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13. ABSTRACT (Maximum 13 lines, use the attachment if necessary)
The DAWN (Dynamic Ad-hoc Wireless Networks) project is developing a general theory of complex and dynamic wireless communication networks. To accomplish this, DAWN adopts a very different approach than those followed in the past and summarized above. DAWN constitutes what is arguably one of the most ambitious research teams of experts ever assembled that can mount a truly cross-disciplinary approach incorporating the effects of the physical layer explicitly into the modeling, analysis, and control of wireless communication networks. DAWN will systematically redefine and reorganize existing models, protocols and controls, and develop new ones in a framework that guarantees realism and cross-layer consistency to enable the efficient design of such complex wireless systems as those required by the Army. The models, tools, algorithms, and protocols developed in DAWN will provide transformational improvements to the way in which a network-centric battlefield is managed in the future.

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wireless networks, tactical networks, mobile ad hoc networks, network coding, multi-packet reception, MIMO, communication protocols, network capacity, performance analysis, network science, information theory, simulation, analytical models, routing, multicasting, channel access, disruption-tolerant networks

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1 Introduction

The network-centric battlefield is a system of systems encompassing large numbers of sensors and mobile or static elements. Sensors, troop carriers, unmanned air vehicle (UAV), aircraft, smart weapons, soldiers, and command centers are all interconnected in the battlefield and can be reached from wired segments of the Internet. The network-centric battlefield is subject to drastic and continuous changes in radio-link quality, connectivity, the location of nodes and services, and the types of information flows that traverse communication links and nodes.

In theory, ad hoc networking technology should be ideal to support the network-centric battlefield; however, the practice today is far from this theory. Despite the intense past and ongoing research and development work on ad hoc wireless networks, adequate principles, tools , and methodologies for designing reliable MANETs that meet given requirements are still missing. To date, most protocol design and analysis approaches for MANETs have essentially decoupled the "network" from the "physical medium." The results from this approach have led to the false belief that system theoretic tools (that focus for example on dynamical system behavior) may lead to scalable methods of design and operation. The focus of most protocol-design approaches for MANETs and sensor networks continue to view a network as a graph in which interference at a receiver occurs only over the links with its "one hop" neighbors (those nodes whose transmissions the receiver can decode on the basis of signal attenuation), which leads to the erroneous model that interference is only due to nodes in close proximity to the receiver. These approaches apply methods of analysis that are similar to those used to study the Internet. In reality, interference at a receiver is a complex function of the characteristics of the wireless media and terrain, and the transmissions allowed by the protocol stack from any source in the network. On the other extreme, communication theorists either continue to focus on point-to-point models (or at most on "uplink" or "downlink" models of cellular paradigms) or pose Shannon-theoretic questions for large networks that are as yet impossible to analyze, and provide insight into the design of sparse networks or networks of moderate sizes.

To address the limitations of the current state of the art in the modeling of ad hoc networks and their protocols, the *Dynamic Ad-hoc Wireless Networks (DAWN)* project was established to develop a general theory of complex and dynamic wireless communication networks. To accomplish this, DAWN adopts a very different approach than those followed in the past and summarized above. DAWN constitutes what is arguably one of the most ambitious research teams of experts ever assembled that can mount a truly cross-disciplinary approach incorporating the effects of the physical layer explicitly into the modeling, analysis, and control of wireless communication networks. DAWN will systematically redefine and reorganize existing models, protocols and controls, and develop new ones in a framework that guarantees realism and cross-layer consistency to enable the efficient design of such complex wireless systems as those required by the U.S. Army. The models, tools, algorithms, and protocols developed in DAWN will provide transformational improvements to the way in which a network-centric battlefield is managed in the future.

DAWN is meant to operate as a distributed laboratory with investigators in multiple campuses: (a) working jointly in research and student supervision, (b) ensuring that our research has relevance to the Army and the DoD, and (c) fostering cross-fertilization with other programs. Professor Garcia-Luna-Aceves is the Principal Investigator (PI) of DAWN. The rest of the faculty supported in DAWN are Professor Anthony Ephremides from the University of Maryland (Maryland); Professor Muriel Medard from MIT; Professor Pravin Varaiya from UC Berkeley; Professors Mario Gerla and Rajive Bagrodia from UCLA; Professors Katia Obraczka and Hamid Sadjadpour from UC Santa Cruz (UCSC); Professor Nitin Vaidya from the University of Illinois at Urbana-Champaign (UIUC); and Professor Andrea Goldsmith from Stanford University (Stanford).

2 Scientific Progress

This section summarizes the scientific progress made during the second year of the project in establishing a mathematical framework for the modeling of MANETs and sensor networks, exploiting cross-layer interactions in protocol stacks, developing a theory of scalable and robust protocols for MANETs and sensor networks, enabling large and high-fidelity simulations, and introducing novel approaches for network control.

We organize the discussion of our contributions according to the major research areas addressed in the project, rather than individual contributions by each principal investigator (PI). The PIs in DAWN have collaborated across multiple tasks and many of the technical accomplishments in DAWN involve more than a single task and a single PI.

2.1 Executive Summary

During its third year, the DAWN project has continued making fundamental contributions to the development of a science of networks. The caliber of the contributions made in the project can be easily measured by the quality and quantity of publications, as well as the production of the graduate students who have benefited from the research carried out in DAWN.

We have made remarkable progress in the development of new mathematical frameworks for the characterization of MANETs, which will help researchers and practitioners understand mobile ad hoc networks, and the design of protocols that are much better suited for MANETs. During the third year of the DAWN project, we have had 20 journal papers and book chapters published or accepted for publication, 71 peer-reviewed papers published or accepted for publication in conference proceedings and books, and more than 12 manuscripts being considered for publication. The quality of the work is as impressive as the number of publications achieved by the researchers participating in DAWN. It is worth noting that the DAWN research team has received Best Paper awards in each of the past three years. The following three papers received Best Paper awards during the past year:

- Best Paper Award (out of 250 submissions):
  R. Menchaca-Mendez and J.J. Garcia-Luna-Aceves, "Scalable Multicast Routing in MANETs Using Sender-Initiated Multicast Meshes," *Proc. IEEE MASS 2008: Fifth IEEE International Conference on Mobile Ad-hoc and Sensor Systems*, September 29–October 2, 2008, Atlanta, Georgia.

- Best paper award:
  D. ONeill, A. Goldsmith, and S. Boyd, "Optimizing Adaptive Modulation in Wireless Networks via Utility Maximization," *Proc. IEEE ICC 2008*, Beijin, China, May 2008.

- Best Student Paper Award:
  Vartika Bhandari and Nitin H. Vaidya, "Capacity of Multi-Channel Wireless Networks with Random (c, f) Assignment, " *Proc. ACM MobiHoc 2007*, September 9-14, 2007, Montral, Canada.

Our journal publications appear in some of the most selective journals covering the field of wireless networks, including the IEEE Transactions on Information Theory, the IEEE Transactions on Communications, IEEE Transactions on Wireless Communication, IEEE JSAC, WINET, and Ad Hoc Networks. Many of the conference papers published over the past year by DAWN researchers appeared in such technical conferences as ACM MobiCom, and ACM MobiHoc, IEEE SECON, IEEE MASS, and IEEE Infocom, which are more selective than many technical journals. In addition, to these publications, several Ph.D. theses and M.S. theses were completed.

During the past year, we have obtained many new results on the capacity of wireless networks. Key results in this area include: (a) The computation of optimal unicast capacity and the technique to achieve this capacity; (b) showing that network coding (NC) does not provide order throughput gains in wireless ad hoc networks subject to multicast communication when combined with multi-packet reception (MPR) or multi-packet transmission (MPT) techniques; and (c) establishing a unifying framework for the capacity of wireless networks subject to any type of information dissemination modalities, and employing MPR or MPT. We have also developed the first modeling of topology evolution in wireless mobile ad hoc networks (MANETs), a unifying analytical framework for routing protocols in MANETs, more accurate performance models for 802.11 channel access, and novel results on the impact of network coding on the throughput and delay in wireless networks. We describe these results in the context of Task 1.

As part of Task 2, we continued to make progress on cross-layer design issues in wireless networks. During the past year, this aspect of our research has focused on a cross-layer perspective on multicasting in wireless ad hoc networks. Most previous work on random access assumes that a collision results in the loss of all transmitted packets. In contrast to the usual approach, we take a cross-layer perspective and allow for successful reception depending on the level of interference. We have investigated the use of network coding in obtaining the ultimate limits on performance, in terms of both throughput and delay, for multicast transmission.

As part of Task 3, we continued to investigate new theories of scalable and robust protocols with emphasis on channel access and routing. We have focused primarily on the theory of channel access and routing protocols. We investigated new medium access control (MAC) protocols for multi-channel networks and MAC protocols aimed at increased channel access concurrency by exploiting OFDMA or by exploiting space and temporal dimensions in the context of fading environments. We have studied several topics regarding the application of network coding (NC) to network control. In particular, we analyzed methods that reduce the overhead incurred in using network coding, two novel approaches (RelayCast and Probe-

Cast) for multicasting in wireless ad hoc networks subject to major disruption, analyzed the resource consumption incurred by using NC for content distribution in MANETs, and compared NC with erasure coding in the context of reliable multicasting in a wireless ad hoc network. We introduced and analyzed a new approach called MeDeHa for routing in networks subject to disruption. We introduced three different approaches aimed at attaining scale-free routing in MANETs, namely: interest-based signaling for unicast and multicast routing, which confines the majority of signaling overhead to those nodes that must connect sources with destinations; integrated addressing and routing, which establishes implicit routes when it assigns addresses to nodes and eliminates most of the flooding signaling requirements; and multi-dimensional routing, which provides many more alternate routes and therefore makes routing much more resilient to failures and mobility.

In Task 4, we focused on the development of a tool for rapid prototyping communication protocols over 802.11 networks, and a testbed for the various protocols and algorithms we have designed in DAWN. In Task 5, we focused on techniques and evaluation of platforms that improve the modeling and simulation and emulation of large-scale heterogeneous wireless networks. In Task 6, we addressed several aspects of information flow patterns in networks, with special focus on beam-forming, fault detection, and epidemic dissemination in sensor networks and vehicular networks. We examined the dissemination of patient medical records across a team of nurses connected via Bluetooth and the safe delivery of such information to the attending doctor and the central record repository. We study the delivery delay as a function of nurses' motion pattern. We also considered information dissemination in the urban grid of the environment data collected by vehicles (e.g., license plates, vehicle images, traffic congestion, pollution measurements etc); we further harvest the data of interest using "agents" (e.g., patrol cars). We proposed a bio-inspired multiple agent foraging motion model to effectively harvest the data. The agent motion model can be described as a continuous time Markov model where the agent moves from state to state with probabilities depending on the data found in a neighborhood. We showed the effectiveness of the "bio-harvest" motion model by comparing with traditional models such as uniform search and random way-point search.

The DAWN project has continued delivering tangible and fundamental results in the "science of networks" challenge identified by the U.S. Army Research Office. The rest of this section summarizes the technical accomplishments in each task of the project.

2.2 Task 1: Mathematical Modeling Frameworks

In this task, we have made several contributions on the modeling of ad hoc networks. These contributions include:

- The computation of optimal unicast capacity and the technique to achieve this capacity.
- Proving that network coding (NC) *does not* have any contribution in the scaling laws of wireless ad hoc networks in multicast communication when combined with multi-packet reception (MPR) or multi-packet transmission (MPT) techniques.
- Providing a unifying framework for the capacity of wireless networks subject to any type of information dissemination modalities, and employing MPR or MPT.

- The first modeling of topology evolution in wireless mobile ad hoc networks (MANETs).
- A unifying analytical framework for routing protocols in MANETs.
- More accurate performance models for 802.11 channel access.

The rest of this section provides more information about each of the above contributions.

2.2.1 Optimal Scaling Laws of Multicommodity Flows in Wireless Ad Hoc Networks

All prior work on the capacity of wireless networks has focused on what is attainable with specific approaches to handle multiple access interference (MAI). No prior work has focused on first establishing what is the optimal capacity of a wireless network in the absence of MAI, and then determining whether that capacity is attainable when MAI is present. Over the past year, we have been able to characterize the optimal interference-free capacity of a wireless network, and also showed how it can be attained in the presence of interference [96].

We model a *random network* with n nodes, a homogeneous communication range of $r(n)$, and unicast traffic for k source-destination (S-D) pairs. In the absence of interference, such a network corresponds to a *random geometric graph* (RGG) with an edge between any two nodes separated by a distance less than $r(n)$. We define a *combinatorial interference model* based on RGGs, and use it to express all the protocol models used in the past and a model that we use to show that the optimal capacity of a wireless network is indeed attainable. We introduce a protocol model in which nodes have the ability to decode correctly multiple packets transmitted concurrently from different nodes, and transmit concurrently multiple packets to different nodes. We refer to this as the multi-packet transmission and reception (MPTR) protocol model.

The task of concurrently maximizing the data-rate for k S-D pairs is an instance of the multi-commodity flow problem. Hence, the *maximum concurrent multi-commodity flow-rate* (MCF) in a RGG equals the interference free capacity (i.e., the optimal capacity) of the network. To derive upper bounds on the optimal network capacity, we use the fact that the MCF is less than the minimum capacity of a multi-commodity cut for any arbitrary graph. The max-flow min-cut theorem by Ford and Fulkerson [1] establishes that this bound is tight for a single commodity. However, in general, the min-cut does not provide a tight bound on the max-flow [2]. The bound is known to be tight only for special cases [3], and in general can exhibit a gap of $\Theta(\log n)$ [2]. We establish a tight max-flow min-cut theorem for RGGs for the first time, and show that $\Theta\left(n^2 r^3(n)/k\right)$ is a tight bound on the optimal capacity of a wireless network.

We have generalized prior results by Gupta and Kumar [4] and by Garcia-Luna-Aceves et al. [18], and proving that the optimal capacity of wireless networks is attainable in the presence of MAI. We utilize the max-flow min-cut theorem to deduce tight order bounds for the capacity of random networks under various interference models. We show that the per-commodity capacity, under the protocol model suggested in [4], exhibits a tight order bound of $\Theta(1/r(n)k)$. This result generalizes Gupta and Kumar's result to any $k \geq \Theta(n)$ S-D pairs. Similarly, we generalize Garcia-Luna-Aceves et al.'s analysis for the MPR protocol model

[18]. We show that, under the MPR model, the per-commodity capacity of the network scales as $\Theta(nr(n)/k)$, which means that it is bounded away from the optimal capacity by a factor of $\Theta\left(nr^2(n)\right)$.

We show that MPTR achieves the optimal capacity of $\Theta\left(n^2r^3(n)/k\right)$. Hence, MPTR provides a gain of $\Theta\left(nr^2(n)\right)$ over MPR and any previously reported feasible capacity. What is just as striking is that MPTR can achieve the dual objective of increasing capacity and decreasing the transmission range as n increases. This is in stark contrast to the commonly held view that the capacity of multihop wireless networks cannot increase as the number of nodes increases. Indeed, our results demonstrate that the capacity of ad-hoc networks can actually *increase* with n while the communication range tends to zero!

The results of this work were reported recently at the IEEE MASS 08 and Asilomar 08 conferences, and they have also been submitted to an IEEE journal for publication.

2.2.2 On the Scaling Laws of Network Coding in Wireless Ad Hoc Networks

The seminal work by Gupta and Kumar [4] has sparked a significant interest in investigating the fundamental capacity limits of wireless ad hoc networks. Several techniques [5, 6, 7] have been developed with the objective of improving the capacity of wireless ad hoc networks. Network coding (NC), which was originally proposed by Ahlswede et al. in [8], is one such technique. Unlike traditional store-and-forward routing, network coding scheme encodes the messages received at intermediate nodes, prior to forwarding them to subsequent next-hop neighbors. Ahlswede et al. [8] showed that network coding can achieve a multicast flow equal to the min-cut for a single source and under the assumptions of a directed graph. This and other works in network coding (NC) [9, 10] has motivated a large number of researchers to investigate the impact of NC in increasing the throughput capacity of wireless ad hoc networks. However, Liu et al. [11] recently showed that NC does not increase the order of the throughput capacity for multi-pair unicast traffic. Nevertheless, a number of efforts (analog network coding [12], physical network coding [13]) have continued the quest for improving the multicast capacity of ad-hoc networks by using NC. Despite the claims of throughput improvement by such studies, the impact of NC on the multicast scaling law has remained uncharacterized.

Promising approaches [12, 13] implicitly assume the combination of NC (transmitting multiple packets encoded in a single transmission) with Multi-packet Transmission (MPT) and Multi-packet Reception (MPR) [14, 15, 16] (i.e., the ability to transceive successfully multiple concurrent transmissions by employing physical-layer interference cancelation techniques). MPR has been shown to increase the capacity regions of ad hoc networks [17], and very recently Garcia-Luna-Aceves et al. [18] have shown that the order capacity in wireless ad hoc networks subject to multi-pair unicast traffic is increased with MPR. These prior efforts raise three important questions: (a) What is the multicast throughput order achieved by the combination of NC with MPT and MPR? (b) Does this combination provide us with an order gain over traditional techniques based on routing and point-to-point communication? (c) If yes, what exactly leads to this gain? Is NC necessary or does the combination of MPT and MPR suffice?

In our work, we addressed the above three questions. The answers can be summarized by our main results:

- When each multicast group consists of a constant number of sinks, the combination of NC, MPT and MPR provides a per session throughput capacity of $\Theta(nT^3(n))$, where $T(n)$ is the communication range.
- This scaling law represents an order gain of $\Theta(n^2T^4(n))$ over a combination of routing and point-to-point communication.
- The combination of only MPT and MPR is sufficient to achieve a per-session multicast throughput order of $\Theta(nT^3(n))$. Consequently, NC does not contribute to the multicast capacity when MPR and MPT are used in the network!

These results are crucial for the Army and the DoD in general. Our results basically show that NC may not necessarily be the right approach to improve the scaling laws of wireless ad hoc networks. We are currently investigating whether or not network coding along with simple point-to-point communications in wireless ad hoc networks provides any gain compared to the simple point-to-point communications. The results of this line of research will be reported next year.

2.2.3 Scaling laws of Information Dissemination Modalities with MPT and MPR

AS we reported last year, we have developed the first unified modeling framework for the computation of the throughput capacity of random wireless ad hoc networks in which information is disseminated by means of unicast routing, multicast routing, broadcasting, or different forms of anycasting. More specifically, we define (n, m, k)-casting as a generalization of all forms of one-to-one, one-to-many and many-to-many information dissemination in wireless networks. In the context of (n, m, k)-casting, n, m, and k denote the number of nodes in the network, the number of destinations for each communication group, and the actual number of communication-group members that receive information (i.e., $k \leq m$), respectively.

During the past year, we extended this modelling framework to account for the use of multi-packet reception (MPR) or multi-packet transmission (MPT). We have obtained the first results on the capacity of ad hoc networks with MPT under different forms of information dissemination, and the first results for the capacity of networks with MPR for dissemination modalities other than unicast traffic. We show that the per source-destination (n, m, k)-cast throughput capacity $C_{m,k}(n)$ of a wireless random ad hoc network with MPT or MPR is tight bounded (upper and lower bounds) by $\Theta\left(T(n)\sqrt{m}/k\right)$, $\Theta\left(1/k\right)$ and $\Theta\left(T^2(n)\right)$ w.h.p.[1] when $k \leq m \leq \Theta\left(T^{-2}(n)\right)$, $k \leq \Theta\left(T^{-2}(n)\right) \leq m$, and $\Theta\left(T^{-2}(n)\right) \leq k \leq m$, respectively. In these results, the transceiver range $T(n)$ in MPT or MPR is different from the transmission range $r(n)$ used in the capacity results for networks with point-to-point communication [**?**]. For comparison purposes, we also show the (n, m, k)-cast capacity result for point-to-point communication.

[1] An event occurs with high probability (w.h.p.) if its probability tends to one as n goes to infinity. Θ, Ω and O are the standard order bounds.

We have also characterized the capacity-delay tradeoff of wireless ad hoc networks with MPT, MPR, and point-to-point communication. We show that MPT and MPR have similar throughput capacity and delay behavior in wireless ad hoc networks. We modelled the behavior of the capacity of an ad hoc network with MPT, MPR, or point-to-point schemes as a function of the (n, m, k)-cast parameters and as a function of the transceiver range. For the minimum value of $T(n) = r(n) \geq \Theta\left(\sqrt{\log n/n}\right)$ for the transceiver range to guarantee network connectivity [**?**], the (n, m, k)-cast throughput capacity of MPT or MPR is shown to have a gain of $\Theta(\log n)$ compared to that of point-to-point communication. When $T(n) > \Theta\left(\sqrt{\log n/n}\right)$, the (n, m, k)-cast with MPT or MPR can achieve higher throughput capacity gains than that of point-to-point communication. Furthermore, the capacity-delay tradeoff for MPT or MPR is fundamentally different from point-to-point communications tradeoff.

2.2.4 Modeling Topology Evolutions and Implication on Proactive Routing Overhead

Mobility brings fundamental challenges to the design of protocol stacks for mobile ad hoc networks (MANETs). The mobility of nodes implies that the routing protocols of MANETs have to cope with frequent topology changes while attempting to produce correct routing tables. Proactive routing protocols, which are the focus of this paper, provide fast response to topology changes by continuously monitoring topology changes and disseminating the related information as needed over the network. However, the price they pay is the increase in signaling overhead as the topology changes increase, and this can further lead into smaller packet-delivery ratios and longer delays. In the worst case, "broadcast-storms" [19] can result, congesting the entire network. Hence, it is essential to understand the intricate relations between routing overhead and topology changes for the design of routing protocols in MANETs.

Characterizing the impact of mobility on the performance of proactive routing protocols is a very complex problem. Consequently, the provision of such characterization has been limited to simulation-based approaches [20, 21, 22, 23, 24]. Few if any analytical studies have been pursued on this topic. Zhou et. al [25] gave an analytical view of routing overhead of reactive protocols, assuming static network (manhattan grid) with unreliable nodes and concludes the scalability of reactive protocols with localized traffic pattern. Topology changes resulting from node mobility was not considered in [25]. In [26], an information theoretic analysis is pursued to bound the memory requirement and overhead incurred by a hierarchical routing protocol for MANETs based on entropy rate of topology changes.

In this work, we provide the first analytical framework for the modeling of proactive routing overhead as a function of node mobility. In so doing, we model topology changes explicitly as a function of node mobility. The topology (or connectivity graph) of the network at time t can be obtained by replacing the available wireless links with lines connecting the corresponding node pairs. we seek answers to the following questions:

- *Is there an analytical model to statistically characterize the distribution of topology chnages in MANETs? If so, are we able to derive the associated parameters analytically?*

- *If there is such a model, are we able to apply the model to analyze the effect of mobility on the control overhead of proactive routing protocols? Or mathematically, could we find the function $\mathcal{F}$ that projects the control overhead $\mathcal{O}_d$ in MANETs given that we know the node mobility $\mathcal{V}$ and the control overhead $\mathcal{O}_s$ incurred by the protocol in a static topology?*

$$\mathcal{F} : \mathcal{O}_s \times \mathcal{V} \rightarrow \mathcal{O}_d \tag{1}$$

We could have a function $\mathcal{F}$ that projects the control overhead $P(\mathcal{O}_d)$ in MANETs with the knowledge of mobility $\mathcal{V}$ and control overhead $P(\mathcal{O}_s)$ of protocol at static scenarios. And the function can be written as

$$\mathcal{F} : P(\mathcal{O}_d) = \gamma(\mathcal{V}) * P(\mathcal{O}_s), \tag{2}$$

where γ is the *penalty factor* that measures the cost in graph transitions for an active node and as we will see later, it is a function of nodal mobility and stability of the local connectivity graph.

In our analysis, we have computed the control overhead $P(\mathcal{O}_s)$ of protocol at static networks and the penalty factor γ. Utilizing this general framework for proactive routing protocols, we have implemented the results for OLSR protocol model.

2.2.5 A Unified Framework for Analysis of Routing Protocols in MANETs

A unified mathematical framework of evaluating protocol performance for both proactive and reactive routing protocols in mobile ad hoc networks is missing in literature. Given that proactive and reactive routing in MANETs have relative advantages and disadvantages, comparing the two is important. Significant work (e.g., [27, 28, 29, 20]) has been conducted to evaluate and compare these protocols under network profiles of various mobility and traffic configurations. Such performance comparisons have been mostly conducted via discrete-event simulations. Simulation-based studies of routing schemes is indeed a powerful tool to gain insight on their performance for specific choices of network parameters. However, it is difficult to draw conclusions involving multidimensional parameter spaces, because running several simulation experiments for many combinations of network parameters is impractical. We were able to develop a unified framework to analyze routing protocols in MANETs.

As the first analytic treatment of this problem, this work provides a unified analytical framework that parameterizes and quantitatively evaluates the performance of routing protocols from a joint characterization of routing logics and MAC layer. The solution emerges from a combinatory model that synthesizes the evaluation of both routing logics and MAC layer, with each of them being parameterized individually.

Simplified as a two-customer queuing model, the MAC layer at nodes is also simplified through a simple two customer queuing model, where the packet loss probability and delay at nodes can be effectively computed. When analyzing the performance of MAC, we consider both cases of scheduled MAC and contention-based MAC. And in the combinatory model, the computed metrics are synthesized along with routing logics to produce quantitative measures on routing protocols in terms of end-to-end packet loss probability and delay.

We model each wireless node as a two customers queue without priority where the two kinds of customers are unicast and broadcast packets. By analyzing the service time and waiting time in queue for any unicast packet, we derive the transmission delay time for any link; by summarizing the transmission delay time for each link of any routing path, we finally can derive the end-to-end delivery delay.

As the first analytical treatment of the problem, the unified framework provides a parametric view of protocol performance, which in turn enables deeper insight into protocol operations and reveals the compounding and interacting effects of protocol logics and network parameters. Such knowledge further contributes to general guidances on preference of reactive or proactive routing protocols upon various network configurations. The analytical guidance from the model in comparison with discrete-event *Qualnet* simulations has shown similar behavior.

2.2.6 MAC-Layer Performance in IEEE 802.11 Networks

Reference [79] study the MAC layer performance of 802.11 and 802.15.4 CSMA/CA networks. Both papers develop an analytical approach based on a per station Markov model to capture the state of each station at each moment in time. The Markov model yields formulas for throughput and energy which are then validated by comparison with simulations: We use [79] an OPNET simulation and a Matlab simulation. This research was reported in the 2007 Progress Report, so the findings are not discussed here. The papers were revised and have now been published.

2.3 Task 2: Cross-Layer Interactions and Robustness

A Key area of our research has been to study a cross-layer perspective on multicasting in wireless networks, which allows for successful reception depending on the level of interference. Most previous work on random access assumes that a collision results in the loss of all transmitted packets.

Multicast transmission, in which data is sent from a source to multiple destinations, is an important form of data communication in wireless networks. Numerous applications require multicast transmission, including content distribution, conferencing, and military and emergency messages, as well as certain network control mechanisms, such as timing synchronization and route establishment. In addition to practical interest, there is significant theoretical interest in multicast transmission due to its use in demonstrating the benefits of the emerging paradigm of *network coding* [30, 31], in which data from disparate sources is coded together at nodes in a network. We investigate the use of network coding in obtaining the ultimate limits on performance, in terms of both throughput and delay, for multicast transmission.

Infrastructureless or ad hoc wireless networks require distributed implementation. In our work, we consider multiple sources transmitting over a shared channel, which requires some form of medium access control (MAC). We choose to focus our studies on the use of *random access* to the channel [32], which can be implemented in a purely distributed fashion. In the random access protocol we consider, each transmitting node decides independently and randomly whether it will transmit in each time slot. Clearly, this leaves open the possibility that multiple nodes will decide to transmit simultaneously; the resulting *collision* on the channel may cause a level of interference that makes reception impossible.

The specific form of network coding we consider is random or rateless coding, which overcomes the inefficiencies of traditional forms of error correction. For instance, a protocol in which lost packets are retransmitted can be particularly inefficient for multicast communication since each destination may require a different packet to be resent. Alternatively, fixed-rate error control coding can be used, but may not be adaptive to the time-varying nature of the wireless channel. The notion of random or rateless coding has arisen in the fountain coding literature [33],[34] as well as the network coding literature [35]. It involves sending random linear combinations of data packets rather than the original data packets themselves. By sending a potentially limitless stream of random linear combinations of data packets, the effective rate of the code is adapted to the needs of each receiving destination node.

The random coding approach we consider is as follows. Let $s_1, s_2, \ldots, s_K$ denote K data packets awaiting transmission from a node in a wireless network. These data packets are symbols from a finite field $\mathbb{F}_u$, where we refer to u as the alphabet size. Equivalently, the data packets $s_1, s_2, \ldots, s_K$ can be viewed as strings of $\log_2 u$ bits. Random linear coding is performed in the following manner. A coded packet is randomly generated as

$$\sum_{i=1}^{K} a_i s_i \tag{3}$$

where the coefficients a_i, $i = 1, 2, \ldots, K$, are generated randomly and uniformly from the set $\{0, 1, 2, \ldots, u-1\}$ and $\sum$ denotes addition in the finite field $\mathbb{F}_u$. We note at this point that it is possible to generate $a_i = 0$ for all $i = 1, 2, \ldots, K$, meaning that the coded packet is an all-zero packet and contains no information. This outcome could lead to inefficiencies, particularly for small values of K where it is more likely, and would clearly be avoided in a practical system. A coded packet formed according to (3) is the same length as one of the original packets s_i and can be transmitted over the channel within the same time period and utilizing the same bandwidth as one of the data packets s_i. For each coded packet sent, the coefficients a_i used in generating the packet will be appended to the header of the packet. Many different coded packets can be generated according to (3) and a receiving node must recover enough of these random linear combinations in order to decode the original packets.

Parts of our work also deal with the scenario in which multiple transmitting nodes access a shared channel through the use of *random access*. Random access is of particular interest because it is robust to variations in the channel and network topology and can be implemented without coordination among the transmitting nodes. The form of random access we consider is a slotted version of Abramson's ALOHA protocol [32]. We assume that within the network, there is a common time frame established among the nodes so that all nodes maintain a common notion of time evolving in synchronized time slots. In each time slot, if node n has some data to send, it accesses the channel with probability p_n. The value of p_n can be chosen by the node using information it has collected regarding previous transmissions and the level of interference those transmissions suffered.

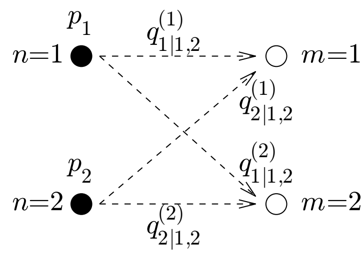

Figure 1: Two source nodes randomly access the channel to multicast to two destination nodes. When both sources transmit, which happens in a slot with probability p_1p_2, the reception probabilities are as shown above.

2.3.1 Throughput and capacity regions for random access

We study throughput and capacity regions for the random access system shown in Fig. 1, consisting of two source nodes and two destination nodes. The data generated at source $n = 1$ is assumed independent of the data generated at source $n = 2$. In each time slot, if source n has a packet to transmit, then we refer to the source as being *backlogged*; otherwise the source is *empty*. A backlogged source transmits in a slot with probability p_n.

The channel model we consider is similar to the model used in [36]. A transmitted packet is received without error with a certain probability. Otherwise, the packet is lost and cannot be recovered. We assume that the channels between different source-destination pairs are independent. We introduce the following *reception probabilities* for sources $n = 1, 2$ and destinations $m = 1, 2$.

$$q^{(m)}_{n|n} = \Pr\{\text{packet from } n \text{ is received at } m | \text{ only } n \text{ transmits}\} \quad (4)$$

$$q^{(m)}_{n|1,2} = \Pr\{\text{packet from } n \text{ is received at } m | \text{ both sources transmit}\} \quad (5)$$

We assume throughout that interference cannot increase the reception probability on the channel, i.e., $q^{(m)}_{n|n} > q^{(m)}_{n|1,2}$. The reception probabilities inherently account for interference and also allow for multipacket reception (MPR). The collision channel model used in many previous works is given by $q^{(m)}_{n|n} = 1$, $q^{(m)}_{n|1,2} = 0$.

We first describe the Shannon capacity region. We say that the rate pair (R_1, R_2) is *achievable* if there exists encoding and decoding schemes such that the data at the source nodes is received at both destinations with an error probability approaching zero as the code blocklength approaches infinity. The capacity region is the closure of all achievable rate pairs and is given in the following result.

Theorem 1: The capacity region of the random access system with two sources and two destinations is the closure of (R_1, R_2) for which

$$R_1 < \min_{m=1,2} \left\{ p_1(1-p_2)q^{(m)}_{1|1} + p_1p_2q^{(m)}_{1|1,2} \right\}$$

$$R_2 < \min_{m=1,2} \left\{ (1-p_1)p_2q^{(m)}_{2|2} + p_1p_2q^{(m)}_{2|1,2} \right\}$$

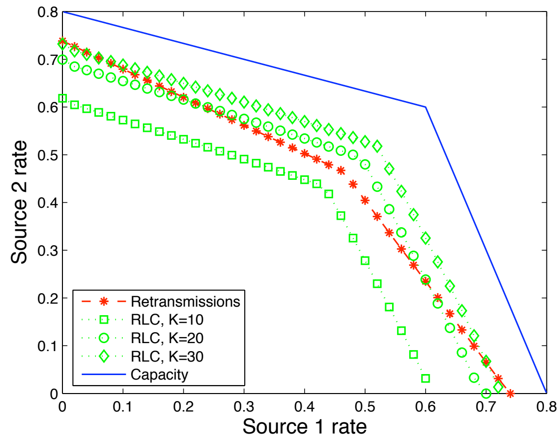

Figure 2: Closure of the throughput and capacity regions for a channel with reception probabilities $q^{(1)}_{1|1} = q^{(2)}_{2|2} = 0.8$, $q^{(2)}_{1|1} = q^{(1)}_{2|2} = 0.7$, and $q^{(m)}_{1|1,2} = q^{(m)}_{2|1,2} = 0.6, m = 1, 2$. Random linear coding is carried out over binary symbols ($u = 2$) and is labeled RLC.

for some $(p_1, p_2) \in [0, 1]^2$. The rate R_n is expressed in packets/slot.

Next, we consider a system in which data packets randomly arrive at nodes $n = 1, 2$ according to a Bernoulli process of rate λ_n packets/slot and are queued in an infinite-capacity buffer. For a given transmission protocol, we define the *stable throughput region* of the system as the set of all arrival rates (λ_1, λ_2) for which there exists a set of transmission probabilities (p_1, p_2) such that the queues of packets at both source nodes remain finite while both destination nodes receive all data arriving at the sources. The stable throughput region is described as follows.

Theorem 2: For a network with two sources and two destinations, the stable throughput region is given by the closure of $\mathcal{L}(p_1, p_2)$ where

$$\mathcal{L}(p_1, p_2) = \bigcup_{i=1,2} \mathcal{L}_i(p_1, p_2) \tag{6}$$

and

$$\mathcal{L}_1(p_1, p_2) = \left\{ (\lambda_1, \lambda_2) : [][c]r'l'l\lambda_1 < \frac{\lambda_2}{\mu_{2b}}\mu_{1b} + \left(1 - \frac{\lambda_2}{\mu_{2b}}\right)\mu_{1e}\lambda_2 < \mu_{2b} \right\}$$

$$\mathcal{L}_2(p_1, p_2) = \left\{ (\lambda_1, \lambda_2) : [][c]r'l'l\lambda_1 < \mu_{1b}\lambda_2 < \frac{\lambda_1}{\mu_{1b}}\mu_{2b} + \left(1 - \frac{\lambda_1}{\mu_{1b}}\right)\mu_{2e} \right\}$$

for some $(p_1, p_2) \in [0, 1]^2$.

To quantify the stable throughput region for random linear coding, we developed schemes to numerically compute the values of μ_{nb} and μ_{ne}. We compared this to a baseline result on the stable throughput when using a retransmission scheme. One example of our numerical results is shown in Fig. 2, where we have plotted the throughput for random linear coding (labeled "RLC") with various values of K. The results show that the Shannon capacity region is strictly larger than the throughput regions for both the retransmissions and random linear coding schemes. Additionally, the throughput region for random linear coding grows with increasing values of K. The random linear coding scheme does not necessarily outperform the retransmission scheme. For small values of K, the coding scheme is inefficient due to the occurrence of all-zero coded packets. However, as K increases, the throughput performance of random linear coding is enhanced.

2.3.2 Queueing delay

Given the well-documented tradeoff between the throughput and delay in a network, a natural question that arises is: does the throughput gain offered by random coding result in a penalty in terms of the delay? There has been relatively little attention given to this question. Queueing delay in a network coding setting has been explored in [37], where in each transmission opportunity, the source node sends a random linear combination of the packets queued in a finite-capacity buffer. Simulation studies are used in [37] to investigate the delay as a function of buffer size. In our work, the random linear coding scheme is analyzed as a discrete-time bulk-service queue. The bulk-service property of the queue captures the fact that packets are encoded, transmitted, and removed from the queue in groups.

In studying queueing delay, we consider a single source multicast system. Data packets arrive to a source node through a Bernoulli process with rate λ packets/slot. Upon arrival, the data packet is placed in a buffer of infinite capacity in order to await transmission. The buffer forms a first-in-first-out (FIFO) queue. When a packet reaches the front of the queue, it is transmitted over the channel. There are M destination nodes and all of them must receive the data packets from the source node. The channel between the source and destination m is an independent erasure channel with reception probability in each slot given by

$$q^{(m)} = \Pr\{\text{coded packet received at destination } m\}, \quad 1 \leq m \leq M. \tag{7}$$

We assume that each source-destination pair has an independent, error-free feedback channel over which the destination node can send acknowledgement messages. We analyze the queueing delay performance when coding is performed over k packets.

We considered two different strategies for random linear coding. In the first strategy, the number of packets used in random linear coding, or the expected coding rate, is a fixed parameter $k = K$, meaning that an additional delay may be incurred in waiting for enough packets to arrive before encoding and transmission can begin. We have developed analytical models for a bulk-service queue that functions in this way, and a numerical example of our results is shown in Figs. 3 and 2.3.2 for a channel with $(q^{(1)}, q^{(2)}, q^{(3)}) = (0.9, 0.9, 0.9)$. In addition to computing delay based on analytical queueing models, we have performed and plotted the results of a Monte Carlo simulation of the random linear coding scheme over 20,000 slots in order to compare the outcome with our approximation. Our approximation closely matches the simulation results for small values of λ, however, the two outcomes differ when the queues saturate. We note that, in contrast to typical queueing delay results, the delay for random linear coding with fixed expected coding rate is not a monotonic increasing function. The delay penalty for $\lambda \approx 0$ is due to the time spent waiting for additional packets to arrive in order to form a group of K packets for encoding and transmission.

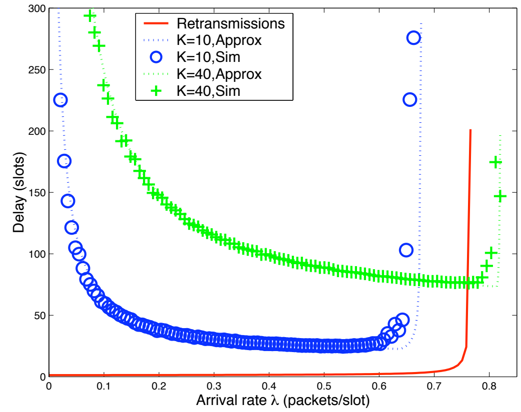

Figure 3: Delay versus arrival rate λ for random linear coding with fixed coding rate over $K = 10$ (o) and $K = 40$ (+) packets. The results are for transmission to M=3 destinations with $(q^{(1)}, q^{(2)}, q^{(3)}) = (0.9, 0.9, 0.9)$.

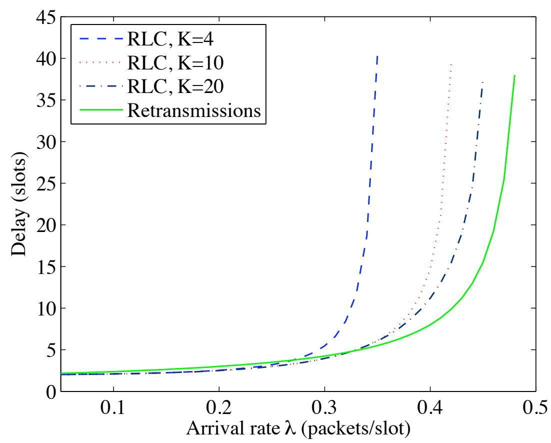

Figure 4: Average delay versus arrival rate λ for variable expected coding rate, $M = 1$ and $q^{(1)} = 0.5$.

In the second strategy we consider, the number of packets used in random linear coding k is a variable, and adapts to the number of packets waiting in the queue when the encoder and channel become free. The variable k can take values between 1 and K depending on the occupancy of the buffer. An example of a numerical evaluation of our bulk-service queueing model for this strategy is shown in Fig. 2.3.2 for a system with $M = 1$ destination node and $q^{(1)} = 0.5$. If this adaptive coding scheme uses parameter K as an upper bound on the number of packets used in coding, and that same parameter K is used in the fixed expected coding rate scheme, then the two schemes support the same stable throughput. In terms of delay, the adaptive coding rate scheme overcomes the delay penalty for $\lambda \approx 0$, and the delay performance of random linear coding is superior to a retransmission scheme.

2.3.3 Packet length, overhead, and effects of the wireless medium

In the discussion above we considered the throughput and delay performance of random coding while treating the wireless channel as an erasure channel with fixed, arbitrary erasure probability. We treated a data packet as a fixed-length sequence of bits without regard for its information content. Additionally, we have disregarded in our analysis the need to transmit overhead information containing the coefficients of the random code. We now revisit the performance of random coding for multicast transmission and specifically account for the packet length, overhead, and physical effects of the wireless channel.

Our motivation stems from interest in applying the technique of network coding to wireless multihop networks of arbitrary topology. The random linear coding we consider throughout this work can be viewed as a localized (single-hop) version of random network coding [35]. It has been shown that by applying network coding for multicast transmission in a wired network, it is possible to achieve the maximum flow capacity in the limit as the symbol alphabet size (or field size) approaches infinity [30, 31]. For random network coding, the overhead inherent in communicating the coding coefficients becomes negligible only as the number of data symbols in the packet n grows large. The "length" or information content of a transmitted packet (i.e., the number of bits conveyed by the packet) depends on both the symbol alphabet u and the number of symbols per packet n. From the previous works listed above, it is clear that transmitting sufficiently long packets is crucial to approaching the maximum flow capacity and to allowing random coding to operate with low overhead.

However, the long packets needed for network coding are more susceptible to noise, interference, congestion, and other adverse channel effects. In particular, we cite the following reasons that the erasure probability will increase with the length of the packet.

- If we try to fit more bits into the channel using modulation, then for fixed transmission power, points in the signal constellation will move closer together, making them more susceptible to noise and errors more likely.
- If we try to fit more bits into the channel by decreasing symbol duration, then we are constrained by bandwidth, a carefully-controlled resource in wireless systems.
- Longer packets take up more space in memory, leading to an increase in the chance of buffer overflows and dropped packets.

In our work, we model the erasure probability q as a function of the packet length, where we denote erasure probability by $q(n,u)$, and investigate the implications on performance of random coding for multicast. We also scale the throughput to account for the overhead needed to perform random coding - we refer to this scaled throughput as *effective throughput*. We obtain the following results, which provide a lower bound on the throughput.

Theorem 3: The effective multicast throughput S in packets per transmission for random linear coding over groups of K packets is lower bounded, for any $t > 0$, as

$$S \geq \frac{tK}{\ln M + \ln E[e^{t\widetilde{T}}]} \frac{n}{n+K} \tag{8}$$

where random variable $\widetilde{T}$ denotes the number of slots needed for successful recovery of the K packets at one of the M destinations and n denotes the number of information symbols per packet.

Theorem 4: For a time-invariant channel with erasure probability $q(n,u)$ in each slot and random linear coding over K packets, the moment generating function of the number of slots needed for a destination node to recover the original K packets is given by

$$E[e^{t\widetilde{T}}] = \prod_{j=0}^{K-1} \frac{q(n,u)\left(1 - u^{-K}(u^j - 1)\right)}{e^{-t} - 1 + (1 - u^{-K}(u^j - 1))\, q(n,u)} \tag{9}$$

2.4 Task 3: Theory of Scalable and Robust Protocols

In this task, we investigated new theories of scalable and robust protocols with emphasis on channel access and routing. We investigated channel access protocols for multi-channel networks based on the insight gained in our prior work on fundamental limits for multi-channel networks, and back-off mechanisms that operate reliably in fading environments.

We explored the scalability of a epidemic dissemination approach (RelayCast) in an intermittent topology environment with delay tolerant applications. We developed bounds for throughput, delay and buffer requirements as a function of network size. We also developed ProbeCast, a novel methodology for QoS support of multicast inelastic flows. We use incremental bandwidth probing instead of bandwidth preallocation to establish QoS paths with guaranteed bandwidth for accepted flows.

We evaluated the efficacy and feasibility of network coding (NC) for multicast applications in challenged environments (mobility, interference, jamming and random loss). We addressed the tradeoffs involved in the design of a NC protocol on resource limited (buffer, processing) nodes. We illustrate our design tradeoffs with a real implementation. Our research addressed the well known and controversial issue of NC vs. Erasure Coding, which works best in different mobility, topology, error and load situations. We obtained preliminary results for a static, error-free topology where NC prevails over Erasure Coding.

2.4.1 MAC Protocols for Multi-Channel Wireless Networks

In prior work [54, 55], we had investigated the implications of interface heterogeneity, i.e., having radios with heterogeneous channel switching capabilities, in a multi-channel wireless network, through asymptotic capacity analysis. Another form of heterogeneity that arises in a multi-channel wireless network pertains to different channels having different rate characteristics. This can complicate the task of scheduling in a multi-channel network. For instance, consider the class of scheduling policies termed "maximal schedulers". In a multi-channel wireless network with heterogeneous channels, the performance of these policies can exhibit severe degradation dependent on the skew in channel-rates. In our work, we have studied this issue. We have established a lower bound on the performance of a centralized greedy maximal scheduler in a multi-channel wireless network with heteregeneous channels, but identical radios [56]. This bound improves upon an existing bound for such networks. We have also designed a scheduling algorithm that uses a maximal scheduler in conjunction with a local rule to distribute load on a link across channels [56]. This algorithm does not require information about queue-lengths at interfering links, as compared to other recently proposed approaches in the literature. While the performance with lesser information is inferior to the performance of the latter, the provable performance bound is much better than a naive maximal scheduler. Thus, this algorithm exemplifies a viable performance-information trade-off. We have also done some work on the issue of interface heterogeneity in the context of maximal scheduling policies.

Building upon intuition derived from our theoretical results, as well as from other existing theoretical results in the literature, we have designed a protocol for channel and interface management in a heterogeneous multi-channel multi-radio wireless network as a proof-of-concept of the possibility of evolving a generalized conceptual design approach toward handling various kinds of physical layer heterogeneity [57]. Our objective has been to incorporate awareness of radio and channel heterogeneity as well as traffic-awareness into the channel and interface management procedures. While we have designed our protocol in the context of 802.11 networks with 802.11a, 802.11g, and 802.11ag radios, with certain assumptions on node configuration, many aspects of the design, and many of the algorithms used by the protocol, are conceptually formulated in fairly general terms, and have broader relevance for a wide range of networks with heterogeneous radios and/or channels. We have evaluated the performance of this protocol via simulation.

2.4.2 Exploiting Concurrency in Channel Access Protocols

Our recent results on the capacity of wireless neteworks [18, 96] clearly show that embracing concurrency at the physical layer is a must in order to make wireless networks scale. In practice, this requires nodes to send and receive multiple concurrent transmissions successfully. The approaches that have been reported to embrace concurrency to date have focused on using MIMO radios, directional antennas, analog network coding [97], and combining frequency and space division with the use of multiple radios per node [98, 99, 100, 101]. However due to physical restrictions on the number radios per node, the number of non-overlapping channels and channel switching delays, and the complexity of MIMO radios, the concurrency that can be attained in real ad hoc networks is limited.

Orthogonal Frequency Division Multiple Access (OFDMA) [102] has been selected for use in multi-user environments (e.g., IEEE 802.16 and DVB-RCA [103]) due to its ability to combat wireless channel's multipath effects and to facilitate flexible concurrency. In OFDM systems, *subcarriers* or *tones* are orthogonal carriers of lower-rate input data streams that result in longer symbol duration compared to channel delay spread to mitigate multipath effects. In OFDMA, a group of non-overlapping subcarriers called *subchannels* can be assigned to each user to enable simultaneous data transmission to a single receiver or simultaneous data transmission from a single transmitter to multiple receivers. While OFDMA has inherent concurrency characteristics in the way that multiple subchannels can be allocated and used, they have not been truly exploited in the past. The adoption of OFDMA in ad hoc networks has been explored only by a number of recent works [104], [105], [106], [107], [108], [109] that focus on achieving the best arrangement of bits, power and subcarrier per user to maximize channel throughput. Scheduling in time and frequency for mesh networks where centralized mesh routers are responsible for subchannel assignment is discussed in [106], and [107]. Although proposals exist [108, 109] for resource allocation algorithms in terms of power, bit, and subcarriers, they do not propose a complete medium access control (MAC) protocol.

Over the past year, we have studied a number of alternatives for channel access in MANETs based on the use of OFDMA to exploit concurrency. We have published preliminary results at the ICCCN 2008 conference [121], and we have submitted additional work to other conferences. The Parallel Interaction Medium Access (PIMA) protocol constitutes our first result in this area. PIMA is the first MAC protocol that permits a sender to transmit multiple packets concurrently and a receiver to decode multiple packets concurrently by orchestrating channel access over time and frequency based on OFDMA in ad hoc networks where nodes are equipped with single half-duplex antennas. We used an analytical model and simulation results to show that PIMA provides substantial benefits compared to prior multichannel MAC protocols based on channel switching. The key advantage of PIMA over prior multi-channel MAC protocols is the elimination of channel switching delays and the reduction of handshake signaling overhead for the assignment of channels to nodes.

Based on our preliminary results, we continue to work on similar schemes, which in a nutshell consist of three components: (a) a *channel state assignment* algorithm with which each node pair in the two-hop neighborhood of a node is given a transmission state; (b) a joint negotiation algorithm that is receiver initiated and allows a node to exchange with neighbors lists of targeted transmitters, as well as associated channels selected for each transmission; and (c) mechanisms for frequency and time synchronization of OFDMA transmissions. The objective of using these elements is to embrace concurrency by allowing a node endowed with a single antenna and a half-duplex OFDMA radio to either transmit or receive multiple concurrent transmissions at a given time.

2.4.3 Dynamic Spatial Backoff in Fading Environments

Past research on contention-based medium access control (MAC) protocols usually focused on temporal approaches. That is, when two nodes are competing for the same channel, their accesses to the channel are separated in time to ensure successful transmissions.

In our prior work [58], we studied an alternative contention resolution approach for wireless networks, called *spatial backoff*, which adapts the space occupied by transmissions so that channel contention may be resolved more efficiently in both spatial and temporal dimensions. Usually, the space is in part dependent on the transmission rate and the carrier sense (CS) threshold of the wireless stations. One popular scheme that is used in most wireless LANs is Carrier Sense Multiple Access (CSMA), in which each station senses the medium before it attempts to transmit. If the signal level detected at a station is below a CS threshold, the channel is assumed to be idle, and the station may access the medium. Assuming a fixed transmission power is used by every station in the network, a node will transmit more aggressively using higher CS thresholds. On the other hand, notice that the more concurrent transmissions (i.e. the higher the CS thresholds used at the transmitter), the higher the interference at the receivers, and therefore the lower the signal-to-interference-and-noise ratio (SINR), and consequently the lower transmission rates the packets can be transmitted reliably. In contrast, the fewer the concurrent transmissions, the higher the SINR at the receivers, and the higher transmission rates the packets can be transmitted reliably. Therefore, in order to achieve a high aggregate throughput in a network, there exists a trade-off between CS threshold and transmission rate at each transmitter.

A spatial backoff algorithm developed in our prior work [58] attempts to approach the optimal trade-off between CS thresholds and transmission rates to achieve high aggregate throughput in an ad hoc network. However, the performance of this algorithm suffers in the presence of fading. We have developed a modified version of this algorithm that can resolve channel contention using spatial backoff and achieve a high level of aggregate throughput in ad hoc networks, even with channel and traffic variations over time [59].

2.4.4 Applications of Network Coding to Network Control

While network coding has been shown to offer various advantages over traditional routing, in order to see network coding widely deployed in real networks, it still remains to show that the amount of overhead incurred by additional coding operations can be kept minimal and eventually outweighed by the benefits network coding provides.

Whereas most network coding solutions assume that the coding operations are performed at all nodes, we have pointed out in our previous work [46] that it is often possible to achieve the network coding advantage by coding only at a subset of nodes. Determining a minimal set of nodes where coding is required is NP-hard [46], as well as finding its close approximation [53]. Thus, we have previously proposed evolutionary approaches [46, 47, 48, 49, 50] toward a practical multicast protocol that achieves the full benefit of network coding in terms of throughput, while performing coding operations only when required at as few nodes as possible.

Suppose that we have an ad hoc wireless network currently operating solely with traditional routing and wish to employ network coding on the network to achieve the maximal throughput promised by the network coding theory. However, for handling of the mathematical operations required for coding, the network nodes that have to perform network coding may need some changes in their software and/or hardware, which would necessarily incur some additional cost. Therefore, a very interesting and also practically important question that may arise in such a situation is whether we should change the entire nodes in the network for network coding or modifying only a subset of nodes would be enough. Along the same direction, many more interesting questions may follow: If only a small number of coding nodes are enough, as previously found in [46, 48] with a generic network model, where should those coding nodes be located? Can we fix their locations despite varying communication demands? How should those coding nodes interact with other non-coding nodes?

In our recent work [51], we considered the multicast scenario in a heterogeneous ad hoc wireless network where a number of coding nodes are to be placed among the legacy nodes that do not handle network coding operations well, to which our previously proposed evolutionary framework is applied to provide better understanding, if not complete answers, of the questions raised above. In particular, we have shown in [51] that our evolutionary approach is well generalized to the case of heterogeneous wireless networks with only slight changes in the fitness function and the backward evaluation phase, retaining its key advantages over existing centralized approaches [52, 53] in terms of the efficient operation through its spatially and temporally distributed structure and the superior performance in finding a minimal set of coding nodes.

Once a set of coding nodes is given, we can run our distributed algorithm directly over the network, during which each legacy node temporarily emulates coding operations on the application layer to find a minimal set of the legacy nodes that have to keep performing the emulated coding during the normal operation of the network to achieve the multicast capacity.

More interestingly, we can utilize our evolutionary approach to investigate various issues regarding where and how to place coding nodes in heterogeneous wireless networks, which led to the following findings:

1. To find out how many coding nodes are required to achieve the multicast capacity, we generated 100 random wireless topologies for each of 18 different parameter sets with either 20 or 40 nodes and the number of receivers varying from 4 to 10. We found that in most cases (over 90% for networks with 20 nodes and from 57% to 86% for networks with 40 nodes) network coding is not needed at all to achieve the multicast capacity. Even in the cases where network coding is needed, the number of coding nodes does not exceed 15% of the total number of nodes; the number of required coding nodes is at most 3 in networks with 20 nodes and 5 in those with 40 nodes. This suggests that, while it is necessary to *assume* network coding everywhere initially to calculate the multicast capacity, to actually achieve the calculated maximal throughput coding operations may *not* be needed *at all*, and even when needed, only at a very small subset of nodes, thus incurring very little amount of overhead.

2. With a slight change in the fitness function, we have found that, for the coding nodes found above, it is often possible to find alternative nodes to perform coding operations to achieve the given multicast capacity. In other words, the location of the coding nodes can often be flexible rather than fixed for a specific traffic pattern, implying that the found coding nodes can be shared by different multicast requests.

3. To verify the intuition of sharing coding nodes among different communication demands, we picked two representative topologies from those used in the first experiment, and for each topology we randomly generated another 30 multicast requests to find the locations of the coding nodes required for different traffic patterns. We found out that at least half of the coding nodes were in common with different communication demands for the both topologies. This indicates that placing coding nodes at the commonly found locations for a number of sampled traffic patterns may be a good strategy in practice.

4. It is worth to point out that at the end of the iteration, our algorithm yields not only the set of the coding nodes, but also the relevant network code at each interior node, whether it indicates coding or routing. Therefore, the problem of interactions among coding and non-coding nodes is already dealt with implicitly within the framework of our algorithm, whether the algorithm is used to minimize the number of legacy nodes that have to emulate coding with the given set of coding nodes or to find the potential locations of the coding nodes.

In addition, given the vulnerability of ad hoc wireless networks to various kinds of losses, we have shown in [51] that, thanks to its iterative nature, our algorithm can operate without much disruption even in the presence of a moderate level of packet erasures caused by various reasons. Furthermore, our algorithm may become even more robust by employing the temporally distributed structure in our previous work [49], whose initial motivation was better utilization of computational resources over the network. In [51], we have shown that the temporally distributed structure also offers a significant advantage in overcoming the adverse effect of packet erasures on the performance of the algorithm.

2.4.5 RelayCast: Scalable Multicast Routing in Delay-Tolerant Networks

Routing and multicasting in a Delay Tolerant Network (DTN) is challenging because conventional MANET protocols can withstand only very short term path interruptions; they systematically fail when the network stays disconnected for a prolonged time. In favorable motion conditions, DTN routing protocols can overcome such intermittent connectivity by exploiting a *mobility-assisted "epidemic dissemination"* strategy: a node receives, carry in storage, and waits for opportunities to transfer packets to destination node(s).

Scalability is a very important metric when designing a routing protocol both in MANETs and in DTNs. For unicast, the scaling behavior is well understood. In their seminal work, Gupta and Kumar showed that the scalability of *wireless multi-hop routing* is limited; in fact, in a wireless network with n static nodes, each engaged in a data transfer to a random destination the per node throughput decays as $\Theta(1/\sqrt{n \log n})$. Realizing that the increasing hop length of a path is the key limiting factor when the number of nodes increases, Grossglauser and Tse showed that under random mobility assumptions a two-hop relay routing strategy, a mobility-assisted routing protocol that exploits mobility and carry-forward to reduce number of hops can achieve $\Theta(1)$ throughput per node, thus exhibiting a perfect scaling behavior. However, the throughput improvement comes at the cost of increased delay. This result has been followed by a flurry of research activities that tried to characterize the delay/capacity relationship as a function of node mobility.

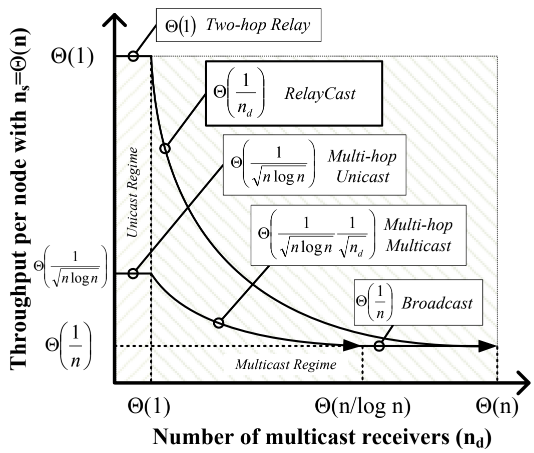

Figure 5: Throughput scaling result comparison

The scaling throughput properties in static wireless networks were recently generalized also to multicast and broadcast. Assuming that there are n_s sources each of which is associated with n_d random destinations and that the packets are delivered on multicast trees, the throughput per multicast source is $\Theta(\frac{\sqrt{n}}{n_s\sqrt{\log n}}\frac{1}{\sqrt{n_d}})$. The penalty of using a multicast tree is high; namely, it corresponds to a factor of $\sqrt{n_d}$ throughput decrement. When the number of multicast receivers is above a threshold value of $\Omega(n/\log n)$, multicasting scales as network wide broadcasting. Thus, its throughput becomes $\Theta(1/n_s)$. This follows from the fact that above the threshold the multicast protocol can fully benefit from the wireless broadcasting effects.

In this task, we have sought to improve the pathological throughput bound of wireless multicast using a mobility-assist routing algorithm. Namely, we proposed RelayCast, a routing scheme that extends the Grossglauser and Tse's two-hop relay strategy by requiring that a relay node be responsible for delivering packets directly to each multicast receiver [69]. This extended protocol was analyzed under the assumption that inter-contact time of an arbitrary pair of nodes follows an exponential distribution with rate λ. We compared throughput and delay properties of RelayCast with those of conventional multicast. In favorable mobility conditions, RelayCast offers two main benefits: it improves throughput scalability with increasing number of nodes, and; it provides reliable delivery even in DTN scenarios with intermittent connectivity. We then analyzed the impact on RelayCast throughput performance of various network and routing parameters including buffer size, multicast receiver relay, and delay constraints.

The following is the main results of this part of of work [69].

- We found throughput upper bound of DTN multicast routing and proposed RelayCast, a two-hop relay based DTN multicast routing protocol, that achieves the upper bound, namely throughput = $\Theta(n\lambda)$ for the case $n_s n_d = O(n)$ and $\Theta(\frac{n^2\lambda}{n_s n_d})$ for the case $n_s n_d = \omega(n)$, or simply $\Theta(\min(n\lambda, \frac{n^2\lambda}{n_s n_d}))$. We also showed that the delay of RelayCast is $\Theta(\frac{\log n_d}{\lambda})$. In particular, for a given $\lambda = \Theta(1/n)$, the throughput and delay of RelayCast are $\Theta(\min(1, \frac{n}{n_s n_d}))$ and $\Theta(n \log n_d)$ respectively. The throughput scaling results are summarized in Figure 5.

- We showed that RelayCast requires buffer space of size $\Theta(nn_d)$ for the case $n_s n_d = O(n)$ and of size $\Theta(\frac{n^2}{n_s})$ for the case $n_s n_d = \omega(n)$. Given finite buffer space of size K, the throughput per multicast source of RelayCast is reduced to $\Theta(\frac{K\lambda}{n_d})$.
- We proved that multicast receiver relay where multicast receivers cooperate in delivering the packets cannot improve the delay with respect to RelayCast, unless the number of multicast receivers scales as $n_d = \Theta(n)$. In this case, we showed that there is an optimal multiple message "gossiping" protocol that can reduce the delay to $\Theta(\frac{\log n}{n\lambda})$.
- We computed throughput-delay trade-offs where throughput is traded for delay. In particular, we analyzed RelayCast with k-copy replication and show that its throughput and delay are $O(\min(\frac{n\lambda}{k}, \frac{n^2\lambda}{kn_s n_d}))$ and $O(\frac{k \log k + n \log n_d}{nk\lambda})$ respectively.

Work is progressing in several directions. First, we are investigating the impact of general mobility models that show power-law inter-contact time distribution on the DTN multicast capacity. Second, using our analytic tools, we will evaluate the scaling behavior of other DTN multicast routing protocols proposed in the field. Also, we will compare the performance in a unified DTN framework (using both analysis and simulations). Finally, we are working on actual DTN multicast testbeds by implementing alert message or advertisement dissemination using Bluetooth enabled cell phones. This extends our previous work on Bluetooth based content distribution [70].

2.4.6 ProbeCast: MANET Admission Control via Probing

An inelastic flow is a flow with inelastic rate: i.e., the rate is fixed, it cannot be dynamically adjusted to traffic and load condition as in elastic flows like TCP. Real time, interactive sessions and video/audio streaming are typical examples of inelastic flows. Reliable support of inelastic flows in wireless ad hoc networks is extremely challenging because flows and routes dynamically change and flows compete for the shared wireless channel. Bandwidth must be reserved for inelastic flows at session set up time. To avoid repeated attempts to set up reservations in a "volatile" network and prevent serious network capacity degradation due to call set up overhead, a Call Admission Control strategy robust to mobility must be developed. In this project, we propose ProbeCast [71], a probe based call admission control scheme with QoS guarantees for inelastic flows. ProbCast was designed for multicast streams but can also work, by default, for unicast.

In ProbeCast, a path (or a tree) is probed for capacity availability. If an intermediate link along the probed path fails to meet the QoS requirement, the flow is "pushed back" via backpressure upstream to the source. As a result, the incoming flow is either rerouted or rejected. The backpressure principle is simple; however its implementation requires some care to avoid unfairness and eventually capture by one of the flows sharing a congested bottleneck. Proportional fairness among inelastic contenders can prevent capture. Say, when the link queue overflows, packet drop rate is proportional to flow rate; thus, drop

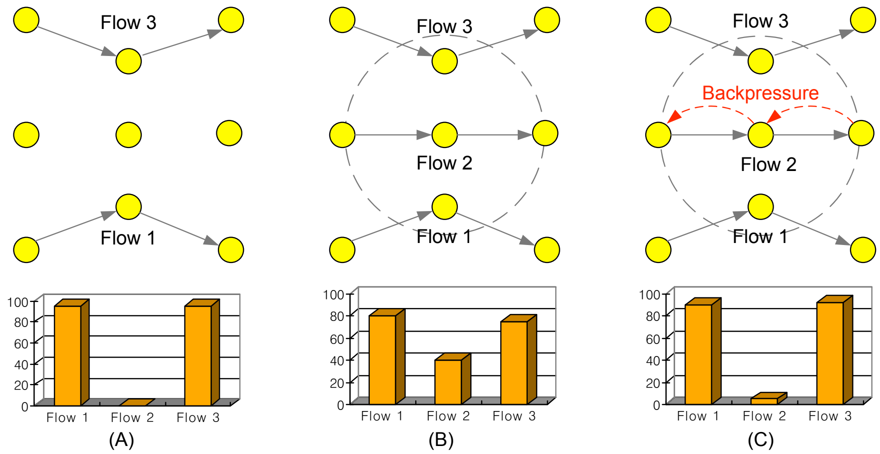

Figure 6: Three flows in the simple topology. Lower graphs show packet delivery ratios, presented by percentage. (A) Two flows are present oth with have high delivery ratios over 90%. (B) Flow 2 starts transmitting and other flows' delivery ratios decrease because of hannel contention. (C) Flow 2 packet drop rate is exceed the threshold and backpressure start.

probability is uniform across flows. This condition is automatically met by Internet links. In the wireless medium, there is opportunity for unfairness and capture by "big flows" when the medium becomes congested. To prevent this, we have developed a distributed fairness scheme, *Neighbor Proportional Drop (NPROD)* which guarantees uniform drop probabilities among flows competing in the same contention domain.

In Probe Cast, each flow has a drop probability threshold. The incoming flow has by design a consistently lower drop probability threshold than the preexisting flows. If during probing the drop rate increases beyond the threshold, the flow is backpressured. If the bottleneck becomes over allocated and there is no available detour, Probe Cast will reject the incoming flow. Figure 6 shows a ProbeCast process in the simple topology. This is accomplished without incurring the reservation and allocation overhead of traditional CAC. The major contribution of ProbeCast is to enable CAC and fair allocation of inelastic flows in MANETs, for both unicast and multicast streams, without requiring prior resource reservation and thus overcoming the overhead limitations of traditional MANET reservation and allocation CAC schemes.

We have developed ProbeCast using the Qualnet simulation platform and compare the performance of ProbeCast to pure ODMRP. We first tested N-PROD ability to enforce proportional unfairness. The scenario is a simple parallel topology in Figure 6 and each source sends data packets at different rates. Namely, Flow 1 = 800Kbps, Flow 2 = 400Kbps and Flow 3 = 200Kbps. As we expected, flow 1 and 3 capture the channel and Flow 2 is starved when ODMRP runs. With N-PROD each flow drops packets proportionally to its demand. Thus, drop probabilities are uniform and achieved throughputs are staggered as the demand as shown in Figure 7. This result is clearly different from what could be achieved with other ad hoc fairness algorithms. Now we tested ProbeCast performance in which 30 nodes are randomly distributed in a 1000m by 1000m area. 3 multicast sessions send packets

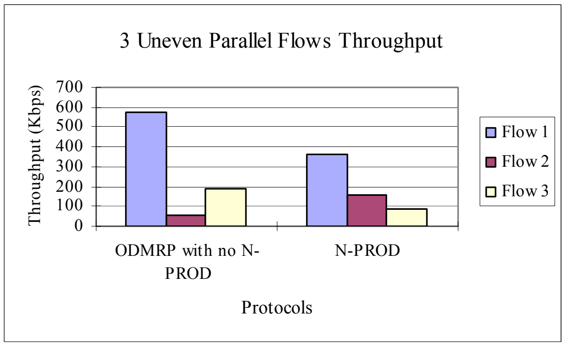

Figure 7: Throughput of uneven three inelastic flows case. Flow 1, 2, and 3 are 800Kbps, 400Kbps, and 200Kbps respectively.

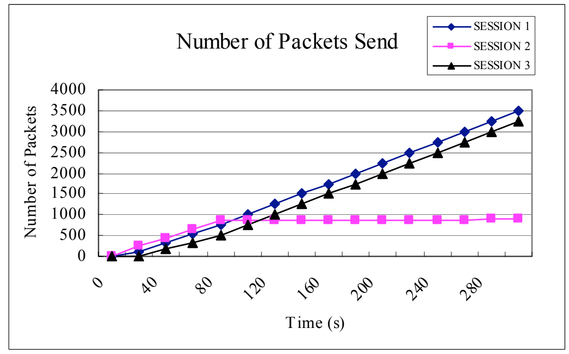

Figure 8: Three multicast sessions are present. Session 2 is rejected.

with 500Kbps rate and each of them has 3 receivers. When all three sessions are injected, the results reported in Figure 8 shows that one of three sessions is consistently rejected. At the beginning, the sessions try to balance drop rates and partially succeed. However, as time progresses, this balance collapses. One session starts dropping packets in bursts and packet drop rate suddenly skyrockets, exceeding the threshold and triggering backpressure on the newcomer (with lower drop threshold).

As we mentioned, major contributions of ProbeCast are: the robust and efficient CAC mechanism based on probing; the ability to handle both uni and multicast inelastic CAC, and; the ability to handle both inelastic and elastic flows at the same time. Simulation results show confirm our claims. Future work will examine the use of QoS multicast tree construction heuristics to facilitate rerouting in case of CAC blocking and extended ProbeCast to interfering (but not communicating) nodes.

2.4.7 Network Coding Processing Overhead Analysis in Content Distribution Scenarios

Content distribution in vehicular networks, such as multimedia file sharing and software updates, poses a great challenge due to network dynamics and high speed mobility. In recent years, network coding has been shown to efficiently support distribution of content in such dynamic environments, thereby considerably enhancing the performance. While the benefits of network coding have been extensively demonstrated in theory and for a number of applications, the practical issues of implementing and deploying network coding for content distribution in MANETs have not been well established. In this work, we develop models that account for nodal resource constraints such as CPU consumption, memory access, and disk I/O. These factors are critical since network coding introduces significant processing overhead at the intermediate nodes.

The goal of this work [64] is to model and evaluate network coding resource consumption in content distribution applications. To this end, we abstract the overall behavior of the application, develop models for computation and disk I/O for a given network coding configuration, and integrate these models into an off-the-shelf discrete time network simulator. This allows us to better understand the impact of limited resources in large scale VANET scenarios.

Disk I/O overhead model: Disk access delay consists of three factors: seek time, rotational latency, and transfer time. In order to generate a k-KB coded packet, we need to read all the corresponding k-KB data per. We call this "chunk-based reading." The access pattern will be a sequence of seek/read pairs. Let θ denote the average latency to perform a pair of seek/read operation. The overall time to read all the relevant data takes $T_d = \theta \cdot G$. The seek latency may be quite prohibitive in the case of mechanical disks compared to SSDs, because the latency is proportional to the generation size. As an alternative, a node can sequentially read the entire piece at once (as in E/S pipeline). In this case, the disk I/O latency is given as $\frac{GB}{R_d}$ where B is the piece size and R_d is the data transfer rate.

Computation overhead model: For a given generation g, let $p'_{g,k}$ denote the kth code symbol in a coded piece, and $p_{g,i,k}$ denote the kth symbol of the ith piece in the buffer. Let c_i for $i = 1, \cdots, G$ denote the ith encoding coefficient, which is randomly chosen over a Galois Field of size 256 once at the beginning of the entire procedure (i.e., symbol size is 8bit). Each code symbol $p'_{g,k}$ is generated as follows:

$$p'_{g,k} = c_1 \cdot p_{g,1,k} + c_2 \cdot p_{g,2,k} + \cdots + c_G \cdot p_{g,G,k}$$

For each symbol ($p_{g,i,k}$) it requires a pair of multiplication (i.e., $c_i \cdot p_{g,i,k}$) and addition ($p'_{g,k}$ $+= c_i \cdot p_{g,i,k}$). The per-symbol encoding time is proportional to the generation size G, i.e., $T_e = G \cdot \delta$ where δ is the time of executing the pair of operations. Let R_e denote the per-symbol encoding rate (byte/sec). Then, the rate is given as follows:

$$R_e = \frac{1}{T_e} = \frac{1}{\delta} \cdot \frac{1}{G} \tag{10}$$

Equation 10 shows that the encoding rate is the function of δ and G. The value δ is purely dependent on the Galois field operation implementation and the processing power.

Effects of Disk I/O and Computation O/H: Figure 9 shows the results of the ideal case. The figure shows that as the number of generations increases, the download delay also increases. This confirms that the number of generations must be kept as small as possible to achieve a good performance. In Figure 10, we show the case of buffer size = 100% to show the impact of computation overhead. Unlike previous results, we notice that the single generation scenarios perform worse than other scenarios, especially when the file size is large (i.e., 50M and 25M). Yet it is still better than the *No Coding* scenario where the generation size is 2500 for a 50MB file and 1250 for a 25MB file, and the corresponding encoding rates are 19.2KB/s and 38.4KB/s respectively. This clearly shows that the encoding rate is a bottleneck. As the number of generations increases, the effect of computation overhead reduces. However,

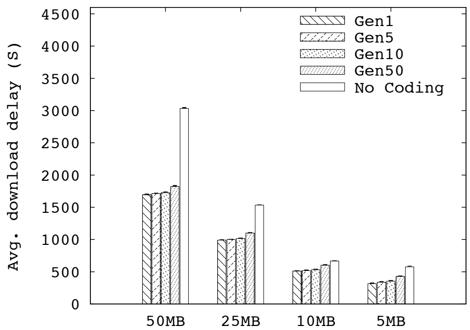

Figure 9: Download delay without O/H

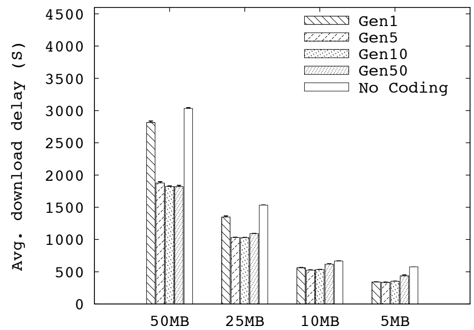

Figure 10: Download delay with O/H: Buffer 100% (10MB:N and 5MB:N show the results of Nokia N800 configuration)

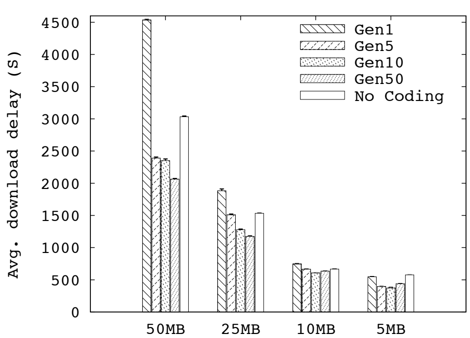

Figure 11: Download delay with O/H: Buffer 50%

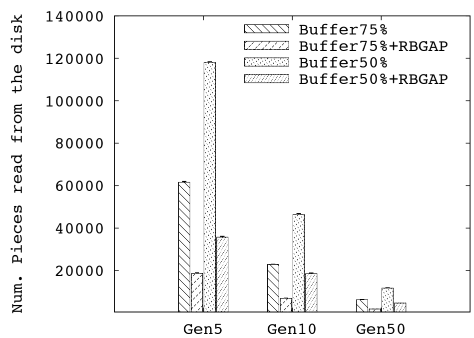

Figure 12: Total number of pieces read from the disk (50MB file)

if the number of generations is above a certain threshold, the download latency begins to increase. The figure shows a "U" shape delay curve for both 25MB and 50MB files.

Figure 12 shows the download delay and the number of pieces read from the disk with different buffer sizes, namely 100%, 75% and 50%. Note that the disk I/O overhead is proportional to the number of pieces per generation, and the probability that the requested generation is not in the buffer is mainly determined by the buffer size. Thus, the impact of finite buffer decreases with the number of generations. The figure shows that RBGAP can effectively reduce unnecessary disk I/Os, thereby reducing the total downloading delay.

From the simulation study, we have obtained several new insights that will help improving the performance of applications based on network coding. They include: (1) resource constraints have a significant impact on the performance of network coding; (2) data pulling that considers the resource constraints of remote nodes can significantly improve the performance; (3) the benefit of sparse random network coding is not always obvious, and its parameter should be carefully chosen to perform well; and (4) generation level pre-coding is not as efficient in a highly dynamic environment as VANETs.

2.4.8 Network Coding vs. Erasure Coding: Reliable Multicast in Ad hoc Networks

Network coding has recently been applied with success in wireless ad hoc networks. Several protocols, e.g., CodeCast, ExOR, and MORE, exploit network coding for reliable data transfer in various wireless ad hoc and mesh network scenarios. Erasure coding is such a well known coding scheme considered to be used for wireless ad hoc reliable communications. Both schemes provide excellent ammunition against channel error and network disruption. However, "*the jury is still out*" regarding which scheme is better in wireless ad hoc networks, especially multicasting. In this project, we report on experiments that provide information on how to select the coding scheme in ad hoc multicast scenarios [72].

We compare reliability and overhead that a multicast application can achieve with network coding and with erasure coding in unreliable wireless networks using random linear programming formulations. A source splits the stream of original packets into batches of k packets, called *generation size*. Erasure coding generates n encoded packets so that we define *code rate* $c = k/n$ $(n \geq k)$. Accordingly, network coding code rate is always 1. Network coding, however, allows re-encoding of packets at intermediate nodes whereas intermediate nodes in erasure coding relay received packets. We adopt simple probabilistic routing instead using constructing sub-graph algorithms in order to providing the same conditions to both schemes. We call *forwarding probability* f $(1 \geq f \geq 0)$ hereafter. To realize an erasure channel, each node drops received packets with a certain probability called *packet drop probability* d $(1 \geq d \geq 0)$.

Figure 13 (a) is the simple grid topology that has a single source and 3 receivers. There are r multiple paths and h hops between the source and receivers. In the grid topology, any single-hop transmissions except the first hop can be represented by single-hop models in figure 13 (b) and (c). Since packets are re-encoded at intermediate nodes, we approximately consider network coding to be an asymptotic single-hop model having multiple senders, r. However, erasure coding can be assumed a sender transmits the same packet r times similar

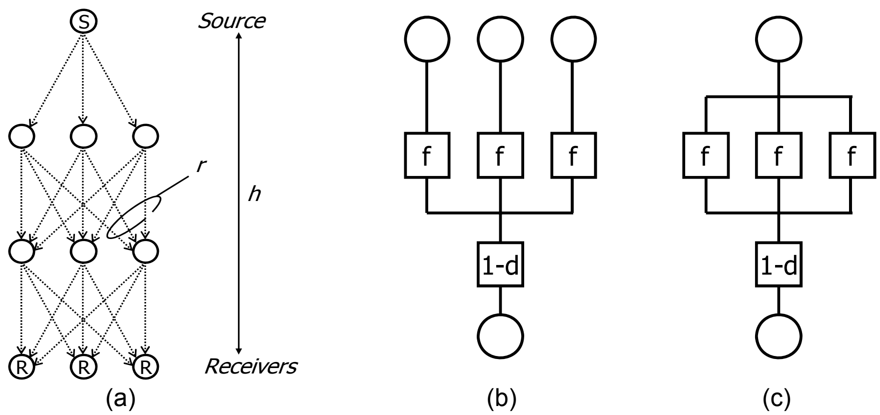

Figure 13: (a) A simple illustration of the grid topology. There are r multi-path and h hops between a source and eceivers. (b) Network coding abstract model. f is forwarding probability and d is drop probability. (c) Erasure coding abstract model. f is forwarding probability and d is drop probability.

to a parallel model: a packet is delivered if one of muliti-path works.

Let N_{NC} sand N_{EC} denote the number of innovative packets a node can potentially receive in network and erasure coding respectively. Given the single-hop models, we can derive:

$$N_{NC}(h) = \begin{cases} fr(1-d)N_{NC}(h-1), & if\ N_{NC}(h-1) \leq k, \\ fr(1-d), & if\ N_{NC}(h-1) > k, \\ (1-d)k, & if\ h = 1, \end{cases} \tag{11}$$

$$N_{EC}(h) = \begin{cases} \{1-(1-f)^r\}(1-d)N_{EC}(h-1), & if\ h > 1, \\ (1-d)\frac{k}{c}, & if\ h = 1, \end{cases} \tag{12}$$

where k is the generation size; c is the code rate $=k/n$; h is the number of hops; r is the number of multiple paths.

Equation (11) and (12) show that multi-path increases the number of received packets in the case of network and erasure coding. However, the number of received packet in network coding is linearly increased as multi-path increases while the number of received packet in erasure coding is slowly increased. Moreover, coding rate significant affects the erasure coding performance. Figure 14 illustrates delivery ratio in the grid topology in Figure 13 when h=5. A network coding delivery ratio line corresponds to the erasure coding delivery ratio with coding rate c=1/3. Though erasure coding outperforms network coding in terms of the delivery ratio when $c < 1/3$, erasure coding overhead is at least 2 times bigger than network coding's in Figure 15. In the random topology, results are similar to the grid topology. We compare the delivery ratio of both schemes with varying node density which is the average number of neighbor nodes within a radio range. In Figure 16, network coding delivery ratio shows the same performance to erasure coding with coding rate c=1/6. When nodes have mobility or packet drop rate increases, the results in the random topology show the same pattern.

In this project, we compare network and erasure coding in wireless multicast scenarios. The simulation results and theoretical analysis demonstrated that network coding can achieve high reliability with lower overhead. To improve theoretical analysis part and simulation with existing novel routing schemes are an extension of our work and under investigation.

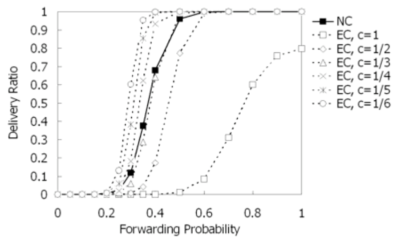

Figure 14: Delivery ratio in the grid topology.

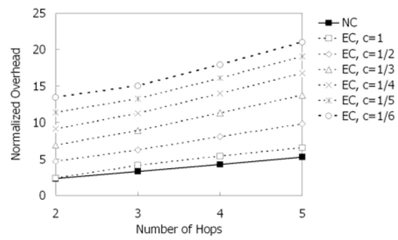

Figure 15: Overhead in the grid topology.

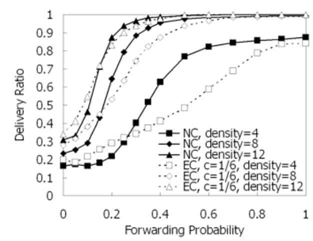

Figure 16: Delivery ratio in the random topology.

2.4.9 Routing Protocols for Disruption-Tolerant Networks

We Introduced the first message delivery mechanism that enables distribution/dissemination of messages in an internet connecting heterogeneous networks and prone to disruptions in connectivity. We call our protocol MeDeHa (pronounced "medea") for Message Delivery in Heterogeneous, Disruption-prone Networks. MeDeHa is complementary to the IRTF's Bundle Architecture: while the Bundle Architecture provides storage above the transport layer in order to enable interoperability among networks that support different types of transport layers, MeDeHa stores data at the link layer addressing heterogeneity at lower layers (e.g., when intermediate nodes do not support higher-layer protocols). MeDeHa also takes advantage of network heterogeneity (e.g., nodes supporting more than one network) to improve message delivery. For example, in the case of IEEE 802.11 networks, we use both infrastructure- and ad hoc modes to deliver data to otherwise unavailable destinations. Another important feature of MeDeHa is that there is no need to deploy special-purpose nodes such as message ferries, data mules, or throwboxes in order to relay data to intended destinations, or to connect to the backbone network wherever infrastructure is available. The network is able to store data destined to temporarily unavailable nodes for some time depending upon available storage as well as quality-of-service issues such as delivery delay bounds imposed by the application.

2.4.10 New Routing Frameworks for Scalable MANETs

The price, performance, and form factors of processors, radios and storage elements are such that mobile ad hoc networks (MANETs) can finally support distributed applications on the move. These applications (e.g., disaster relief) require point-to-point and many-to-many communication, and very few destinations or groups are such that a large percentage of the nodes in the network have interest in them. These application requirements are in stark contrast with the way in which today's MANET routing protocols operate. First, they have been tailored to support either unicast routing or multicast routing. Hence, supporting both point-to-point and many-to-many communication in a MANET involves running a unicast and a multicast routing protocol in parallel, which is very inefficient from the standpoint of bandwidth utilization. Second, the proactive and on-demand routing protocols for unicasting and multicasting proposed to date are such that the network is flooded frequently with link-state updates, distance updates, route requests, or multicast updates. This is the case even when the protocols maintain routing information on-demand (e.g., AODV and ODMRP).

This part of our research in DAWN has focused on investigating a new theory of routing applicable to scalable MANETs. A key motivation of our work are our recent results on the capacity of wireless networks. During the past year, we have investigated three ways in which routing can be attained in very large MANETs:

- Exploiting the natural temporal and spatial locality of reference of information flows.
- Integrating addressing and routing in a way that eliminates most of the flooding requirements for route signaling.
- Exploring mechanisms to allow nodes to establish correct orderings among them in order to obtain as many viable routes as possible to intended destinations.

Interest-Driven Approach to Integrated Unicast and Multicast Routing

There have been a large number of routing protocols proposed and implemented to date for MANETs, and many proposals have been made to reduce the communication overhead incurred in updating routing tables. These approaches include organizing the network into clusters, reducing the rate at which signaling packets propagate away from the origin of an update, and using geo-location information to direct signaling packets to particular regions of the network. A careful review of the prior work reveals that (a) existing routing protocols for MANETs support either unicast routing or multicast routing, and (b) the dissemination of signaling traffic in MANETs is not closely linked to the interest that nodes have on destinations, and is structured as either strictly on-demand, strictly proactive, or the use of both types of signaling by dividing the network into zones.

Over the past year, we have investigated the development of a unifying framework for all types of routing, and unicast and multicast routing in particular, and the use of interest in certain destinations (nodes, groups of nodes, roles, information objects, etc.) as the means to reduce signaling overhead. Our research has resulted in the design, verification and analysis of two new protocols, which we describe next.

Hydra: We first developed the Hydra multicast routing protocol [84], which was motivated by the desirability of providing the best features from the two alternatives in the existing design space for multicast routing in MANETs proposed to date. Whether multicast routing protocols for MANETs build multicast trees or meshes, all of them are based on network-wide dissemination of control packets to inform the rest of the nodes about the existence of multicast groups. In receiver-initiated schemes, only one node, which in many schemes is called the core of the group, originates the dissemination of information about a multicast group reaching all other nodes, and receivers send explicit requests towards the core to join the group. This approach was originally introduced in the core-based tree (CBT) [85] multicast routing protocol. In contrast, source-based or sender-initiated schemes have each multicast source originate the dissemination of state information that reaches all nodes in the network. This approach was originally introduced by Deering [86]. Given that the sender-initiated protocols proposed to date use per-source flooding, they do not scale well as the number of groups and sources increases. However, they can provide shortest paths from sources to destinations and avoid hot spots. On the other hand, core-based protocols incur far less overhead, but they do not establish shortest paths from sources to destinations, which leads to higher delays than the ideal shortest paths from sources to receivers.

Hydra is the *first* sender-initiated multicast routing protocol that incurs the same order overhead as receiver-initiated approaches, and yet establishes multicast meshes that approximate routing structures containing the shortest paths from multicast sources to their destinations. Hydra creates a multicast mesh formed by a mixture of source-specific and shared sub-trees (or sub-meshes) using as few control packets as receiver-initiated schemes. The key ideas behind Hydra are: Restricting the dissemination of control packets to those regions of the network where other dynamically designated sender has previously discovered receivers, aggregating control messages from non-core senders, and electing a sender as the core in non-destructive manner. We have proven that the algorithm used in Hydra to perform multicast-state aggregation is correct (i.e., it establishes loop-free routes from sources to receivers). We used simulation experiments to compare Hydra's performance with that of ODMRP [87], which is the existing proposal for multicast routing in the IETF, by considering different numbers of sources, group sizes, node density and the use of 802.11 and TDMA as the underlying MAC protocol. The results demonstrate the performance benefits that should be expected from the approach implemented in Hydra, and that Hydra provides substantial performance improvements over ODMRP even in scenarios involving relatively small networks with few multicast sources. Hydra attains the same or better delivery ratios than ODMRP, and does so while transmitting from one third and one half the number of data packets sent in ODMRP and incurring end-to-end delays that are close to an order of magnitude smaller than in ODMRP.

We presented our results on Hydra at the IEEE MASS 2008 conference, and the paper describing our research received the Best Paper award at the conference.

PRIME: Based on our success with Hydra, we studied the integration of unicast and multicast routing a single protocol, and developed the Protocol for Routing in Interest-defined Mesh Enclaves (PRIME) [88]. PRIME establishes and maintains a routing mesh for each active multicast group, i.e., for each group with active sources and receivers and for each unicast destination with at least one active source. The first source that becomes active for a given unicast or multicast destination sends its first data packet piggybacked in a Mesh Request (MR) packet that is flooded up to a horizon threshold. If the interest expressed by the source spans more than the single data packet, the intended receiver(s) of a MR will establish and maintain a routing mesh spanning the active sources and the destination (a single node in the case of unicast and a set of nodes in the case of multicast). In the case of a multicast flow, the receivers of the multicast group run a distributed election using Mesh Announcement (MA) packets to elect a core for the group, which is the only receiver that continues to generate MAs for the group. No such election is needed for a unicast destination. An elected core or unicast destination continues sending MAs with monotonically increasing sequence numbers for as long as there is at least one active source interested in it. When no active sources are detected for a flow, the destination or core of the flow stops generating MAs, which causes the routing information corresponding to the mesh of the flow to be deleted. To save bandwidth, MAs for different unicast and multicast flows are grouped opportunistically in signaling packets. Furthermore, to confine control traffic to those portions of the network that need the information, an enclave (or region of interest) is defined for an established mesh. The enclave of a flow is a connected component of the network spanning all the receivers and sources of the flow and the relay nodes needed to connect them. The frequency with which MAs for a given flow are sent within an enclave is much higher than the frequency with which MAs are sent for a flow outside it, and depending on the flow type (e.g., bidirectional unicast or multicast) MAs are not propagated outside enclaves.

We compared the performance of PRIME against traditional routing protocols for unicast and multicast routing in MANETs using simulation experiments. Our comparison addressed the performance of the protocols purely for multicast routing, and their performance in supporting unicast and multicast routing. We compared PRIME with ODMRP [87] to determine PRIME's effectiveness as just a multicast routing protocol, and consider different numbers of sources, groups, node densities and the use of group and random waypoint mobility models. We also compared PRIME against the use of AODV and ODMRP, and the use of OLSR and ODMRP. The results show that PRIME is a very efficient multicast routing protocol and provides substantial performance improvements over the traditional approach to supporting unicast and multicast routing. PRIME attains similar or better delivery ratios and significantly lower delays and communication overhead than the traditional approaches.

The results of our research on PRIME are presented at the IEEE ICNP 2008 conference [88].

Integrated Addressing and Routing for Scalable MANETs

We studied an approach to attaining scale-fre routing that complements the use of interest-based signaling and which consists of assigning addresses to nodes in a way that eliminates the need for most of the flooding incurred in the route signaling of a MANET.

The vast majority of these proposals are based on maintaining routing tables consisting of next hops to destinations that are labeled with unique position-independent identifiers, i.e., names (MAC addresses or IP addresses). The algorithms used to maintain these routes vary considerably, involve the exchange of different types of information (e.g., distances, paths, link states), and can be carried out proactively (e.g., OLSR [**?**], FSR [91], WRP

[90]) or on-demand (e.g., AODV [92], DSR [89]). However, as varied as these approaches have been, none of the common proactive or on-demand schemes are very efficient under all traffic load scenarios. At least in part, this is due to the fact that the destination identifiers used in routing tables are position independent and are assigned statically. Accordingly, considerable signaling overhead is incurred to either notify each network node in order to update the node with the updated state of a link or destination, or find a destination for whom there are sources with data to be sent. Interestingly, while hashing approaches and schemes based on position-dependent identifiers have both been proposed in the past for routing in MANETs, no solution has been proposed that combines using positional labels with distributed hashing to eliminate most of the signaling overhead incurred in MANET routing.

During the past year, we developed the Positional Routing Over Searched Elements (PROSE) protocol as an example of routing in MANETs based on the integrated use of distributed hashing and position-dependent identifiers for destinations. PROSE uses three simple mechanisms to establish routing to destinations incurring signaling overhead that is sub-linear with the number of nodes, and optimizations aimed at reducing the stretch of the resulting routes compared with the shortest routes that an optimal routing gene would attain.

In PROSE, nodes transmit neighbor-to-neighbor messages periodically. The dissemination of these messages elects a node as the origin of the positional labeling space, and builds a labeled directed acyclic graph (DAG) rooted at the elected origin, with each node receiving a positional label over the DAG. These positional labels are very similar to those proposed by Levine and Garcia-Luna-Aceves [93] in the context of Internet multicasting, and which are in turn derived from interval routing [94]. In PROSE, however, nodes elect their labeling origin, and belong to an DAG rather than just a labeled tree. Each node hashes its own name (e.g., MAC or IP address) to the positional label of an "anchor", and publishes its mapping with the anchor. The anchor of a node is then a node or nodes that have the closest positional label(s) to the positional label of the node.

To contact a destination, a source simply hashes the name of the destination to obtain the positional label of the anchor, and contacts the anchor for the positional label of the destination. Once the source obtains the positional label of the destination, the route from source to destination is implicit in their positional labels. The hashing algorithm to be used should render a fair spreading of position-independent identifiers to the positional-label space. The anchoring function serves the role of a distributed hash table (DHT) in PROSE, and given that positional labels are assigned over a DAG that is created in direct correlation with the underlying topology, the DHT used in PROSE is directly related to the actual topology of the MANET. Nodes also store the positional labels of all its neighboring nodes, which provides additional implicit paths.

PROSE supports the routing of packets among network nodes in three separate phases. First, during its *substrate building* phase, PROSE constructs a routing substrate in the form of a directed acyclic graph (DAG) that is rooted at a distributively elected node in the network. As the DAG is built, each node in the DAG is assigned a *positional label* that unambiguously denotes the position of the node in the DAG with respect to this root. The routing substrate (DAG) is updated periodically.

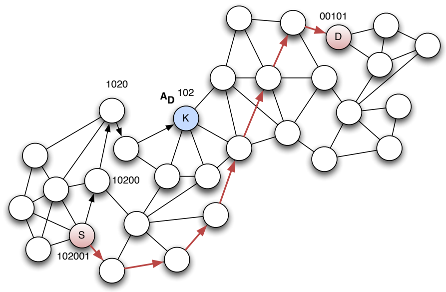

Figure 17: Overview

Each network node has a unique position-independent identifier (e.g., a MAC address or an IP address), which we call *Globally Unique Identifier* (GID). In the second phase, which we call the *substrate building* phase, each node in the DAG identifies its *anchor* node, which then is responsible for storing the mapping between the node's GID and the positional label of the node. A node determines its anchor node by using a consistent hashing function that takes as input the GID of the node and returns a positional label. The node updates its anchor with the mapping from its GID to its positional label periodically, and these updates are routed using the implicit route derived from the positional labels of the node and its anchor.

In the third phase, which we call the *destination discovery* phase, nodes requesting a route to a particular destination compute the anchor node for the destination and ask such an anchor for the most recent mapping of the GID of the destination to the positional label of the destination. These messages take advantage of the routing substrate and route implicitly to the anchor nodes. Upon reception of the mapping, messages are sent directly to the destination node using the implicit routes derived from its positional label and the positional label of the destination.

Note that the positional labels are not nodes but positions in the network. As the topology of the network changes over time, the positions remain in the network, provided the labels are consistently maintained. The labeled DAG serves as a *'churn-tolerant'* resilient structure, that defaults to a node with the closest positional label, when a request for an exact match fails.

The routing algorithm that routes over the labeled DAG, follows a greedy strategy and in the occurrance of a local minima, chooses the next hop with the lowest GID. Since the labels are positional the choice of GID does not affect the path to the destination.

To illustrate the performance of PROSE, we simulated the protocol using the Qualnet simulator to compare the performance of PROSE against that of AODV and OLSR. We chose these protocols because they have served as the benchmark for popular proactive, reactive and hybrid routing schemes. The total number of nodes in each of these simulations are 500 and the number of active flows was set to 250 flows. The interval of each of these simulations were distributed exponentially with mean equal to 1/20th of the duration of simulation. Each protocol was also run at 3 separate pause times from static to continuously mobile scenarios. The performance of all protocols is observed for different mobility rates with increasing pause times from 0 to the duration of the simulation. The duration of the simulation is set as 1200s. n the second scenario we show the overhead incurred while observing scaling characteristics of the protocol in scenarios containing different number of nodes starting with 50 nodes to 500 nodes and in each of these setups, the total number of exponentially distributed flows is set at half the number of total nodes in the network. Also to further demonstrate the worst case scenario, nodes move with 0 pause time. The duration of each of the experiment is set to 1200s. In both scenarios the simulations were run with 10 random seeds to ensure that the results are not an artifact of the scenario. The area of the network is a 600m X 900m region with nodes randomly placed across it. Node mobility is configured using the 'Random Waypoint' mobility model in which a node selects a point randomly and moves towards it at a uniformly distributed speed ranging from 1 to 10 m/s.

The results of the simulation experiments showed that delivery ratios of all the protocols decrease with increasing mobility. When the nodes are static, the delivery ratio is almost 100% for some protocols. As the nodes begin to move around, the delivery ratio drops. However, note that the delivery ratio of PROSE drops the slowest and manages the highest ratio among both reactive and proactive routing protocols. The usage of anchors allows routes to converge in places where other protocols failed to deliver successfully. In addition to this, the control overhead incurred in setting up the anchors and periodic updates via a clamped hello message, is lesser than the ones incurred by other protocols. As the number of flows in the network increase, control traffic for on-demand schemes starts to increase at a much faster rate while the proactive protocols start stabilizing at these regions. The signaling overhead in PROSE, owing to its clamped hello messages, manages to grow slower than the proactive protocols as well.

We have submitted our work on PROSE for publication to an IEEE conference.

Multi-dimensional Routing

To date, most routing protocols order the nodes in the network with respect to the destination and a sequence number of time stamp is used to validate the ordering of the nodes. While it is intuitive that the shortest path, in terms of number of hops, may not be the best path, routing protocols have not evolved beyond this dependence. Parameters such as available bandwidth and mobility can affect the quality and longevity of a path and by factoring such metrics into the routing decisions, routing protocols can take the next step in its evolution:*multidimensional ordering.*

In this part of our research, we address the concept of multi-dimensional ordering for routing environments (MORE) in an effort to improve routing protocols for ad-hoc wireless networks. Within this framework, flooding can be viewed as zero dimensional, since there is no ordering among the nodes, and distance vector protocols can be viewed as either one or two dimensional depending on whether they use a sequence number or not. The next step in the evolution of routing protocols is to go beyond the two dimensional ordering which dominates today's state of the art routing protocols.

The benefit of multidimensional ordering is twofold, it increases the quality of the resulting paths and the number of usable paths between the source and the destination. While the quality of route established has been studied in the context of of *quality of service* routing, maximizing the number of usable paths and the benefits thereof remains unexplored. Intuitively, the more paths available, the more resilient the resulting routing and in this paper we focus on the use of multidimensional ordering to increase the number of usable paths to the destination by including and possibly prioritizing paths which would not be used by traditional distance vector routing protocols. Increasing the dimensionality of a routing protocol can increase the number of usable paths, but the selection of the metrics used for the dimensions is of paramount importance. These metrics should allow the creation of a meaningful, loop-free ordering between the source and destination.

Routing, in its simplest form, involves the flooding of data packets throughout the entire network. No routing metric or any routing information is necessary in such protocols, but such flooding can be a very inefficient use of network resources. In many popular on-demand unicast routing protocols, flooding is used but only in the route discovery phase of the protocol and proactive routing protocols are even more dependent on the global dissemination of topology information.

To increase efficiency, modern routing protocols use a metric such as distance vector or link state to order the nodes into a directed acyclic graph (DAG) and unicast routing decisions are based on this DAG. The resulting ordering can be called one-dimensional since only one metric is used to order the nodes. Flooding, on the other hand, can be viewed as zero-dimensional since no routing metric is used. In an effort to provide loop freedom, most contemporary routing protocols use a time stamp, or sequence number, to differentiate up-to-date to information from outdated information.

We define the *dimensionality* of a routing protocol as the number of metrics used to distinguish nodes from each other. There is a significant improvement when moving from zero-dimensional routing to one-dimensional routing and another step between one-dimensional and two dimensional routing. We will explore the possibility of further improvements by increasing the dimensionality of routing protocols beyond the second dimension, in a framework called Multi-Dimensional Ordering for Routing Environments.

We define the *degree* of an ordering, denoted δ, to be the average size (geometric mean) of the partitions resulting from the metric used in that ordering. The degree of an ordering is indicative of the extent to which the nodes can be uniquely ordered in that dimension and thus the quality of the ordering, with a smaller degree representing a better ordering.

Definition: If an ordering partitions the N nodes of a network into n disjoint sets, with the cardinality of the i^{th} partition being x_i, the *degree* of that ordering is given by:

$$\delta = \sqrt[n]{\prod_{i=1}^{n} x_i}$$

For example, consider the seven node network shown in Figs. 18 to 2.4.10 , which show four different orderings. The degree of ordering A in Fig. 18 is $\sqrt[3]{2*3*2} = 2.29$. The same is true for ordering B, and Ordering C has a degree of $\sqrt[2]{6*1} = 2.45$. Ordering D, however, has a degree of 1, which is the minimum value and therefore the best possible ordering. The figures also show a successor relation and thus the relationship between the degree of an ordering and the size of successor relationships: for a given topology, the smaller the degree the greater the average number of successors per node. The reason for this comes from the fact that a node cannot make an adjacent node its successor if they have the same value in the ordering. Consequently every link between nodes with the same value can be considered a wasted link since it cannot be used in that particular ordering. Using ordering D, the maximum number of successor relations is achieved. In a DAG in which each node has a different value, every link can be used in a successor/predecessor relationship.

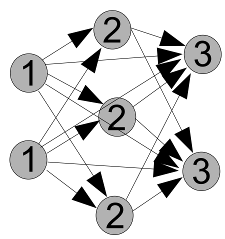

Figure 18: Ordering A

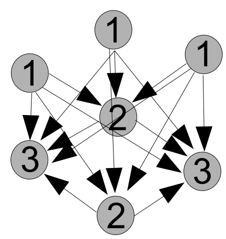

Figure 19: Ordering B.

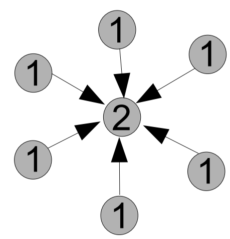

Figure 20: Ordering C.

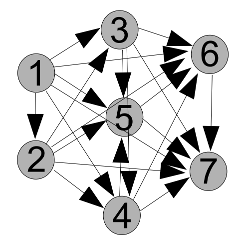

Figure 21: Ordering D.

While the degree of an order does not give an exact number of paths created from the source to the destination, it is sufficiently useful for comparisons between the orderings established by different routing protocols. Ordering with a small degrees are desirable because they facilitate route repair with minimal overhead.

In a mobile environment, link failures are common as nodes move out of transmission range of each other. In the next section we will study ways to maximize, or at least increase, the number of successor relations by increasing the number of orderings used.

To show that multi-dinemsional ordering can be used to improve routing in MANETs, even with simple metrics, we developed . the CaSH routing protocol, a simple distance vector routing protocol. CaSH resembles other distance vector protocols such as AODV, but differs in that it uses a four-dimensional ordering instead of two.

The signaling in CaSH is hybrid in that paths are established on-demand and are maintained proactively to achieve routing which incurs both low overhead and low latency. Route Request packets (RREQ) are used in a similar manner to that of AODV in setting up routes to destination nodes. As the RREQ propagates through the network, the nodes determine and store the distance to the source in their routing table. The nodes use an incremental sequence number to distinguish new information from older, and possibly outdated, information. Once a destination node receives a RREQ it increases its sequence number and initiates a Route Reply packet (RREP). This RREP packet then propagates through the network allowing nodes to determine their shortest path distance to the destination. Each RREQ and RREP has a unique source ID-sequence number combination which allows nodes to determine if they have seen a packet before and ensures that no packet is processed more than once at any node in the network. If a node no longer has a route to the destination it will initiate a local Route Error packet (RERR). Its neighbors, upon receiving this RERR, will update their routing table. If they have a route to the destination then nothing further is done, if they do not have a route to the destination then they issue a RERR. If RERRs propagates to the source, then there is no known path from the source to the destination and the source increments its sequence number and sends a new RREQ to the destination. After receiving the initial RREQ, the destination node will proactively flood the network with periodic RREPs and the source will do the same. This results in nodes maintaining up-to-date distance information to both source and destination as long as the path is needed. Once there is no longer need for the link between the source destination pair the proactive updates will cease.

Table 1: Simulation Results

	Scenario A			Secnario B		
	Delivery Ratio	Latency	Net Load	Delivery Ratio	Latency	Net Load
AODV	0.600±0.102	0.086±0.037	14.4±5.3	0.901 ± 0.031	0.072 ± 0.015	5.04 ± 1.31
DSR	0.142±0.099	18.47±15.91	5.02±1.24	0.1435 ± 0.04	42.69 ± 12.89	2.65 ± 0.34
OLSR	0.300±0.081	0.072±0.015	67.46±1.2	0.714 ± 0.044	0.104 ± 0.021	17.21 ± 0.16
CaSH	0.782±0.103	0.147±0.104	7.9 ±2.7	0.975 ± 0.031	0.067 ± 0.047	1.92 ± 0.20

To gauge the performance of the CaSH routing protocol, we implemented in the Qualnet 3.9.5 simulator and its performance was compared to that of several well known routing protocols, representing each of the main approaches to routing in MANETs. DSR is representative of source routing schemes, OLSR of proactive routing and AODV of on-demand routing. The implementations of DSR, OLSR and AODV used in the simulations are the versions built into the Qualnet simulator. Two scenarios were used in the simulations. The first of which was designed to rigorously test the performance of the protocols in a dynamic environment with volatile links. The second scenario uses a greater radio range to add more stability to the links and create more multi-path opportunities, for which the CaSH routing protocol was designed to exploit.

The first, Scenario A, comprised of 100 nodes uniformly distributed in a grid of size 1000m x 1000m with the transmission range of the radio set to 150m. This choice of parameters satisfies the minimum standards for rigorous MANET protocol evaluation as prescribed in [95] as it results in an *average shortest path hop count* [95] of 4.03 and *average network partitioning* [95] of 3.9%. Other relevant simulation parameters are summarized in Table 2 and Table 3. This would ensure that packets should travel several hops from source to the destination and thus would test the robustness of the protocols. The mobility model chosen was that of random waypoint with minimum speed of 1m/s and maximum speed of 10m/s with a pause time of 20s. Each experiment lasted for 900s.

Ten nodes were chosen at random to be sources of CBR flows and ten nodes were chosen at random to be destination of these flows. Care was taken to avoid the case where a node was both the source and destination to any particular flow. There were no restrictions on nodes being multiple sources or multiple destination or a source to one flow and a destination of another. Each source would send a maximum of 800 packets of size 512 Bytes at a rate of 4 packets per second. The start time of each flow was randomly determined using a uniform distribution and was within the duration of the experiment.

Using a time based seed, 250 random scenarios were generated with the above specifications and the results were used to compare the performance of the protocols. The large

Parameter	Value
Simuation time	900s
Number of Nodes	100
Simulation Area	1000m x 1000m
Node Placement	Uniform
Mobility Model	Random Waypoint
Min-Max Speed	1-10m/s
Pause time	30s
Propagation model	Two-ray
Physical layer	802.11
Antenna model	Omnidirectional
MAC Protocol	802.11 DCF
Data Source	CBR
Number of packets per flow	800
Packet rate	4 packets per second
Node density	0.001 nodes/m^2

Table 2: Simulation Parameters

Parameter	Scenario A	Scenario B
Network diameter	9.4 hops	7.07
Neighbour count	7.06 nodes	12.56 nodes
Transmission range	150 m	200m

Table 3: Varied Simulation Parameters

number of randomly generated scenarios were used to avoid bias in the results. Each series of random numbers was generated using the *lrand48* command in C and using the current time as a seed for the random number generator.

The average number of neighbors is 7.06 nodes. It is especially important than nodes have several neighbors as the CaSH routing protocol was designed for such environments as it thrives of the existence of multiple paths from the source to the destination.

Scenario B, was the same except with transmission range was increased to 200m. The purpose of this was to increase the average neighbor count from 7.06 to 12.6 nodes. Also, the increased range will make some of the links more stable as nodes will take longer to move out of range of each other. Also, the number of flows was increased to 30 flows in order to increase the load in the network. The value of the average shortest path would certainly be less than 4 nodes will the average network partition would be less than 5%. The differences between Scenario A and Scenario B are summarized in Table 3.

The mean and a 95 percent confidence interval were given. The simulation results for the four tested routed protocols are summarized in Table 1.

Three metrics were used to evaluate and compare the performance of the protocols. Delivery ratio is the fraction of packets that arrive at the corresponding destination by the end of the simulation. Latency is the average end-to-end delay each data packet experienced. Net load is the number of overhead packets (RREQs, RREPs, RERRs, Hellos, etc.) which were initiated or forwarded divided by the number of data packets sent. This takes into account packets that were sent into the network and were dropped or did not make it to the destination for any reason. This last metric gives an indication of the number of overhead packets needed to send a packet from the source to the destination.

In both scenarios, the CaSH routing protocol delivers a higher number of packets. The poor delivery ratio of DSR and OLSR in large networks clearly shows that these protocols are not scalable. Packets are dropped when there is no longer a know forward route to the destination. AODV sets up only a single route while CaSH sets up multiple routes thus allowing greater resilience to link failure and this is reflected in scenario A. There is a considerable improvement in performance between scenarios A and B since links are more stable and fewer packets are dropped due to lack of a route to the destination. Also, CaSH set up even more paths to the destination and achieve a near perfect delivery ratio. There is a significant difference in the performance of AODV and CaSH versus that of DSR and OLSR. In the CaSH there are two interesting contributors to the value of the latency experienced. On one hand is an alternate path exists the time taken to perform local route repair is negligible. On the other hand, if such a path does not exist there will be some backtracking during which time nodes may discover broken links as they try different neighbors. In the case where there is no longer a known valid path from the source to the destination, it will take a significantly longer time for the source to discover this than in AODV. Consequently, in a volatile environment as in Scenario A, CaSH experiences a slightly higher latency while in a more stable environment, Scenario B, CaSH enjoys a performance advantage over AODV. The most remarkable performance result was in terms of network overhead. This is the number of non-data transmissions (including forwarding RREQ, REEPs and hello messages). Even though CaSH employs some proactive signaling, it incurs considerably lower overhead than the fully reactive AODV and DSR routing protocols. This might be influenced by the lifetime of valid links. If the expected lifetime of a link is small, then reactive routing protocols would have to frequently flood the network as thy search for a path and CaSH demonstrates that in such scenarios it would be a better idea to introduce some proactive path maintenance. Also, the fact that the proactive updates in CaSH are restricted to a subset of nodes contributes to its relatively low overhead. Furthermore, the feedback mechanism employed in CaSH would allow the nodes to find and optimal period for proactive updates. It is not surprising that OLSR experienced the greatest overhead as it is a fully proactive routing protocol.

2.5 Task 4: Energy-Efficient Networks

In this task of DAWN, we have focused on developing Prawn (Prototyping Architecture for Wireless Networks), a tool for rapid prototyping communication protocols over 802.11 networks, and on developing the Santa Cruz mObile Radio Platform for Indoor and Outdoor Networks (SCORPION).

2.5.1 PRAWN

Prawn (Prototyping Architecture for Wireless Networks) is a tool for *rapid prototyping* communication protocols over IEEE 802.11 networks. Prawn provides a software environment that makes prototyping as quick, easy, and effortless as possible and thus allows researchers to conduct both functional assessment and performance evaluation as an integral part of the protocol design process. Since Prawn runs on real IEEE 802.11 nodes, prototypes can be evaluated and adjusted under realistic conditions. Once the prototype has been extensively tested and thoroughly validated, and its functional design tuned accordingly, it is then ready for implementation. Prawn facilitates prototype development by providing: (i) a set of building blocks that implement common functions needed by a wide range of wireless protocols (e.g., neighbor discovery, link quality assessment, message transmission and reception), and (ii) an API that allows protocol designers to access Prawn primitives. We show through a number of case studies how Prawn supports prototyping as part of protocol design and, as a result of enabling deployment and testing under real-world scenarios, how Prawn provides useful feedback on protocol operation and performance.

We also proposed Prawn as an effective tool for teaching wireless networking classes with a hands-on focus; through its API and library, Prawn provides an abstraction from lower-level implementation details, allowing students who are not necessarily skilled programmers, to quickly instantiate basic networking concepts and functions onto prototypes. Students can then run and observe the behavior of the resulting prototypes in a variety of real-world scenarios. Prawn then makes a number of important contributions to networking education, namely: it allows students to quickly and easily put theoretical and abstract concepts into practice, consolidating learning. It also provides students with much needed hands-on experience early in their academic career; additionally, it motivates learning as it makes the classroom experience considerably more enjoyable.

2.5.2 SCORPION

The Santa Cruz mObile Radio Platform for Indoor and Outdoor Networks (SCORPION) is a heterogeneous wireless networking testbed that includes a variety of nodes ranging from ground as well as autonomous aerial devices. Node diversity in terms of mobility and capabilities (e.g., processing, storage, and communication) makes SCORPION well- suited for testing and evaluating a variety of wireless network protocols, including multi-radio, multi-channel medium access control, multi-hop wireless ad-hoc routing, as well as disruption-tolerant routing and message delivery protocols for networks with varying connectivity.

In its current implementation, SCORPION includes nodes outfitted with three different types of vehicles, namely: (1) iRobot Create ground robots (identical to a Roomba without the vacuum), (2) RC airplanes equipped with Paparazzi autopilot software and hardware, and (3) a self-stabalizing remote-controlled helicopter.

Additionally, two non-autonomous mobile nodes complement the testbed, one that will be carried by people in order to mimic human mobility and another that will be installed in buses. More specifically, the bus nodes will be deployed in the UC Santa Cruz campus shuttle bus network.

The testbed currently has four airplane nodes, four helicopter nodes, 20 iRobot Create nodes, 40 human carried nodes, and 40 bus nodes.

The varied mobility patterns exhibited by SCORPION nodes allow for unique and innovative ways to test network protocols for current as well as next-generation network applications. SCORPION's management suite facilitates updating system and protocol software running on the nodes as well as monitoring the current status of nodes. Conditions that the management tool reports include whether the node is up or down, what versions of the operating system, test software and protocols are running. Besides detailed per-node information, the management tool also provides a birds-eye view of the testbed showing all the nodes, their types, and status (e.g., on-off). The management tool provides scheduling capabilities allowing the tetsbed to be time-shared by multiple users by providing scheduling capabilities. One of our goals is to make SCORPION available to the networking research community at large.

2.6 Task 5: Large, High-Fidelity Simulations and Testing

Our work in this task of DAWN has focused on techniques and evaluation of platforms that improve the modeling and simulation/emulation of large-scale heterogeneous wireless networks. In the following subsections, three activities are summarized that have yielded significant results.

The first activity proposes a high-fidelity emulation platform that can faithfully profile power dissipation and capture its impact in order to accurately predict and compare performance of embedded networked sensor (ENS) systems. By utilizing the strength of physical test-beds and compensating for their inadequate support of simulating realistic operation environments, the proposed emulation platform addresses heterogeneity, scalability and power-awareness, which feature realistic ENS systems. The architecture of the emulation platform uses an event scheduler to channel the interaction between the simulated wireless network environment and the emulated sensor nodes. The former is specifically optimized for wireless networking simulation with high-fidelity models for the entire wireless network protocol stack and models unique to sensor networking scenarios, while the latter delivers cycle-precision simulation of sensor applications running on various processors and systems.

The second activity proposes an evaluation framework for vehicular networks to achieve accurate, scalable, flexible and repeatable performance studies through the utilization of a hybrid emulation test-bed and incorporation of high fidelity protocol and environment models. The proposed framework not only addresses the unique challenges of vehicular networks but also enables new types of network analyses via the capability of conducting application-centric evaluation. Compared to traditional network-centric evaluation, case studies show scenarios where network-level statistics do not clearly discriminate between the two routing protocols while significant performance differences were observed using the application-level metrics, for example, order of tens dB on PSNR improvement and 38.3% reduction on mean square root of error achieved by AODV over GPSR. A set of future research directions can be enabled with the proposed evaluation framework.

In the third activity, our work in environmental mobility (EM) is further enhanced by implementing EM models in a network simulator that provides an evaluation and analysis environment for studying the impact on the network protocol stack and applications. Our simulation based studies spanned multiple layers of the protocol stacks; including the impact on transport layer in a TCP flow for a wired-in-wireless network, the impact on network layer for routing protocols and finally, the impact on medium access layer for data rate adaptation algorithms. Based on the simulation results, we present a cross layer optimization scheme, called Recovery from Earlier Good State (REGS) that allows protocols to become aware of the underlying channel operation so that they are able to adequately utilize the durations when the channel conditions are good, thereby minimizing under-achievement of expected performance. Results show that REGS is capable of improving performance by up to 100%.

2.6.1 High-Fidelity Emulation for Energy-Aware Embedded Networked Sensor (ENS) Systems

Over the last few years, embedded networked sensor (ENS) systems have been widely explored for a diverse set of real world applications in both academia and industry. Compared with traditional wireless network infrastructure, ENS systems are distinguished by high heterogeneity, large scale, and extreme energy-sensitivity. Further, most ENS systems solely operate on batteries. Many components of ENS platform are designed and manufactured to be controllable by software, which, for example, allows dynamic voltage and frequency scaling (DVFS), or resource hibernation. Consequently, energy dissipation can vary significantly between nodes, applications, and from time to time. On the other hand, the actual capacity delivered by batteries during their lifetime is significantly shaped by load pattern. Accordingly, in order to accurately predict performance of real world applications built in ENS systems, the evaluation platform should address issues rising from high heterogeneity, large scale and extreme energy-sensitivity. Presently, traditional simulations are widely rejected for evaluation due to lack of fidelity. In contrast, physical test-beds are favored as system effects can be accurately captured. However, they are still not suitable for predicting behavior of ENS systems deployed for real world applications. Further, emerging energy-aware ENS systems add one more dimension of complexity to the evaluation process. ENS consists of three components: computing platform, communication module, and sensing subsystem. All components are software-adjustable in order to optimize some aspects of system performance, in particular, to minimize energy consumption and improve system lifetime. To meet this end, a broad range of system architectures and techniques have been proposed and implemented. For example, D. McIntire et al [73] propose real time energy monitoring mechanism to be integrated in ENS so that performance tuning can be achieved. Further, the level of power capacity of batteries can also be fed back to other components of ENS, which may subsequently adjust their behavior, for example, slowing down CPU frequency to lower power consumption or temporarily increasing transmission rate at cost of higher power dissipation but benefiting the lifetime of batteries. According to our observation, energy aware interaction mechanisms proposed for real world ENS systems are not adequately modeled in existing evaluation platforms. In this project, we propose an evaluation platform that can faithfully profile power dissipation and capture its impact in order to accurately predict and compare performance of ENS systems. Thanks to recent advance in micro-architecture emulation technology, we do not have to rely on physical test-beds to study power dissipation and performance. For example, G. Contreras et al [74] developed XTREM, power and performance simulator with cycle-precision for the Intel XScale core. B. Titzer et al [75] designed and developed Avrora, cycle-accurate and instruction-level simulator for Crossbow AVR/Mica2 motes. Although micro-processor emulators can provide the highest behavior and timing accuracy for individual sensor platform, accurate and efficient evaluation platform for entire ENS systems face more challenges. As we have summarized above, the evaluation platform has to address heterogeneity, scalability and power-awareness, which feature realistic ENS systems.

In the area of sensor network research, researchers largely prefer physical test-beds to conventional simulation-based study, which is attributed to the unique characteristics of ENS systems. First, in general, hardware platforms designed and manufactured for individual sensor nodes have very limited capacity in terms of computation, communication and power supply. Consequently, system effects in nodal platforms, e.g., processing delay, power dissipation pattern, battery characteristics and capacity, and the like, introduce significantly wide range of dynamics to the deployed ENS systems. By using physical test-beds, impact of all system effects can be inherently replicated. Second, for ENS systems, the applications, the OS and the hardware platform tend to be more closely integrated. And there are many more proprietary sensor nodal platforms than traditional wireless mobile computers. So it is desirable to have the evaluation environment interchangeable with the programming environment to avoid implementing test-only version applications for all ENS platforms. By using physical test-beds, the necessity for test-only implementations can be largely eliminated.

On the other hand, physical test-beds do not satisfy all requirements for sensor network studies. First, it can be extremely difficult to replicate or model realistic operation environment for ENS system performance study. Where ENS systems are to be deployed may be difficult to access, or may be undergoing constant and drastic change, e.g., battlefields, or may be experiencing a variety of dynamics due to an extended operation cycle, e.g., seasonal cycle in eco-environment. By using physical test-beds in lab environment per se, impact of environment effects, e.g., interference in radio transmission, or realistic patterns of sensing activities, can be largely overlooked, which results in biased performance prediction against real deployment. Second, it can be prohibitively expensive to build physical test-beds for some futuristic ENS systems.

Our new sensor network evaluation platform is aimed at utilizing the strength of physical test-beds and compensating for their inadequate support of simulating realistic operation environments. Specifically speaking, the evaluation platform shall deliver the following features in the order of priority. First, both system effects in nodal platforms and environment effects in real deployment can be evaluated for the purpose of studying ENS systems. Second, sensor software developed for physical nodal platforms can be used for evaluation with little or no modification. Third, the evaluation platform must employ scalable software architecture, which can exploit parallel operations abundant in ENS systems, since realistic sensor scenarios may grow both spatially, in terms of number of nodes, and temporally, in terms of duration of operation.

There are three main software components in our software architecture: the discrete event network simulation engine (DES), the SENQ event scheduler (SENQ), and the sensor node platform emulator (EMU).

The discrete event network simulation engine (DES) delivers three major functions. First, the entity performs as full-fledged discrete event simulator, which has an event queue, an event scheduler, and handlers for various types of events. Second, the entity is specifically optimized for wireless networking simulation with high-fidelity models for the entire wireless network protocol stack (e.g., radio propagation models, wireless PHY models, etc.). Third, the entity is integrated with models unique to sensor networking scenarios (e.g., sensing models, battery models, etc.). In short, all environment effects concerning wireless communication and sensing activities are simulated here.

The sensor node platform emulator (EMU) delivers cycle-precision simulation of sensor applications running on various processors and systems. The entity reproduces the accurate timings of all events as in a real physical platform and power measurements of the microprocessor.

The SENQ event scheduler (SENQ) channels the interaction between the simulated environment and the emulated sensor nodes. Individual sensor nodes, emulated by the EMU, produce global events, e.g., sending radio, retrieving sensor reading etc, which essentially change the environment and affect other sensor nodes. The SENQ is responsible for scheduling these events into the simulated environment, i.e. the DES, which, subsequently, updates the simulated environment and also uses the SENQ to inform all affected sensor nodes of relevant events, e.g., radio arrival.

2.6.2 High-Fidelity Application-Centric Evaluation Framework For Vehicular Networks

A key goal in our work is to propose an evaluation framework for vehicular networks to achieve accurate, scalable, flexible and repeatable performance studies through the utilization of a hybrid emulation test-bed and incorporation of high fidelity protocol and environment models. Based on the evaluation technique used, the past performance studies in the context of vehicular networks can be divided into two broad categories: measurements and simulation. Although physical test-beds allow real implementation to be tested, the resource investments needed to deploy a large-scale physical test bed make it prohibitive to be used as the tool to evaluate vehicular networks in a scalable matter. Further, it is difficult to provide repeatable experiments for a given input configuration via physical test-beds, particularly with diverse operating conditions like vehicles on a highway. While simulation offers a flexible and scalable approach, models adopted in simulation evaluation may fail to capture network deployment and vehicle movement patterns in a very faithful fashion. Further, simulation tools do not provide the flexibility of executing operational software (e.g., real application implementations). These observations indicate that in order to perform accurate, scalable, flexible and repeatable evaluation of vehicular networks, an evaluation framework that utilizes the relative benefits of both physical test-beds and simulation is required to be in place.

In this project, we use a hybrid emulation test-bed TWINE [75], which combines simulation, emulation and physical networks in an integrated test-bed for evaluation of wireless networks. Using TWINE, evaluation of large-scale vehicular networks can be performed in lab environment. Further, we incorporate into the framework high fidelity models of lower protocol layers and physical environments that are specific to vehicular networks (e.g., channel, mobility, deployment data), which provide realistic details as well as the flexibility to support diverse network conditions. The advantages of the proposed evaluation framework are manifested not only in the scalability and accuracy of the obtained results, but also in the types of analyses that are enabled. The latter part, referred to as application-centric evaluation in this paper, provides an understanding of the performance of vehicular networks at the application level, which is essential to improve the experience of end users, i.e. their satisfaction of perceived services provided by the underlying vehicular networks.

Evaluation Framework Design:

Our proposed evaluation framework is realized by a three-fold approach:

- A hybrid emulation test-bed TWINE is utilized to enable evaluation of vehicular networks in an accurate, scalable, repeatable and flexible manner. TWINE allows the execution of real applications in a simulated network environment, which effectively averts the difficulty in modeling applications by using operational applications.

- Objective application-level metrics are identified and used in the performance evaluation, which not only effectively capture the overall user perceived performance but also facilitate the development of practical approaches to measure them, as compared to the commonly used time and resources consuming perceptual evaluation.

- Realistic vehicular network environment settings are created by integrating into the evaluation framework high fidelity mobility traces and real deployment data.

Provided with the advantages of TWINE, our first approach is to move the vehicular network evaluation paradigm from simulation or physical test-beds to emulation-based evaluation. This emulation-based framework is integrated into our evaluation paradigm and tailored for vehicular network performance studies. More specifically, in order to assess application-level metrics, a client or end-host is emulated, i.e. being operational to run real applications that interface with the operating system. This client or end-host communicates with other hosts in the network using an emulated network interface. In addition to being able to run operational applications, realizing a client or end-host in emulation facilitates the capture of real time interactions between applications and lower layers. Simulation is used to model the rest of the network, the operation of which does not need to be modeled at the same level of fidelity as the emulated hosts, thus enabling evaluation of large-scale vehicular networks. When prototype or implementation of a specific network protocol or system is available, a subnet of operational nodes can be created through emulation, where the actual code can be evaluated. High-fidelity channel and host mobility models are integrated as well to create realistic environment settings for the vehicular network under investigation.

Objective application-level metrics refer to the set of measures, the estimation of which does not require an end user's subjective assessment but can be directly computed from statistics collected at the application layer. In order to expedite the evaluation process and improve efficiency, objective application-level metrics that closely reflect user-perceived performance are employed in our evaluation. For example, we propose the use of PSNR as the application-level objective metric for streaming video, which is highly correlated with the subjective discernment of humans, thus serves as a good candidate of measures of the user-perceived video quality. Our focus lies in developing methods so that the measurement of objective metrics such as PSNR can be effectively integrated into network performance studies.

In order to create realistic environment settings of vehicular networks, high fidelity models of vehicle movements as well as real deployment data of roadside APs, are integrated into our evaluation framework to enable meaningful analysis. To configure the target vehicular network in a realistic manner, real roadmaps of selected regions are used in our studies. Locations (e.g., GPS coordinates) of the currently deployed roadside APs are obtained through the AP locator website www.jiwire.com and used as the AP positions in the experiments. In order to produce close to reality vehicular movement patterns, we aim at integrating vehicular traffic simulators (e.g., CORSIM, TRANSIMS, VISSIM, MMTS) into the network simulator Qualnet. At the current stage, we have successfully incorporated vehicle mobility traces into our evaluation framework. The mobility traces are obtained from a detailed vehicular movement simulation over real road maps using MMTS. It contains a 24-hour movement pattern (coordinates, moving directions and speeds) of a total number of 259,978 vehicles in Switzerland (area of 41,559 km2). The traces are further parsed to obtain movement data for selected regions and scenarios (e.g., city, highway) and with desired vehicle density and speed.

To summarize, the proposed evaluation framework extends the existing paradigms by emulating components of a vehicular network, in an environment that captures with high fidelity mobility and topology aspects, and provides efficient metrics to analyze the networks in their application/user-perceived performance.

Research Directions Enabled by Application-Centric Evaluation Paradigm:

The proposed framework facilitates the investigation of cross-layer interactions across the protocol stack, which includes applications. This is demonstrated in our study by examining the effects of the interaction among video streaming application, two network-layer routing protocols and a MAC-layer rate adaptation protocol. The evaluation framework enables the study of possible correlation between application-level and network-level performance, which is crucial in identifying appropriate network-level metrics that are capable of accurately reflecting application layer performance. We conduct three case studies to showcase the possible research directions enabled by application-centric evaluation paradigm.

- Evaluate application design and implementation in the target vehicular network. Application-level performance is closely related to the implementation details of an application including application-layer adaptation, optimization, or parameter configuration. Through the proposed application-centric evaluation framework, the investigation of application design and implementation choices can be performed within the very context of the target vehicular network, instead of as a stand-alone process. Our results imply that adaptation at the application layer should take into account the properties of the underlying protocols, in this case the loss behavior of a routing protocol.

- Investigate cross-layer interactions across protocol stack including applications. Our results indicate that a lower layer protocol should be aware of the application requirements in determining its adaptation strategy. When the application requirements change, the protocol should adjust its behavior responsively.

- Explore correlation between application-level and network-level performance. Our results indicate that with the current set of network-level metrics, it is hard to represent the correlation between application-level and network-level performance in a quantitative manner. A new set of metrics, which examine the network-level performance in more detail than the first order, are required to be designed in order to effectively study the impact of network operation on end user experience.

Networking Studies Enabled by Evaluation Framework:

With the realization of the evaluation framework, real applications specific to vehicular networks can be executed directly such that network performance can be measured at the application layer using application-specific metrics. In this way, the extent to which the quality of service of various classes of applications (e.g., safety applications, commercial services) provided by the current generation of vehicular networks can be investigated effectively.

Evaluation of Video Streaming by PSNR with Different Routing and Rate Adaptation Protocols under Diverse Environment Conditions:

The experiments conducted in this study [77] investigate the performance of a video streaming application running in a vehicular network. The network scenario has vehicles moving on a freeway, which are equipped with wireless devices and thus able to communicate with each other. Along the freeway there are APs that serve as gateways to the Internet. The vehicles can connect with these APs, possibly over multi-hop routes, to access the Internet. The application used is a media player that displays, on a client device (e.g., laptop, PDA) in the vehicle, a streaming video from a server on the Internet. A MacBook Pro is used as the client machine, which runs VLC streaming video client application. The streaming server runs VLC in server mode. For the routing protocol, AODV and GPSR are used, as they represent two large classes of ad hoc routing protocols: reactive non-geographic and geographic with greedy forwarding. Both protocols are well documented, tested in many research studies and shown to exhibit excellent performance in their respective class of routing protocols. For MAC and PHY, IEEE 802.11b is used, in which the APs and the vehicles maintain a fixed transmission rate at 11Mbps, given the relatively high bandwidth requirement of video. The channel effects are modeled using two-ray path loss and Rayleigh fading model with varied maximum fading velocity. The results demonstrate that application-level metrics are more directly related to end user experience and therefore facilitate the understanding of the impact of the target vehicular network on the application performance. Another point illustrated by the results is that there exist cases where network-level metrics can bear little correlation with application-level performance. In general, the correlation of network-level and application-level performance relates closely to the specific implementation details of the application. This provides another argument to use application-level metrics in network performance analysis to obtain reliable results. By comparing the set of application-level metrics measured for video streaming, it is observed that PSNR constitutes the best candidate to effectively reflect the overall user-perceived video quality over time.

By comparing the Peak-Signal-to-Noise-Ratio (PSNR) of a video streaming application that is delivered by two routing protocols (AODV and GPSR) and their respective network-level performance, we were able to show that application-level metrics are more directly related to end user experience and provide reliable performance results. The study also highlights scenarios where although network-level statistics, including throughput, delay and jitter, do not clearly discriminate between the two routing protocols, significant performance differences were observed using application-level metrics, i.e. order of tens dB on PSNR improvement and 38.3% reduction on mean square root of error achieved by AODV over GPSR.

Evaluation of Multi-hop Relaying for Robust Vehicular Internet Access:

Internet connectivity in vehicular environments using WiFi devices depends on several factors, including: AP density and distribution, vehicle density, distribution and speed, and communication range of nodes (APs and vehicles). Having vehicles not directly connected to APs depend on other vehicles to relay packets, possibly over multiple hops using inter-vehicular communication, offers a seemingly better alternative to improve connectivity as it does not require high power transmissions nor force the use of lower bit-rates. In this study, our goal is to investigate the potential connectivity improvement from using multihop relaying via inter-vehicular communication as opposed to relying only on direct communication between APs and vehicles (referred henceforth direct access). We also study the effect of communication range for both strategies; this is in contrast to prior measurement studies, which focus only on one extreme setting of radio parameters, i.e., lowest bit-rate and maximum transmission power. To meet the above goals, we study spatial-temporal aspects of connectivity for direct access and multi-hop relaying strategies by analyzing real AP location data in conjunction with realistic vehicular mobility trace for a city scenario (in our case, we consider the city of Zurich, Switzerland). We conduct this study [6] independent of any specific vehicular network protocols and applications, but focusing on connectivity metrics like connection duration and percentage of vehicles connected, which are relevant for supporting any application. Our evaluation approach allows us to efficiently study connectivity characteristics of large-scale vehicular network scenarios with several thousands of vehicles and hundreds of APs.

Our study leads to the following key observations:

- Multi-hop relaying provides substantial gains in connectivity relative to direct access, with small number of relays sufficient to achieve most of this gain. The additional gain in connectivity from allowing additional hops tends to diminish after a few hops, and this gain is dependent on the communication range. For the scenarios we considered, going from direct access to two hop relaying provides the highest improvement in most cases (up to 152%).

- In terms of spatial connectivity (measured as percentage of vehicles connected), multi-hop relaying and direct access with increased communication range yield similar improvements. Combination of multi-hop relaying and increased communication range provides the most gain (up to 150%).

- With regard to temporal connectivity metrics (connection and disconnection durations), multi-hop relaying provides greater improvement compared to direct access with increased communication range. As with spatial connectivity, multi-hop relaying with increased communication range is the most effective strategy, which achieves gain as much as 467%.

- Spatial connectivity is unaffected by vehicle density, whereas connection duration improves with higher vehicle density.

Overall, our results show that multi-hop relaying strategy leads to substantial gains in connectivity, and that multi-hop relaying combined with increased communication range provides even greater gains. We have also found that relay paths with few hops are sufficient to realize most of the gain with multi-hop relaying. The focus of our on-going and future work is on understanding the data transfer performance with multi-hop relaying and on developing suite of effective protocols for enabling robust vehicular Internet access.

2.6.3 Performance Implication of Environment Mobility in Wireless Networks

Environment Mobility (EM) refers to the ambient motion of people, vehicles and other objects in the vicinity of wireless communication, in contrast to nodal mobility that accounts only for the effects of movement of the transmitting or receiving nodes. Our previous research on this channel phenomenon included, in the first phase, measurement and characterization of channel fading, followed by, in the subsequent phase, modeling of channel behavior. We have proposed a three knife-edge diffraction model to represent the RF propagation effects and a two-state Markov model of alternating channel quality states to represent the environmental mobility as a superimposition of channel fading states. These models were validated and parameterized against the measured data obtained in the earlier phase.

The models were subsequently implemented in a network simulator that provided an evaluation and analysis environment for studying the impact on the network protocol stack and applications. Having identified earlier that wireless links operating in ambient mobility (of people, vehicles etc) conditions are associated with transient shadowing dynamics wherein the link quality switches back and forth between good state and shadowed or lossy state. We conjectured that such behavior might have unfavorable implications for the suites of protocols that maintain state via feedback from the channel but are otherwise agnostic of the underlying operations. Our simulation based studies spanned multiple layers of the protocol stacks; including the impact on transport layer in a TCP flow for a wired-in-wireless network, the impact on network layer for routing protocols and finally, the impact on medium access layer for data rate adaptation algorithms. The simulation studies confirmed our hypothesis that the protocols learn dynamically about current link conditions and modify their states and operations accordingly. Although the protocols are quick to learn about the bad information, they are slow to discard it as the conditions improve. This is a reasonable mode of operation for fast fading channels, as has been shown by simulations and implementation studies that demonstrate improvements in protocol performance under those conditions, as the average channel conditions vary only very slowly. When such protocols however are operated in an EM channel, the link can switch to a good state while the protocol is still operating under the assumption of the bad state recorded in the recent past. Thus they are not able to adequately utilize the durations when the channel conditions are good, thereby leading to under-achievement of expected performance.

To address these performance bottlenecks, thus identified, we propose a cross-layer optimization technique called recovery from Earlier Good State (REGS) that requires the protocol state achieved while the channel was in shadowed state to be discarded on channel recovery. That is, the operation of the protocols while the channel is good is not influenced by the decision made or state variables changed when the channel was temporarily shadowed. The protocols implementing the REGS optimization technique require cross-layer information from lower layers regarding the time instances when a transition is made from one channel state to other. The link layer monitors the incoming packets, infers the current channel state from the received signal strength, and informs the higher layers as the channel switches from a good to bad state and vice-versa. The individual protocols can subscribe to these announcements. When a REGS enabled protocol learns of a switch to the shadowed channel duration (bad state), it checkpoints the current protocol state; subsequently when informed of the conclusion of this state, it discards all the protocol memory acquired since the previous announcement and starts to operate from the earlier check-pointed state. Such a scheme is not applicable for fast fading channels, which have been the primary focus of earlier research in this area, as the duration of the bad state is very small (hundreds of microseconds to few milliseconds). As such, for fast fading channels, by the time channel conditions are inferred and propagated to the relevant component in network, these conditions would already have changed. The EM channels, on the other hand, are more deterministic, at least at coarse time granularities, and therefore we can potentially gather more information about the channel. The REGS algorithm for inferring state boundaries maintains smoothed trace of signal strength, utilizes hidden Markov model training, and is able to predict state transitions without false positives or negatives at a latency of less than 5 milliseconds.

We applied REGS to TCP and the SampleRate algorithm and observed a significant performance improvement. These results are presented in the above figures. We also compared the performance due to REGS with existing optimization schemes for fast-fading or random channel fading. We discovered that as the magnitude of channel fading increasing in favor of EM compared to fast fading, the performance benefits of REGS overshadow those of the other optimizations.

A number of interesting questions on the impact of EM remain to be addressed. One interesting direction that we are currently exploring is the role of nodal mobility in relation with environmental mobility. Nodal mobility will distort the two-state model that we have presented, but to what extent and what will be the resulting impact on protocols remain to be seen. Furthermore, even though our diffraction models are generic, we have focused in our network study on the specific case of shadowing caused by human beings. We are also studying the impact of shadowing by objects with dimensions larger than humans - vehicles, for example.

Finally, we had made the observation of the possibility of mutual benefit when REGS operates together with other optimization schemes. A thorough and holistic investigation can be conducted to identify the necessary conditions for this synergy of different optimization techniques.

2.7 Task 6: Modeling and Control of Information Flow Patterns

In this task, we have studied several aspects of information flow patterns in networks. Our research addressed sensor networks and vehicular networks, and focuses on beamforming, fault detection, and epidemic dissemination. We examined the dissemination of patient medical records across a team of nurses connected via Bluetooth and the safe delivery of such information to the attending doctor and the central record repository. We study the delivery delay as a function of nurses' motion pattern. We also considered information dissemination in the urban grid of the environment data collected by vehicles (e.g., license plates, vehicle images, traffic congestion, pollution measurements etc); we further harvest the data of interest using "agents" (e.g., patrol cars). We proposed a bio-inspired multiple agent foraging motion model to effectively harvest the data. The agent motion model can be described as a continuous time Markov model where the agent moves from state to state with probabilities depending on the data found in a neighborhood. We show the effectiveness of the "bio-harvest" motion model by comparing with traditional models such as uniform search and random way-point search.

2.7.1 Beamforming and Fault Detection in Sensor Networks

Acoustic signals are wide band in the sense that the ratio of the bandwidth of an acoustic signal to the carrier frequency is very large; by comparison, normal communication channels are narrow band. The wide band of acoustic signals means than different frequencies in an acoustic signal are subject to different amplitude and phase distortion in the acoustic 'channel'. The problem considered in [81] is to adaptively process a moving speaker's signal recorded by multiple receivers (microphones) in such a way as to reinforce the signal and reduce interference from other sources. Thus in addition to in addition to beamforming the procedure must localize and track the speaker. This scheme was also reported in the 2007 Progress Report. The paper was revised and has now been published.

Transmit beamforming used in WiMax is centralized: a single processor controls the carrier phases of a multiple-antenna transmission. Reference [82] presents a novel algorithm for *distributed* transmit beamforming enabling multiple single-antenna nodes to simultaneously transmit a common message such that they constructively interfere at the receiver. In order to constructively interfere, the nodes iteratively estimate and adjust their carrier phases with the help of receiver feedback. The receiver's feedback is based on composite channel estimates calculated from two simultaneous transmissions initiated by the nodes at each iteration. It is shown that starting from arbitrary carrier phases, the algorithm converges to the maximum gain almost surely. The impact of phase estimation errors on the algorithmfs steady-state gain is assessed via simulation and shown to be robust to moderately large errors of $\pm 20°$. The causes of asynchrony in simultaneous message transmissions of IEEE 802.15.4 packets are analyzed and their contribution to phase estimation errors is shown via simulation to be within the acceptable range of $\pm 20°$.

Monitoring the health of a sensor network by detecting its failed sensors is essential to the reliable functioning of any sensor network. Reference [83] presents a distributed, online, sequential algorithm for detecting multiple faults in a sensor network. The algorithm works by detecting change points in the correlation statistics of neighboring sensors, requiring only neighbors to exchange information. The algorithm provides guarantees on detection delay and false alarm probability. This appears to be the first work to offer such guarantees for a multiple sensor network. Based on the performance guarantees, a tradeoff is computed between sensor node density, detection delay and energy consumption. Also addressed are issues of synchronization, finite storage and data quantization. The approach is validated with some example applications.

2.7.2 Patient Record Monitoring and Management using Bluetooth Dissemination

Remote medical monitoring using body sensors and wireless communications has been gaining attention recently because of the potential savings in patient care and the equally impressive enhancements of quality of care for mobile individuals. What makes remote medical monitoring feasible are the advances in the body sensor technology (non-intrusive sensors embedded in actuators to monitor vital signs); and in wireless technology (Body LAN, cellular and Wireless LAN). Currently, most medical vests and body LANs connect to the Internet in a point-to-point fashion, via the cellular system (say SMS). With the growing popularity of ubiquitous computing and opportunistic P2P personal networks, it makes sense to explore beyond the point to point health care paradigm and study new models for remote patient care that exploit P2P networking among patients and care providers (nurses, doctors, emergency personnel). In this project, we developed the first opportunistic medical network applications using Bluetooth.

One of the goals of this project was to demonstrate the feasibility and effectiveness of Bluetooth ad-hoc networking [70] and data dissemination of medical records from patients, which are equipped with lightweight medical monitoring system, to a central medical database [67]. We enhanced the demonstrated system and had a field test [66] to show that Bluetooth P2P networking is both feasible and cost-effective in remote medical monitoring. However, we found out the conventional Bluetooth-based patient monitoring system is not suitable for emergencies, mainly due to Bluetooth connection establishment takes 5 to 10 seconds and this delay can cause severe problems in medical environments. The proposed Enhanced Inquiry Response (EIR), a new feature that will be available in Bluetooth v2.1, based dissemination technique [65] allows us to broadcast data in the Bluetooth overlay without establishing connections, thus enabling emergency applications over Bluetooth. This was demonstrated only via simulation, since Bluetooth v2.1 devices are not yet available. The simulation results show the proposed system works well in the emergency situation and much faster than typical IR way.

2.7.3 Bio-inspired Mobility Models for Urban Harvesting

We are interested in urban sensing for effective monitoring of environmental conditions and social activities in urban areas using vehicular sensor networks (VSNs). We particularly envision *proactive* urban monitoring services where vehicles continuously monitor events from urban streets, maintain sensed data in their local storage, process them (e.g., recognizing license plate numbers), and route messages to vehicles in their vicinity to achieve a common goal (e.g., to allow police agents to pursue the movements of specific cars). For this, we have designed and implemented MobEyes, a novel middleware that supports VSN-based proactive urban monitoring applications [61]. In MobEyes, each sensor node performs event sensing, processing/classification of sensed data, and periodically generates meta-data with extracted features and context information tagged with timestamp and position information. Meta-data are then disseminated to other regular vehicles such that mobile agents, e.g., police patrolling cars, move and opportunistically harvest meta-data from neighbor vehicles. As a result, agents can create a low-cost opportunistic index which enables them to query the completely distributed sensed data storage, thus answering questions such as: which vehicles were in a given place at a given time? which route did a certain vehicle take in a given time interval?, and which vehicles collected and stored the data of interest?

In a typical urban scenario, multiple agents can collaborate in harvesting relevant data, processing them, and searching for key information. It is critical to design a strategy to effectively coordinate and geographically separate the operation of multiple agents, while allowing them to seek most productive fields in a totally distributed matter. However, multi-agent harvesting is a very challenging problem due to the dynamic nature of the target environment (e.g., continuous creation and relocation of metadata) and the scale of operations (e.g., harvesting region ranging over multiple city blocks) without a priori knowledge of the location of the critical information. Incidentally, we note that social animals (ranging from bacteria to vertebrates) solve a similar problem of *foraging* to find a good food source quite efficiently using a simple communication mechanism in a fully distributed

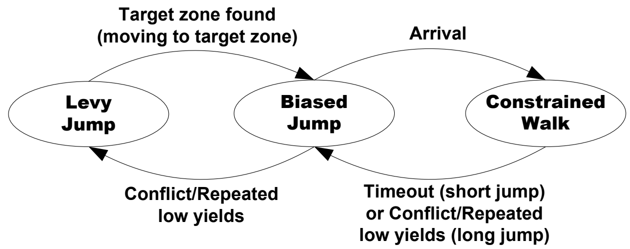

Figure 22: Agent state diagram

manner with lightweight and lazy coordination.

Given this observation, the primary goal of this task is to design a novel multi-agent mobility coordination mechanism inspired to biological systems. We realize that each species may have inched towards foraging optimality for specific tasks and various constraints (e.g., habitat niches, animal size and speed, environment, etc.). Therefore we design a mechanism by encompassing different animal foraging and behavioral ecology strategies, instead of focusing on single animal species. The natural scene examples inspiring MobEyes multi-agent coordination include: (a) Foraging behavior of *Escherichia (E.) coli* bacteria that operate in distinct modes of locomotion based on the level of nutrient concentration; (b) Lévy walk behavior of many biological organisms and groups, e.g., albatrosses and fishing boats, to improve food search over large-scale regions; and (c) Stigmergy found in ants and other social insects that use various types of pheromones to signal nest mates with potential conflicts, e.g., a sort of "no entry" sign. The key contributions of this task are a novel data harvesting mechanism called *datataxis* and the use of Lévy jump to search for areas of rich information.

In our design, MobEyes agents operate in one of the following three modes: (a) the Lévy Jump (LJ) mode, (b) the Biased Jump (BJ) mode, and (c) the Constrained Walk (CW) mode. The LJ/BJ modes are considered as the exploration stage to find the best possible location to start a more focused search, whereas the CW mode can be considered as the exploitation stage where agents try to harvest as much as possible by carefully and finely controlling their movements. Figure 22 presents a transition diagram consisting of the three possible states of operation by MobEyes harvesting agents.

First of all, a MobEyes agent starts with the LJ mode and searches for dense areas with vehicles. In the Lévy jump literature, it is known that the jump distance following a power law distribution with the exponent of 2 is known to be optimal for non-destructive foraging, i.e., a foraging scheme where agent can "productively" visit the same place many times [60]. Recall that since vehicles move in the urban grid, it may be very possible that after a while the same area may become "productive" again. The key idea of the LJ mode is that agents can choose a long distance with some probability, due to the heavy tail of the power law distribution. Thanks to the long jumps, the area covered by the agents will be much larger than the area that would have been covered by only random walk movement patterns [60]. Since the network size is finite in our model, we use a truncated Lévy jump distribution: $f(d) = \frac{d_{max}d_{min}}{d_{max}-d_{min}}\frac{1}{x^2}$ where we set the d_{max} as the network diameter and d_{min} as the communication range. The angle of a jump from the current location is selected randomly. For each jump, the agent steers its movement towards the road segment that minimizes the distance to the new jump location. However, for a given location, it may not be feasible to jump toward a certain direction. For instance, if an agent is located at the bottom left corner of the network, a jump is feasible toward the first quadrant. The key idea of a Lévy jump is to have a long jump with some probability for efficient exploration. Thus, we modify the angle selection such that we only consider the region that can span a chosen distance. In the previous example, the jump direction is chosen from the first quadrant. Once the agent finds a dense area above a certain threshold d_θ, the agent changes its operation state to the BJ mode so that it can move toward that location. The target location is the mid-point of the densest road segment, which is also set as the reference point of the CW mode that will be used by the agent as described below. The agent steers its movement towards the road segment that minimizes the distance to the determined reference point (i.e., a simple greedy movement).

When entering the CW region (the circular area with center the reference point and radius R), the agent switches its mode to the CW mode and starts harvesting metadata within that region. The default choice in MobEyes is to automatically set the distance parameter R as a function of the number of agents and the size of the overall search area. MobEyes supports two operating sub-modes for an agent in the CW state. First, the agent follows the road segment that maximizes the positive per-segment density change. In this case, since we exclude the current road segment from the candidate road segment for the next movement, it is possible that the rate change may be negative. If this occurs, the harvesting agent chooses the road segment that minimizes the change. Second, the agent can follow a biased random walk along a set of road segments in the vicinity; the set consists of the segments with density greater than a configurable threshold. If the explored urban area has the shape of a long strip, staying within a CW region could be inefficient. For this reason, the MobEyes agent periodically performs short range jumps to explore the nearby area after CW duration T_{cw}, thus changing its reference point. To avoid the worst case of continuous jumping around a region where there is not much gain, after a configurable time interval, the agent performs a long jump to a random direction, and switches its mode to the LJ mode to collect the density information again as in the initial phase. This behavior is repeated until the harvesting procedure has ended.

We have compared the above described, bio-inspired agent motion with other popular motion patterns (including random motion) for a representative scenario. The results support the superiority of the "data-taxis" strategy (see [62, 63]). In future work, we plan to analyze the characteristics of the bio inspired harvesting motion more formally. To start, we plan to investigate the influence to the underlying dissemination motion pattern on bio-harvesting motion itself. For example, if the sensor vehicles move according to a "commuter pattern" from home to work, the bio inspired harvesters motion will "follow that pattern" in the most efficient way, leading to a very different behavior than observed with randomly moving sensor vehicles. We will then superimpose several harvesting patterns aimed at different targets and we will examine inter-contact time statistical distributions (e.g., exponential tails vs power law tails). Further, we plan to explore the suitability of urban harvesting motion models to other applications beyond collection of vehicle sensor data. Scenarios to be explore include the search for partners in social activities, the shopping for goods in high demand and spotted distribution. Finally, we plan to compare the bio-harvest model with other motion models encountered in urban environments, from the commuter model to the police pursuit of criminals, to the impulsive shopper, or to the errands runner. This comparative analysis has multiple objectives. One objective is to identify common motion primitives across these various patterns, leading to a systematic taxonomy and to the ability to parameterize and reproduce these models in our simulators. Another important objective will be to create realistic, aggregate urban motion models that result from linear combinations of such patterns.

3 Technology Transfer

We have made Prawn, our wireless network protocol prototyping environment, available as part of Rutgers WinLab's ORBIT testbed. In the context of our work on MeDeHa, we have been actively interacting with the ns-3 development team. The goal is to transfer technology we have been developing for message delivery in hybrid networks prone to connectivity disruptions.

Our work on scalable routing for MANETs is being proposed to the DARPA WAND program as part of the PIRANA project. UCSC is working with BBN in this regard.

4 Publications by DAWN Investigators

4.1 Papers Published in Peer-Reviewed Journals and Books

The following peer-reviewed papers were published or accepted for publication in conference proceedings:

1. B. Shrader and A. Ephremides, "Random access broadcast: stability and throughput analysis," IEEE Transactions on Information Theory, vol. 53, no. 8, pp. 2915-2921, August 2007.

2. A. Goldsmith, S. A. Jafar, I. Maric and S. Srinivasa, Breaking Spectrum Gridlock with Cognitive Radios: An Information Theoretic Perspective, Proceedings of the IEEE, invited, to appear 2008.

3. C. T. K. Ng and A. J. Goldsmith, The Impact of CSI and Power Allocation on Relay Channel Capacity and Cooperation Strategies, To Appear in IEEE Transactions on Wireless Communications, 2008.

4. I. Maric, A. Goldsmith, G. Kramer and S. Shamai (Shitz), On the Capacity of Interference Channels with One Cooperating Transmitter, European Transactions on Telecommunications, Vol. 19, pp. 405-420, Apr. 2008.

5. T. Holliday, A. Goldsmith and H. V. Poor, "Joint Source and Channel Coding for MIMO Systems: Is it Better to be Robust or Quick?," IEEE Transactions on Information Theory, vol. 54, no. 4, pp. 1393-1405, April 2008.

6. C. T. K. Ng, N. Jindal, A. J. Goldsmith and U. Mitra, Capacity Gain from Two-Transmitter and Two-Receiver Cooperation, IEEE Transactions on Information Theory, vol. 53, no. 10, pp. 38223827, Oct. 2007.

7. F. Benbadis, K. Obraczka and J. Cortes, "Exploring Landmark Placement Strategies for Self-Localization in Wireless Sensor Networks", to appear in the EURASIP Journal of Applied Signal Processing Special Issue on Signal Processing for Location Estimation and Tracking in Wireless Environments.

8. T. Spyropoulos, T. Turletti and K. Obraczka, "Routing in Delay-Tolerant Networks Comprising Heterogeneous Node Populations", to appear in the IEEE Transactions on Mobile Computing.

9. I. Solis, K. Obraczka, "Isolines: Efficient Spatio-Temporal Data Aggregation in Sensor Networks", Special Issue on Distributed Systems of Sensors and Applications, WIRELESS COMMUNICATIONS AND MOBILE COMPUTING JOURNAL, WILEY, June 2007.

10. U. Lee, E. Magistretti, M. Gerla, P. Bellavista, P. Lio, K.-W. Lee, "Bio-inspired Multi-Agent Data Harvesting in a Proactive Urban Monitoring Environment," Elsevier Ad Hoc Networks Journal, 2008

11. Y. Yang, R. Bagrodia, "Connectivity with Multihop Relaying for Wi-Fi based Vehicular Internet Access", MOVE (MObile Networking for Vehicular Environments), May 2007.

12. S. Pollin, M. Ergen, S. Coleri Ergen, B. Bougard, L. Van der Perre, F. Catthoor, I. Moerman, A. Bahai, and P. Varaiya, "Performance analysis of slotted carrier sense IEEE 802.15.4 medium access layer," IEEE Transactions on Wireless Communiactions, 7(8):1-13, August 2008.

13. S. Timofeev, A. Bahai, and P. Varaiya, "Wideband adaptive beamforming system for speech recording," IEEE Transactions on Signal Processing, 56(7,Part 1): 2812-2820, July 2008.

14. H. Sadjadpour, Z. Wang, and J.J. Garcia-Luna-Aceves, "The Capacity of Wireless Ad Hoc Networks with Multi-Packet Reception," *IEEE Transactions on Communications*. Accepted for publication, September 2008.

15. R. Moraes, H. Sadjadpour, and J.J. Garcia-Luna-Aceves, "Many-to-Many Communication for Mobile Ad Hoc Networks" *IEEE Transactions on Wireless Communications*. Accepted for publication, 2008.

16. X. Wang and J.J. Garcia-Luna-Aceves, "Embracing Interference in Ad Hoc Networks Using Joint Routing and Scheduling with Multiple Packet Reception," *Ad Hoc Networks*, Elsevier. Accepted for publication, May 2008.

17. X. Wu, H. Sadjadpour, and J.J. Garcia-Luna-Aceves, "A Hybrid View of Mobility in MANETs: Analytical Models and Simulation Study," *Computer Communication*, Elsevier. Invited Paper, Best Paper Series. Accepted for publication, April 2008.

18. X. Wang and J.J. Garcia-Luna-Aceves, “Distributed Joint Channel Assignment, Routing, and Scheduling for Wireless Mesh Networks,” *Computer Communications*, Elsevier. Accepted for publication, 2008.

19. J. Boice, J.J. Garcia-Luna-Aceves and K. Obraczka, “Combining On-Demand and Opportunistic Routing for Intermittently-Connected Networks,” *Ad Hoc Networks*, Elsevier. Accepted for publication, 2008.

20. X. Wu, H. Sadjadpour, and J.J. Garcia-Luna-Aceves, “Modeling of Topology Evolutions and Implications on Proactive Routing Overhead in MANETs,” *Computer Communications*, Special Issue on Algorithmic and Theoretical Aspects of Wireless Ad Hoc and Sensor Networks, pp. 782–792, March 2008.

4.2 Peer-Reviewed Papers Published in Conference Proceedings

1. R. Cogill, B. Shrader, and A. Ephremides, “Stable throughput for multicast with inter-session network coding,” IEEE Military Communications Conference (MILCOM), November 2008, to appear.

2. B. Shrader, R. Cogill, and A. Ephremides, “Random linear coding for multicast over a time-varying channel,” Allerton Conference, September 2008.

3. R. Cogill, B. Shrader, and A. Ephremides, “Stability analysis of random linear coding across multicast sessions,” IEEE International Symposium on Information Theory (ISIT), July 2008.

4. U. Lee, E. Magistretti, M. Gerla, P. Bellavista, P. Lio, K.-W. Lee, ”Bio-inspired Multi-Agent Collaboration for Urban Monitoring Applications,” Proc. of Biowire’ 08, 2008.

5. S.-H Lee, U. Lee, K.-W. Lee, M. Gerla, ”Content Distribution in VANETs using Network Coding: The Effect of Disk I/O and Processing O/H,” Proc. of SECON’ 08, San Francisco, CA, June, 2008

6. S.H. Lee, S. Jung, A. Chang, D.K. Cho, M. Gerla, ”Bluetooth 2.1 based Emergency Data Delivery System in HealthNet,” Proc. of WCNC’ 08, Las Vagas, NV, March 31-April 3, 2008.

7. D.K. Cho, S.H. Lee, A. Chang, T. Massey, M. Sarrafzadeh and M. Gerla, ”Opportunistic Medical Monitoring Using Bluetooth P2P Networks,” Proc. of The Second IEEE WoWMoMWorkshop on Autonomic and Opportunistic Communications, Newport Beach, CA, June 2008.

8. U. Lee, S. Y. Oh, K.-W. Lee, M. Gerla, ”Scalable Multicast Routing in Delay Tolerant Networks,” Proc. of ICNP’ 08, Orlando, Florida, Oct., 2008

9. S. Y. Oh, G. Marfia, and M. Gerla, ”ProbeCast: MANET Admission Control via Probing,” Proc. ACM Q2SWinet 2008, Vancouver, British Columbia, Canada, Oct. 2008 .

10. A. Fujimura, S. Y. Oh, and M. Gerla, "Network Coding vs. Erasure Coding: Reliable Multicast in Ad hoc Networks," Proc. IEEE MILCOM' 08.

11. D. Gunduz, A. Goldsmith and H. V. Poor, "Diversity-Multiplexing Tradeoffs in MIMO Relay Channels," Accepted to IEEE Global Communications Conference (Globecom), New Orleans, LA, November 2008.

12. O. Simeone, D. Gunduz, A. J. Goldsmith, H. V. Poor, S. Shamai, "Compound Multiple Access Channels with Conferencing Decoders," Allerton Conference on Communication, Control, and Computing, Monticello IL, September 2008.

13. D. Gunduz, A. Goldsmith and H. V. Poor, "MIMO Two-way Relay Channel: Diversity-Multiplexing Trade-off Analysis," 42nd Asilomar Conference on Signals, Systems and Computers, Pacific Grove, CA, Oct. 2008.

14. D. Gunduz, E. Erkip, A. Goldsmith and H. V. Poor, "Lossy Source Transmission Over the Relay Channel," IEEE International Symposium on Information Theory (ISIT), Toronto, Canada, July 2008.

15. N. Liu, D. Gunduz, A. Goldsmith and H. V. Poor, "Interference Channels with Correlated Receiver Side Information", Allerton Conference on Communication, Control, and Computing, Monticello IL, September 2008.

16. I. Maric, N. Liu and A. Goldsmith, "Encoding Against an Interferer's Codebook," Allerton Conference on Communication, Control, and Computing, Monticello IL, September 2008.

17. R. Dabora, I. Maric, and A. Goldsmith, "Interference Forwarding in Multiuser Networks," Accepted to IEEE Global Communications Conference (Globecom), New Orleans, LA, November 2008.

18. D. O'Neill, A. Goldsmith, and S. P. Boyd, " Extending Network Utility Maximization to Random, Time Varying Environments in Mobile Ad-hoc Networks," Accepted to Military Communications Conference (MILCOM), San Diego, CA, November 2008.

19. D. O'Neill, A. Goldsmith, and S. P. Boyd, "Cross-Layer Design with Adaptive Modulation: Delay, Rate, Energy Tradeoffs", Accepted to IEEE Global Communications Conference (Globecom), New Orleans, LA, November 2008.

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21. R. Dabora and A. Goldsmith, "On the Capacity of Indecomposable Finite-State Channels with Feedback", Accepted to Allerton Conference on Communications, Control and Computing, Monticello, IL, Sept. 2008

22. R. Dabora, I. Maric and A. Goldsmith, "Generalized Relaying in the Presence of Interference", Accepted to 42nd Asilomar Conference on Signals, Systems and Computers, Pacific Grove, CA, Oct. 2008.

23. I. Maric, R. Dabora and A. Goldsmith, On the Capacity of the Interference Channel with a Relay, IEEE International Symposium on Information Theory (ISIT), Toronto, Canada, July 2008.

24. N. Liu and A. Goldsmith, "Superposition Encoding and Partial Decoding Is Optimal for a Class of Z-interference Channels", IEEE International Symposium on Information Theory (ISIT), Toronto, Canada, July 2008.

25. R. Dabora and A. Goldsmith, Capacity Theorems for the Finite-State Broadcast Channel with Feedback, IEEE International Symposium on Information Theory (ISIT), Toronto, Canada, July 2008.

26. R. Dabora and A. Goldsmith, The Capacity Region of the Degraded Finite-State Broadcast Channel, IEEE Information Theory Workshop (ITW), Porto, Portugal, May 2008.

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28. C. T. K. Ng, N. Jindal, A. J. Goldsmith and U. Mitra, Power and Bandwidth Allocation in Cooperative Dirty Paper Coding, IEEE International Conference on Communications (ICC), Beijing, China, May 2008.

29. D. ONeill, A. J. Goldsmith and S. Boyd, Optimizing Adaptive Modulation in Wireless Networks via Utility Maximization, International Conference on Communications (ICC) Beijing, China, May 2008.
(**Best Paper Award**)

30. I. Maric, A. Goldsmith, G. Kramer and S. Shamai (Shitz), "An Achievable Rate Region for Interference Channels with One Cooperating Transmitter", The 41st Asilomar Conference on Signals, Systems and Computers, 2007, Monterey, CA.

31. I. Maric, A. Goldsmith and M. Medard, "Information-Theoretic Relaying for Multicast in Wireless Networks", Military Communications Conference (MILCOM), 2007, Orlando, Florida.

32. Y. Liang, A Goldsmith, M. Effros, "Distortion Metrics of Composite Channels with Receiver Side Information". IEEE Information Theory Workshop (ITW), 2007, Lake Tahoe, CA. pp 559-564.

33. C. T. K. Ng, C. Tian, A. J. Goldsmith, S. Shamai (Shitz), Minimum Expected Distortion in Gaussian Source Coding with Uncertain Side Information, IEEE Information Theory Workshop (ITW), 2007, Lake Tahoe, CA, pp. 454459.

34. I. Maric, A. Goldsmith, G. Kramer and S. Shamai (Shitz), "On the Capacity of Interference Channels with a Partially-Cognitive Transmitter", IEEE International Symposium on Information Theory (ISIT) 2007, Nice, France.

35. M. Kim, Mdard, M., Aggarwal, V., and OReilly, U.-M., On the Coding-Link Cost Tradeoff in Network Coding, Proc. IEEE MILCOM 07, October 2007.

36. M. Kim, Mdard, M., OReilly, U.-M., Integrating Network Coding into Heterogeneous Wireless Networks, Proc. IEEE MILCOM 08, November 2008.

37. K. Obraczka, J. Boice, L. Martinez-Gomez, J. P. Fancois, A. Levin-Pick, and A. Weitzenfeld "Networking Under Episodic Connectivity for Autonomous Multi-Robot Coordination", in Proceedings of the IEEE Robocomm 2007.

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39. E.B. Decker, V. Rajendran, K. Obraczka, and J.J. Garcia-Luna, "The Multi-Channel Flow-Aware Medium Access Control Protocol for Wireless Sensor Networks", in the IEEE Annual International Symposiumm on Personal, Indoor and Mobile Radio Communications (PIMRC) 2008.

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41. Y. Gao, J. Hou and H. Nguyen, "Topology Control for Maintaining Network Connectivity and Maximizing Network Capacity Under the Physical Model," IEEE INFOCOM, 2008.

42. Z. Chen, X. Yang and N. Vaidya, "Dynamic Spatial Backoff in Fading Environments," *IEEE International Conference on Mobile Ad-hoc and Sensor Systems (MASS)*, September 2008.

43. P. Jeevan, S. Pollin, A. Bahai and P. Varaiya, "Pairwise Algorithm for Distributed Transmit Beamforming," Proceedings IEEE International Conference on Communications, 4245-4249, May 2008.

44. R. Rajagopal, X-L. Nguyen, S. Coleri Ergen and P. Varaiya, "Distributed Online Simultaneous Fault Detection for Multiple Sensors," Proceedings 7th International Conference on Information Processing in Sensor Networks, IPSN 2008, 133-144, 2008.

45. R. Menchaca-Mendez and J.J. Garcia-Luna-Aceves, "Scalable Multicast Routing in Mobile Ad Hoc Networks," Chapter 2, *Encyclopedia of Ad Hoc and Ubiquitous Computing*, World Scientific Publishing Co Pte Ltd, 2008.

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48. H. Kim, H. Sadjadpour, and J.J. Garcia-Luna-Aceves, A General Framework for The Capacity Analysis of Wireless Ad Hoc Networks, *Proc. IEEE MILCOM 2008*, San Diego, California, November 17–19, 2008.

49. X. Hui and J.J. Garcia-Luna-Aceves, "Efficient Broadcast in Wireless Ad Hoc and Sensor Networks with a Realistic Physical Layer," *Proc. IEEE Globecom 2008 Ad Hoc, Sensor and Mesh Networking Symposium*, 30 Nov.–4 Dec., 2008, New Orleans, LA.

50. S. Karande, Z. Wang, H. Sadjadpour, and J.J. Garcia-Luna-Aceves, "Capacity of Ad-Hoc Networks under Multipacket Transmission and Reception," (Invited Paper) *Proc. 42nd Asilomar Conference on Signals, Systems and Computers*, October 26 - October 29, 2008, Asilomar Conference Grounds, Pacific Grove, California.

51. H. Kim, H. Sadjadpour, and J.J. Garcia-Luna-Aceves, "A Closer Look at the Physical and Protocol Models for Wireless Ad Hoc Networks with Multi-Packet Reception," *Proc. 42nd Asilomar Conference on Signals, Systems and Computers*, October 26 - October 29, 2008, Asilomar Conference Grounds, Pacific Grove, California.

52. R. Menchaca-Mendez and J.J. Garcia-Luna-Aceves, "An Interest-Driven Approach to Integrated Unicast and Multicast Routing in MANETs," *The 16th IEEE International Conference on Network Protocols (ICNP 08)*, Oct. 19-22, 2008, Orlando, Florida.

53. D. Sampath and J.J. Garcia-Luna-Aceves. "Proactive Path Maintenance in Regions of Interest," *Proc. LOCAN 2008: 4th International Workshop on Localized Communication and Topology Protocols for Ad hoc Networks*, September 29, 2008 - Atlanta, Georgia.

54. R. Menchaca-Mendez and J.J. Garcia-Luna-Aceves, "Scalable Multicast Routing in MANETs Using Sender-Initiated Multicast Meshes," *Proc. IEEE MASS 2008: Fifth IEEE International Conference on Mobile Ad-hoc and Sensor Systems*, September 29 - October 2, 2008, Atlanta, Georgia.
    Best Paper Award, selected from more than 250 submissions.

55. S. Karande, Z. Wang, H. Sadjadpour, and J.J. Garcia-Luna-Aceves, "Optimal Scaling of Multicommodity Flows in Wireless Ad-Hoc Networks: Beyond the Gupta-Kumar Barrier," *Proc. IEEE MASS 2008: Fifth IEEE International Conference on Mobile Ad-hoc and Sensor Systems*, September 29 - October 2, 2008, Atlanta, Georgia.

56. Z. Wang, H. Sadjadpour and J.J. Garcia-Luna-Aceves, "Capacity Scaling Laws of Information Dissemination Modalities in Wireless Ad Hoc Networks," *Proc. Forty-Sixth Annual Allerton Conference on Communication, Control, and Computing*, September 23–26, 2008 Allerton House, Monticello, Illinois.

57. I. Solis and J.J. Garcia-Luna-Aceves, "Robust Content Dissemination in Disrupted Environments," *Proc. CHANTS 08: ACM MobiCom 2008 Workshop on Challenged Networks*, September 15, 2008, San Francisco, CA.

58. J.J. Garcia-Luna-Aceves, M. Mosko, I. Solis, R. Braynard, and R. Ghosh, "Context-Aware Packet Switching in Ad Hoc Networks," (Invited Paper) *Proc. PIMRC 2008: 19th IEEE International Symposium on Personal, Indoor and Mobile Radio Communications*, September 15-18 , 2008, Cannes, France.

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60. S. Dabideen and J.J. Garcia-Luna-Aceves, "Multi-Dimensional Routing," *Proc. ANC 08: IEEE Workshop on Advanced Networking and Communications 2008*, August 3–7, 2008, St. Thomas U.S. Virgin Islands.

61. B. Smith and J.J. Garcia-Luna-Aceves, " Best-Effort Quality-of-Service," *Proc. IEEE ICCCN 2008: 17th International Conference on Computer Communications and Networks*, August 3–7, 2008, St. Thomas U.S. Virgin Islands.

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63. X. Wang and J.J. Garcia-Luna-Aceves, "Channel Access Using Opportunistic Reservations and Virtual MIMO," *Proc. IEEE ICCCN 2008: 17th International Conference on Computer Communications and Networks*, August 3–7, 2008, St. Thomas U.S. Virgin Islands.

64. X. Wang and J.J. Garcia-Luna-Aceves, "Collaborative Routing, Scheduling and Frequency Assignment for Wireless Ad Hoc Networks Using Spectrum-Agile Radios," *Proc. IEEE ICCCN 2008: 17th International Conference on Computer Communications and Networks*, August 3–7, 2008, St. Thomas U.S. Virgin Islands.

65. X. Wu, H. Xu, H. Sadjadpour, and J.J. Garcia-Luna-Aceves, "Proactive or Reactive Routing: A Unified Analytical Framework in MANETs," *Proc. IEEE ICCCN 2008: 17th International Conference on Computer Communications and Networks*, August 3–7, 2008, St. Thomas U.S. Virgin Islands.

66. Z. Wang, H. Sadjadpour, and J.J. Garcia-Luna-Aceves, "Capacity-Delay Tradeoff for Information Dissemination Modalities in Wireless Networks," *ISIT 08: 2008 IEEE International Symposium on Information Theory*, Toronto, Ontario, Canada, July 6–11, 2008.

67. Z. Wang, H. Sadjadpour, and J.J. Garcia-Luna-Aceves, "The Capacity and Energy Efficiency of Wireless Ad Hoc Networks with Multipacket Reception," *Proc. ACM MobiHoc 2008*, Hong Kong SAR, China, May 26–30, 2008.

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70. X. Wang and J.J. Garcia-Luna-Aceves, "Embracing Interference in Ad Hoc Networks Using Joint Routing and Scheduling with Multiple Packet Reception," *Proc. IEEE INFOCOM 2008*, Phoenix, AZ, April 15–17, 2008.

71. Z. Wang, H. Sadjadpour, and J.J. Garcia-Luna-Aceves, "Closing the Capacity Gap in Wireless Ad Hoc Networks Using Multi-packet Reception," *2008 Information Theory and Applications Workshop*, UC San Diego, Jan 27–Feb 1, 2008.

4.3 Manuscripts Submitted

1. R. Cogill, B. Shrader, and A. Ephremides, "Stable Throughput for Multicast with Random Linear Coding ," submitted to IEEE Transactions on Information Theory, Oct 2008.

2. B. Shrader and A. Ephremides, "On the Shannon capacity and queueing stability of random access multicast," submitted to IEEE Transactions on Information Theory, May 2007.

3. D. Gunduz, E. Erkip, A. Goldsmith, H. V. Poor, "Transmitting Correlated Sources over Multiuser Channels," Submitted to IEEE Transactions on Information Theory, 2008.

4. O. Simeone, D. Gunduz, H. V. Poor, A. J. Goldsmith, S. Shamai, "Compound Multiple Access Channels with Partial Cooperation," Submitted to IEEE Transactions on Information Theory, July 2008.

5. N. Liu and A. Goldsmith, "Capacity Regions and Bounds for a Class of Z-interference Channels", Submitted to IEEE Transactions on Information Theory, July 2008.

6. M. Effros, A. Goldsmith, Y. Liang, "Capacity definitions for general channels with receiver side information, submitted to the IEEE Transactions on Information Theory, 2008.

7. R. Dabora and A. Goldsmith, The Capacity Region of the Degraded Finite-State Broadcast Channel, submitted to the IEEE Transactions on Information Theory, March 2008.

8. C. T. K. Ng, D. Gndz, A. J. Goldsmith and E. Erkip, Optimal Power Distribution and Minimum Expected Distortion in Gaussian Layered Broadcast Coding with Successive Refinement, Submitted to IEEE Transactions on Information Theory, 2007.

9. F.B. Abdesslem, M.D. Amorim, K. Obraczka, and L. Iannone, "A Simple Tool to Support Teaching Wireless Networking Classes with a Hands-On Focus", submitted to the IEEE Transactions on Education.

10. T. Spyropoulos, N.B. Rais, T. Turletti and K. Obraczka, "Routing for Intermittently Connected Networks: Taxonomy and Design", submitted to the IEEE Transactions on Mobile Computing.

11. K. Obraczka, J.J. Garcia-Luna-Aceves, J. Boice, "An Efficient Routing Framework for Self-Organizing Networks with Varying Connectivity" submitted to the ACM TAAS Special Issue on Special Issue on Self-Adaptive and Self-Organizing Wireless Networking Systems.

12. F.B. Abdesslem, M.D. Amorim, K. Obraczka, and L. Iannone, "Rapid Prototyping over IEEE 802.1", submitted to the Elsevier Journal of Networks and Computer Applications.

4.4 Ph.D. and M.S. Theses Completed

The following theses were completed during the past year:

1. Minkyu Kim, "Evolutionary Approaches Toward Practical Network Coding," Ph. D. thesis, MIT, 2008.

2. Brooke Shrader, Ph.D. Thesis, University of Maryland, July 2008.

3. Vartika Bhandari, "Performance of wireless networks subject to constraints and failures," Ph.D. thesis, University of Illinois, 2008.

5 Honors and Awards

The faculty participating in DAWN received the following honors and awards. The list is provided in alphabetical order according to the last names of principal investigators.

5.1 Tony Ephremides

- Invited plenary talks at the Information Theory Workshop in Porto, Portugal, May, 2008
- Imvited plenary talk at the WPMC in Lapland, September 2008

5.2 J.J. Garcia-Luna-Aceves

- Best Paper Award (out of 250 submissions):
  R. Menchaca-Mendez and J.J. Garcia-Luna-Aceves, "Scalable Multicast Routing in MANETs Using Sender-Initiated Multicast Meshes," *Proc. IEEE MASS 2008: Fifth IEEE International Conference on Mobile Ad-hoc and Sensor Systems*, September 29 - October 2, 2008, Atlanta, Georgia.
- Selected to the **Marquis Who's Who in The World**.
- General Chair, ACM MobiCom 2008, September 14–19, 2008, San Francisco, CA.

5.3 Andrea Goldsmith

- Best paper award:
  "Optimizing Adaptive Modulation in Wireless Networks via Utility Maximization," D. ONeill, A. Goldsmith, and S. Boyd, *Proc. IEEE ICC 2008*, Beijin, China, May 2008.
- President-Elect: IEEE Information Theory Society (for 2009)
- Stanford Postdoc Mentoring Award.

5.4 Muriel Medard

- Received the 2007 Gilbreth Lectureship from the National Academy of Engineering
- Elected a Fellow of the IEEE

5.5 Pravin Varaiya

- 2008 AACC Richard E. Bellman Control Heritage Award
- Honorary Doctorate, Technical University of Crete

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