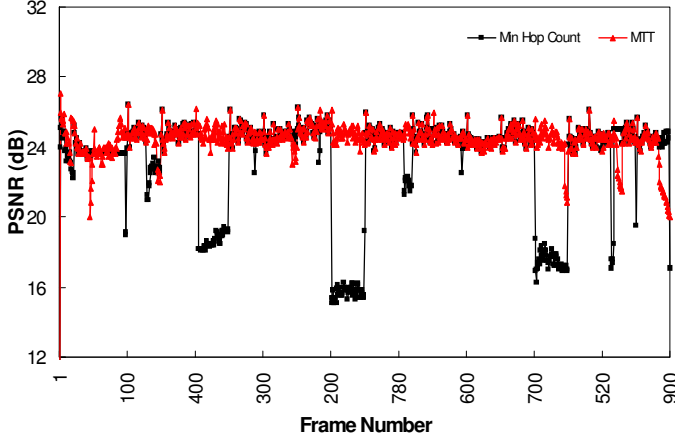


Fig. 1. PSNRs of received frames.

video. This is obvious in the long drop of the PSNR of the decoded video sequence. On the other hand, *MTT* selects links with the minimum losses (total transmission time), and maximum capacity. This has led to stable quality of the decoded sequence.

B. Effect of Packet Loss Rate

Fig. 2 shows the average PSNR of the received frames as we change the packet loss rate of the links. The nodes speed is set as 2 m/s. We examined minimum hop count, as well as the *MTT* path selection schemes. The figure shows that minimum hop count fails to maintain the video quality under high packet loss rate. On the other, the *MTT* maintains the video quality under different packet loss rates. Under low loss rates, the performance of minimum hop count is close to *MTT*. However, with high packet loss rate, the *MTT* maintains the video quality even with long error bursts.

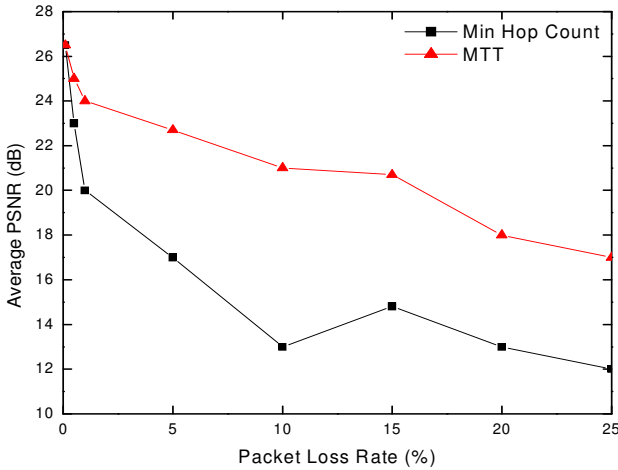


Fig. 2. Average PSNR vs. Packet Loss Rate.

C. Effect of Nodes Mobility Speed

We examined the impact of node mobility on the video with *MTT* and shortest path selection criteria. Fig. 3 shows the PSNR of the received video when the nodes are moving with speed 10m/s, and using shortest path selection. The PSNR curve shows large drops.

In addition the figure shows the PSNR with paths selected with *MTT*. Again the nodes are moving with speed 10 m/s. The PSNR curve is very stable, with only a few narrow drops. The figures show that *MTT* is able to maintain the visual quality, while the quality dropped by ~ 16.5 dB with shortest path selection.

Fig. 4 is the resulting average PSNR for different speeds using the *MTT*. The figure shows that during the initial increase in mobility, routes break down more easily, which leads to an increase in the mean packet loss rate and a drop in the PSNR. As speed further increases, the average PSNR becomes stable, as new topologies are more quickly formed and new routes are more quickly established. As speed increases, the period of time a node remains disconnected is smaller. The turning point (4 m/s in Fig 4) is determined by the node density in the region and the transmission range. We have found that similar phenomenon exists for other scenarios with a different number of nodes or a different transmission range, given that the node density is high enough to maintain a connected network for most of the time [18]. When the nodal speed increases even further, the routing process would be unable to track the quickly changing topology. Therefore, drops in the average PSNR are expected. The figure also shows that *MTT* keeps the PSNR for the received video higher than for the minimum hop count paths at different mobility speeds.

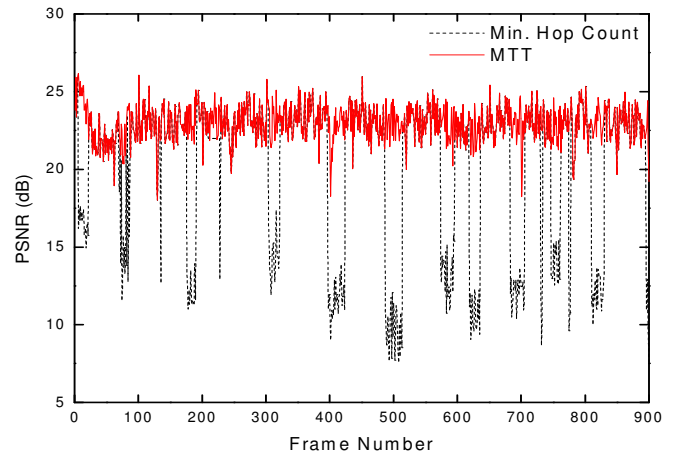


Fig. 3. Comparison between Min. Hop count and *MTT*. 16 nodes in 600 m x 600 m region at a speed of 10 m/s.

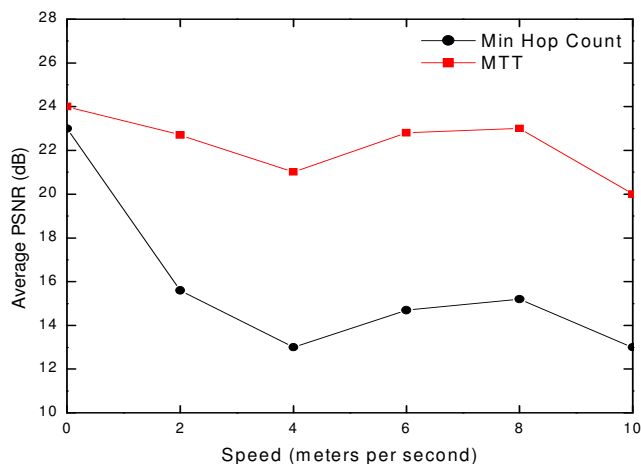


Fig. 4. Average PSNR versus node speed.

V. CONCLUSION

In this paper we have shown that the minimum hop metric tends to select paths with long slow links. As a result, these paths have low effective throughput and increase total network congestion. In addition, these paths are likely to contain long links that result in low reliability.

We have presented the Medium Transmission Time (*MTT*), an improved technique for route selection in ad-hoc wireless networks. The *MTT* is proportional to the time the node takes to transmit a packet on a given link. This metric selects paths that have the highest effective capacity. To evaluate *MTT* path selection and to compare it with other path selection criteria, we have implemented *MTT* as an extension to Split Multi-path Routing (SMR) multi-path routing protocol. We have also examined the effectiveness of *MTT* path selection for video streaming in ad-hoc networks. Our simulation with 802.11b radios show that the *MTT* achieves significantly higher throughput than alternative metrics. We observed better video quality with *MTT* than selecting the transmission path based on minimum hop count. Our results both demonstrate the importance of using medium time for selecting high throughput routing paths, and underscore the need for inter-layer communication in order to efficiently and accurately estimate the medium time.

As a future work, we plan to implement and examine *MTT* based path selection in an ad-hoc network testbed. Also we plan to investigate cross-layer schemes for optimizing the path selection through information obtained from lower layers regarding the path conditions.

DISCLAIMER

The views and conclusions in this document are those of the authors and should not be interpreted as representing the official policies, either expressed or implied, of the Army Research Laboratory, the Air Force Research Laboratory, or the U.S. Government.

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