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Organization: GA Tech Res Corp - GIT

Submitted By:

Ammar, Mostafa - Co-Principal Investigator

Title:

Collaborative Research: NeTS-NBD: Construction of Robust and Efficient Disruption Tolerant Networks

Project Participants

Senior Personnel

Name: Zegura, Ellen

Worked for more than 160 Hours: Yes

Contribution to Project:

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Worked for more than 160 Hours: Yes

Contribution to Project:

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Worked for more than 160 Hours: No

Contribution to Project:

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Name: Jun, Hyewon

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Name: Mansy, Ahmed

Worked for more than 160 Hours: Yes

Contribution to Project:

Name: Shi, Cong

Worked for more than 160 Hours: Yes

Contribution to Project:**Name:** Sachdeva, Puhskar**Worked for more than 160 Hours:** Yes**Contribution to Project:****Undergraduate Student****Name:** Olson, Jon**Worked for more than 160 Hours:** Yes**Contribution to Project:**

Undergraduate Student

Name: Webb, Kevin**Worked for more than 160 Hours:** No**Contribution to Project:****Technician, Programmer****Other Participant****Research Experience for Undergraduates****Organizational Partners****University of Massachusetts Amherst**

This is a collaborative project with Brian Levine and Mark Corner. We have co-authored papers, worked with each others' students and exchanged system building expertise and data.

Cisco Systems Inc

We have received a donation of mobile routers from Cisco. Also we recently received a research grant supporting our research in DTN.

University of Paris

We hosted, Vincent Borrel, a PhD student from the University of Paris 6 during the Spring semester of 2007.

University of Southern California

Working with Konstatnions Psounis

Thomson Research ab, Paris

Mostafa Ammar collaborated with researchers at Thomson research in Paris during a visit in summer 2009.

Other Collaborators or Contacts

We have also received funding for research in Disruption Tolerant Networks from DARPA.

Activities and Findings

Research and Education Activities: (See PDF version submitted by PI at the end of the report)

Findings: (See PDF version submitted by PI at the end of the report)

Training and Development:

The project is training several graduate students in the DTN area and in the wireless and mobile networking area in general. Further, our project has involved undergraduate research students who have been designing and building our testbed. They have accumulated considerable experience in DTN networks as well general networking architectures and testbed deployment.

Wenrui Zhao completed his degree in December 2006. Hyewon Jun, also graduated in December 2007. Both are working at Google.

Pushkar Sachdeva, completed his MS degree in 2009 and is now at Yahoo's mobile group in Sunnyvale, CA.

Outreach Activities:

We have the following activities to report:

- Several invited and keynote talks have been given by Ammar and Zegura in the last two years in this area.
- We have participated in the meeting of DTN research group of the IRTF and have made presentations of our work.
- We plan to make our DTN sharp implementation available for others to use.
- We participated in the DTN inter-operability testing.
- Mostafa Ammar has given talks at University of Illinois (Urbana-Champaign), and Boston University in April and May 2009.
- Mostafa Ammar participated in the 2009 ACM Mobihoc Panel.
- Mostafa Ammar gave the keynote talk at the TCCC Computer Communications Workshop in Lake Arrowhead, CA in 2010.

Journal Publications

Yang Chen, Jeonghwa Yang, Wenrui Zhao, Mostafa Ammar and Ellen Zegura, "Multicasting in Sparse MANETs Using Message Ferrying", Proceedings of the IEEE Wireless Communications and Networking Conference, p. x, vol. , (2006). Published,

Barath Petit, Mostafa Ammar, Richard Fujimoto, "Protocols for Roadside-to-Roadside Data Relaying over Vehicular Networks", Proceedings of the IEEE Wireless Communications and Networking Conference, p. x, vol. , (2006). Published,

- Mukarram Bin Tariq, Mostafa Ammar, Ellen Zegura, "Message Ferry Route Design for Sparse Ad hoc Networks with Mobile Nodes", Proceedings of MobiHoc 2006, the Seventh ACM International Symposium on Mobile Ad Hoc Networking and Computing, p. 37, vol. , (2006). Published,
- Hyewon Jun, Mostafa Ammar, Mark Corner, Ellen Zegura, "Hierarchical Power Management in Disruption Tolerant Networks with Traffic-Aware Optimization", Proceedings of the ACM SIGCOMM workshop on Challenged Networks, p. x, vol. , (2006). Published,
- Wenrui Zhao, Yang Chen, Mostafa Ammar, Mark Corner, Brian Levine, Ellen Zegura, "Capacity Enhancement using Throwboxes in DTNs", The Third IEEE International Conference on Mobile Ad-hoc and Sensor Systems (MASS), p. 31, vol. , (2006). Published,
- Hyewon Jun, Wenrui Zhao, Mostafa Ammar, Ellen Zegura, and Chungki Lee, "Trading Latency for Energy in Densely Deployed Wireless Ad Hoc Networks using Message Ferrying", Elsevier Journal of Ad Hoc Networks, p. 444, vol. 5, (2007). Published,
- Yang Chen, Wenrui Zhao, Mostafa Ammar, Ellen Zegura, "Hybrid Routing in Clustered DTNs with Message Ferrying", ACM/SIGMOBILE Workshop on Mobile Opportunistic Networking, p. , vol. , (June). Accepted,
- badri Vellambi, Ramanan Subramanian, Faramrz fekri, Mostafa Ammar, "Reliable and Efficient Message Delivery in Delay Tolerant Networks using Rateless Codes", ACM/SIGMOBILE Workshop on Mobile Opportunistic Networking, p. , vol. , (2007). Accepted,
- Vincent Borrel, Mostafa Ammar, Ellen Zegura, "Understanding the Wireless and Mobile Network Space: A Routing-centered Classification", ACM Mobicom Workshop on Challenged Networks, p. , vol. , (2007). Published,
- Ahmed Mansy, Mostafa Ammar, Ellen Zegura, "Reliable Roadside-to-Roadside Data Transfer Using Vehicular Traffic", International Workshop on Mobile Vehicular Networks, p. , vol. , (2007). Published,
- Russ Clark, Evan Zasoski, Jon Olson, Mostafa Ammar, Ellen Zegura, "D-Book: A Mobile Social Networking Application for Delay Tolerant Networks", Demo Abstract: ACM Mobicom Workshop on Challenged Networks, p. , vol. , (2008). Submitted,
- Dimitrios Antonellis, Ahmed Mansy, Konstantinos Psounis, Mostafa H. Ammar, "Towards Distributed Network Classification for Mobile Ad hoc Networks", Proceedings of the Wireless Internet Conference (WICON) -- Invited Paper, p. , vol. , (2008). Published,
- Yang Chen, Vincent Borrel, Mostafa Ammar, Ellen Zegura, "A Framework for Characterizing the Wireless and Mobile Network Continuum", SIGCOMM Computer Communications Review, p. , vol. , (2011). Accepted,
- Bahadir Polat, Pushkar Sachdeva, Mostafa Ammar, Ellen Zegura, "Message Ferries as Generalized Dominating Sets in Intermittently Connected Networks", The Second International Workshop on Mobile Opportunistic Networking ACM/SIGMOBILE MobiOpp 2010, p. , vol. , (2010). Published,
- Hyewon Jun, Mostafa Ammar, Mark Corner, Ellen Zegura, "Hierarchical Power Management in Disruption Tolerant Networks Using Traffic Aware Optimization", Computer Communications Journal, p. , vol. , (2009). Accepted,
- Abderrahmen Mtibaa, Martin May, Christophe Diot, Mostafa H. Ammar, "PeopleRank: Social Opportunistic Forwarding", Proceedings of INFOCOM 2010 Symposium, p. , vol. , (2010). Published,
- Abderrahmen Mtibaa, Martin May, Mostafa H. Ammar, "On the Relevance of Social Information to Opportunistic Forwarding", Proceedings of IEEE/ACM MASCOTS Conference, p. , vol. , (2010). Published,
- Mostafa H. Ammar, Deeparnab Chakrabarty, Atish Das Sarma, Subrahmanyam Kalyanasundaram, Richard J. Lipton, "Algorithms for Message Ferrying on Mobile ad hoc Networks", IARCS Annual Conference on Foundations of Software Technology and Theoretical Computer Science, p. , vol. , (2009). Published,
- Cong Shi, Daniel Luo, Patrick Traynor, Mostafa Ammar, Ellen Zegura, "ARDEN: Anonymous Routing in Delay tolErant Networks", Proceedings of INFOCOM 2011, p. , vol. , (2011). Submitted,

Cong Shi, Mostafa Ammar, Ellen Zegura, "iDTT: Delay Tolerant Data Transfer for P2P File Sharing Systems", Proceedings of INFOCOM 2011, p. , vol. , (2011). Submitted,

Books or Other One-time Publications

Web/Internet Site

Other Specific Products

Contributions

Contributions within Discipline:

Our work has provided significant advances in our understanding of the design of delay and disruption-tolerant networking. In particular we have made the following contributions:

- We provided further elaboration on the message ferrying methodology which we developed earlier. We have devised schemes for multicast data dissemination using ferries and we have developed a novel ferry route design algorithm that targets mobile DTNs.

- We have shown the applicability of the DTN technique of using mobility to carry data to the context of moving data in a vehicle-to-vehicle application.

- Our work produced algorithms for the design of a DTN using 'throwboxes' to enhance the data capacity of the network. This is a necessary complement to efforts that provide hardware designs for such relay devices.

- We are the first to consider issues of power management in DTNs. Our work provides an understanding of the relative merits of hierarchical power management in DTNs.

- We investigated novel use of rateless coding in the context of DTNs.

- We developed the first .NET implementation of the DTN bundle spec. The implementation is suitable for PDAs and smartphone use.

- We have developed a novel fundamental framework for understanding the space of wireless and mobile networks and in particular how DTNs related to previous efforts in this domain.

- Developed a refined formalism for understanding the space of wireless and mobile networks.

- Extended the framework to include energy sufficiency.

- Developed a generalization of the connected dominating set concept and algorithms for intermittently-connected networks.

- Refined earlier formalism for understanding the space of wireless and mobile networks.
- Extension of the framework to include energy sufficiency.
- A generalization of the connected dominating set concept and algorithms for intermittently-connected networks.
- Developed a new scheme for anonymous routing in DTNs. The scheme extends onion routing techniques used in wired networks.
- Designed, prototyped and evaluated an architecture that uses delay-tolerant data delivery techniques to achieve ISP-friendly data delivery for peer to peer applications.

Contributions to Other Disciplines:

Contributions to Human Resource Development:

Our work involves several graduate and undergraduate students. Two PhD students (Wenrui Zhao and Hyewon Jun) graduated in December 2006 and December 2007. Both are now working at Google.

Kevin Webb, an undergraduate working on the DTN sharp implementation has been accepted to the PhD program in UC San Diego, Jon Olson another undergraduate is now enrolled in the MS program at Georgia Tech.

Pushkar Sachdeva graduated with an MS degree in 2009. He is now with Yahoo mobile group.

Contributions to Resources for Research and Education:

Our DTN Sharp implementation will be made available to others. It promises to be a useful resource for other DTN researchers as well mobile application builders in general.

Contributions Beyond Science and Engineering:

DTNs have potential applications in challenged or hostile environments (e.g., disaster relief). As such our work has the long-term potential of enabling communication in such environments. Our goal in the testbed construction effort we just started within this project (and expect to report on fully next year) is to provide a proof of concept for the use of DTNs in challenged environments.

Conference Proceedings

Categories for which nothing is reported:

- Any Book
- Any Web/Internet Site
- Any Product
- Contributions: To Any Other Disciplines
- Any Conference

Note: Some work reported in this report overlaps with work reported for NSF Award 0831714: NETS-NECO: The WAM Continuum: Unified Design and Operation for Wireless and Mobile Networks

Findings:

1- Novel schemes using message ferrying

Multicast communication using message ferrying

Using simulations, we evaluate our proposed protocols under various application/network conditions. We show that a single cure-all multicast solution for sparse MANETs is highly unlikely. In addition, the adaptive protocols which are able to adapt their behaviors according to the scenarios generally have good performance.

The objective of a multicast protocol is to maximize message delivery rate while minimizing overhead. Although the applications in sparse MANET should be delay tolerant, a shorter delivery latency is still preferable. Therefore, we use the following metrics in our evaluation of the protocol performance.

² **Message delivery ratio.** *Message delivery ratio* is defined as the ratio of the number of messages actually received by group members to the number of messages that should have been received. The higher the ratio is, the more effective a protocol in delivering messages to receivers.

² **Message forwarding redundancy.** *Message forwarding redundancy* is defined as the number of messages transmitted per delivered message. This is about routing efficiency and accounts for all message transmissions by both source nodes and intermediate forwarding nodes.

² **Message delivery latency.** *Message delivery latency* is defined as the time from the instant when a message is sent out to the instant when the receiver receives this message. Due to the multiple-destination nature of the multicast message, we evaluate the average latency over the delivered messages.

We first investigated the performance of all the multicast protocols we developed. We observed that the aggressive duplication helps ER and MFER to disseminate messages fast. Therefore, they have the best performance especially when the delivery is time-constrained. The delivery ratio of MF is the worse because it relies only on the ferry-node contacts. Due to the utilization of limited epidemic routing, other MF based protocols, e.g., MFER, outperform MF.

We show that ER and MFER are associated with high message redundancy although their performance improves as the group size increases. Meanwhile, MF and MFGR show no redundancy over all the group sizes since every duplication and message exchange involve only group members. The two adaptive protocols, MF/MFER and

MF/MFGR/MFER, achieve reasonable message redundancy since they adjust their flooding scopes according to the group size.

On the other hand, we show that by involving more nodes in the message dissemination process, the delivery latency is significantly less in ER than other protocols. The reason that ER performs better than MFER is that in ER, a source broadcasts directly to neighbors while in MFER, the source sends the message to the ferry to start the broadcasting. This procedure is used to provide a fair comparison on delivery latency with other protocols in which the ferry starts the “real” multicasting after it receives the message from the source node.

We also investigate the tradeoff between delivery redundancy and latency involved in those protocols. The simple extensions of ER and MF suffer from either high redundancy or long delay for certain group sizes. Meanwhile, hybrid protocols stay in the intermediate region due to the combination of MF and ER. And they have much less performance fluctuations due to their adaptation to the group size change.

Message ferry routing for mobile nodes

Our key contribution through this work is an algorithm for designing message ferry routes in sparse networks where nodes have arbitrary movement, that does not require any online collaboration between the ferry and the nodes, and does not disrupt the mobility of the nodes. We call this method the Optimized Way-points (OPWP) ferry routing method. In this method we determine message ferry routes that comprise an ordered set of way-points and waiting times at these way-points, which are chosen carefully based on the mobility model of the mobile nodes in the network. Every time that the ferry traverses this route, it contacts every node in the network with a certain minimum probability. This node-ferry contact probability in-turn determines the distribution of node-ferry contact times, and ultimately the properties of end-to-end message delivery. The ferry can repeatedly follow the same OPWP route, without any changes, as long as the system remains in steady-state, i.e, the nodes continue to use the same mobility model, and the traffic demands do not change. Thus we do not require any kind of adaptability from the ferry once we program it to follow the route.

We evaluate the OPWP routing algorithm relative to several others:

Random Way-point (RWP): The ferry moves according to the random way \ point mobility model.

Restricted Random Way-point (RWP-res): The ferry moves as in the ferry RWP model, with the exception that the way-points are only chosen from the part of the region which is also part of mobility region of one of the mobile nodes.

Space Filling Way-point (SFWP): We divide the region into a grid, and pick the center of the each grid-box as a way-point. Next we order these way-points so as to form a shortest possible tour using an approximate Traveling Salesman (TSP)solution. The ferry route comprises traversing this ordered set of way-points repeatedly

Region Centers Way-point (RCWP): We use the centers of the regions of each of the mobile nodes as the way-points and order these using a TSP approximation.

We compare the the different ferry routing models based on the following three criteria.

- Frequency of Node-Ferry Contacts.
- Fairness of Node-Ferry Meetings Among Nodes.
- Delay and Loss of Messages:.

We find that the SFWP schemes are clearly the worse performers. One might think that since SFWP schemes cover the entire region, they would perform well. However, there are good reasons for their poor performance. The length of the route (tour) for the SFWP-are long. The longer route length means that the ferry takes a long time to complete one loop through the route, and a significant fraction of this time is spent traveling in the parts of the region that have zero (parts that are not covered by smaller rectangles in *_g. 4*) or negligible probability of node presence. Note that even though the ferry covers the entire region, it does not cover the entire region at once, so the nodes and ferry can keep missing each other.

The vanilla RWP ferry model suffers from somewhat same problems as the SFWP models. The ferry may choose random way-points that are not in the region of any mobile node, thus time spent traveling to and from such way-points is completely futile, except for when the path to these waypoints intersects the region of some mobile node. Even when the ferry does choose a way-point that is in the region of a mobile node, the probability that it would be at a point where mobile node is most likely to be is small. The reason that RWP performs better is that the ferry tends to meet the nodes that are in the center of the region quite more often than the mobile nodes that are at the edges of region. This increases the average node-ferry meetings, but without any degree of fairness.

The RWP-res model alleviates part of the problems of vanilla RWP model by ignoring way-points that are not in the regions of the mobile nodes, and as a result, the RWP-res model performs slightly better than vanilla RWP. However, other problems of the RWP model remain. The RCWP model does quite well when the nodes follow the RWP, or AB-RWP model. The reason is that RCWP uses a way-point that is the center of the region of the mobile node, and for the nodes moving under RWP model, the center also happens to be the point where the node is most likely to be present

Our second evaluation criteria is fairness. The OPWP model has the least variation between the meetings for different nodes, and consequently yields an almost perfect fairness index value. We show in our work that the RWP node visits the central region more often than the regions at the edges. This means that a RWP ferry would attempt to meet the nodes that are situated near the center of the region more often than the nodes that are situated farther away from the region, and as a result, the ferry would meet some nodes more often than others.

Finally, we summarize the results from our comparison based on delay and loss. We compute the end-to-end delay only for the messages that have been successfully

downloaded. Note that there still are messages in the ferry and source node buffer at the time of simulation termination. Note that while the OPWP routing model does not do significantly better in terms of end-to-end delay for RWP node mobility model, it does better in terms of the message loss for this model. When the nodes follow more constrained mobility models, OPWP does clearly better both in terms of loss and end-to-end delay.

The OPWP scheme we develop does significantly better than any other model. The main reason is that we balance the traveling time and waiting time, and moreover invest that waiting time at way-points that are most advantageous in terms of increasing the probability of contact with the node.

2- Using DTN schemes in V2V networks

We evaluated our protocols for roadside-to-roadside relaying over moving vehicles using discrete event simulation. The vehicle mobility was generated using the CORSIM traffic simulator. CORSIM is a microscopic traffic simulator where the position of all vehicles at each time-step is calculated using car-following theory. We generated mobility patterns using CORSIM for a 5000m expressway with four lanes. The *free-flow velocity* which is the unconstrained velocity of the vehicle on the freeway was varied from 40mph to 70mph.

We focused on the following metrics:

- The *throughput* of the data uploaded to the sink during an interval of t units of time.
- The *utilization* which is defined as the proportion of time spent by the sink in the transfer phase.

We investigated the variation of the throughput with free-flow velocity of the vehicular traffic. We found that the throughput is relatively insensitive to the velocity of the traffic. This can be attributed to the observation that given sufficient number of eligible vehicles in the hand-off and pick-up zones, the throughput depends on the extent of the reduction of the idle-time of the base-stations. We found that vehicular was a major factor affecting the throughput.

We also investigated the performance as a function of traffic density (not to be confused with vehicular traffic speed). We observed that once the number of vehicles entering the roadway exceeds a certain threshold, there is very little variation in the obtained throughput among the four protocols. This two-phased behavior of the throughput with respect to the vehicle density seems to suggest a *critical mass* for traffic density in order to maximize the throughput. At lower vehicle densities, there exists a paucity of eligible vehicles which leads to an increase in the number of vehicles that have data payload (to be relayed) with increase in vehicle entry rates. This in turn reduces the idle-times of the base-stations which leads to an increase in throughput. In the case of higher vehicle

densities, we find that the dominating factor is the protocol being used at the base-stations which dictate the idle-time ensued by the respective source and sink.

Finally, we considered the *penetration ratio*, defined as the proportion of vehicles that are equipped with the required network interfaces and/or GPS equipment. In the scenario where the free-flow velocity is set at 65mph, we varied the penetration ratio from 0.05 to 0.5. As expected, an increase in the penetration ratio results in increase of the throughput.

It is important to note that one of our main contribution here is the inclusion of vehicular traffic metrics, into the consideration of data throughput in a DTN where the nodes are vehicles. We also showed that it is indeed feasible to use moving vehicles as a data transport pipe using DTN carrying and forwarding techniques.

3- Use of throwboxes to enhance DTN data carrying capacity

In evaluating our algorithms for placement and routing using throwboxes, we wanted to verify several hypotheses: (i) throwboxes enhance throughput in a DTN, (ii) deployments that utilize information about load and jointly optimize for routing see greater gains than those that ignore that information, and (iii) throwboxes enhance connectivity in multiple mobility scenarios. To demonstrate this, we evaluate our deployment and routing algorithms using ns simulations.

We consider two performance metrics, namely message delivery ratio and delay. The Message Delivery Ratio is defined as the ratio between the number of unique messages being delivered and the number of generated messages. Messages might be dropped because of buffer overflows or message timeout. The delivery ratio is computed over the simulation duration, which measures how successful each scheme is in delivering messages. The Message Delay is the average time from the generation of a message to the earliest reception of the message at the destination. The message delay considers delivered

We first consider the case of multi-path routing. We make the following observations. First, the delivery ratios improve significantly as the number of throwboxes increases under CTB deployment. For example, for multi-path routing with CTB deployment, the delivery ratio increases by a factor of three using four throwboxes. Throwboxes are able to improve delivery ratios under both regular mobility and random mobility. Second, contact based deployment generally performs worst than CTB deployment, especially in the case of regular mobility. This is because CTB deployment is able to utilize traffic information to optimize performance. Third, for oblivious deployment, the delivery ratio increases under random mobility but is not affected with regular mobility. This is due to the constrained node movement and the relatively large span of the area, random deployment tends to place throwboxes to locations where nodes do not visit. In contrast, due to random movement, nodes in the RWP models would meet with throwboxes eventually even though throwboxes are placed randomly in the area.

For the case of single path routing we found that as the number of throwboxes increases, the delivery ratio improves with CTB deployment. Contact based deployment, on the other hand, improves the delivery ratio only when the number of throwboxes is relatively large. This confirms that using traffic information improves the effectiveness of throwbox deployment. Oblivious deployment, as expected, does not affect the delivery ratio.

For the case of epidemic routing, message forwarding is based on actual contact between nodes, so we focused on the deployment issue. Specifically, we compute throwbox locations as in multi-path routing. The intuition is that epidemic routing is able to exploit all paths available to propagate messages. So epidemic routing would benefit from the use of throwboxes even if the deployment is intended for multi-path routing. We found that all deployment schemes achieve similar delivery ratios. As compared to the cases of multi-path and single path routing, throwboxes have limited improvement on the delivery ratio because of the poor utilization of resources caused by message flooding in epidemic routing. We observe that contact-based and CTB deployment achieve significant improvement in the delivery ratio, while random deployment has modest improvement. Epidemic routing benefits from the use of throwboxes for all deployment schemes because of the random movement of nodes.

To summarize, our results suggest several findings to guide the design and operation of throwbox-augmented DTNs:

- Throwboxes are very effective in improving throughput and can also reduce data delivery delay. The improvement in throughput is generally more significant than improvements in delay.
- Throwboxes are most useful for routing algorithms that use multi-path routing and when nodes follow structured mobility patterns.
- Throwbox deployment that incorporates knowledge about contact opportunities performs better than deployment that ignores this knowledge. Additionally, if deployment is customized to existing traffic patterns, the algorithms are more effective than assuming that traffic is equally distributed.

4- Power management in DTNs

In our evaluation of our power management schemes, we consider two metrics:

(1) *contact discovery ratio*, which is the discovered contact time divided by the total contact time that the high-power radio could have discovered when using CAM, and

(2) *energy efficiency*, which is the average amount of discovered contact time per unit energy used. In other words, we measure how much energy the system spends to find a certain amount of message transfer time.

Using simulation, we find that the contact discovery ratio of PSM decreases as the wake-up interval increases. On the other hand, GPSM discovers a moderate proportion of contacts for all wake-up intervals because the low-power radio assists the discovery. In addition, its discovery ratio is always more than that of SPSM because of the high-power

radio. We also find that the energy efficiency of GPSM and PSM increases and then decreases as the wake-up interval decreases. Once most of the contacts are discovered, using smaller wake-up intervals wastes energy without discovering much more contacts. Also, the energy efficiency of GPSM outperforms those of PSM and SPSM for all wake-up intervals. Our experiments demonstrate the usefulness of the second radio is highly dependent on that wake up interval selection.

We also evaluated our wakeup interval selection algorithms. Our goal in evaluating our to show two things. First, we show that our analytical model can accurately predict the fraction of contacts for any particular wake up interval. If the predicted number of contacts is correct, then the wake up interval can be tuned to a particular traffic load. Second, we show that these wake-up mechanisms save significant energy in a DTN node; and compare the use of single and multiple radio search mechanisms.

From this evaluation, we conclude the following. First, all of the techniques save considerable energy by utilizing the traffic-aware optimization. Secondly, the performance of each mechanism depends on the node mobility model: PSM works better when nodes pass by each other at long distances, and SPSM works better when nodes tend to meet at short distances. GPSM is flexible enough to adapt its behavior to both mobility scenarios and achieve the equivalent performance of the best performing power management scheme in each scenario. Third, traffic-aware optimization saves energy more efficiently than purely optimizing for discovery performance. However, in both mobility scenarios PSM achieved similar performance to GPSM, bringing into question the necessity of a second radio. As we have stated, the second radio is most useful in other mobility scenarios where the second radio is in range of other nodes most of the time.

To summarize, our results show that the generalized power management mechanism could augment the usefulness of the low power radio and achieve better energy efficiency than mechanisms relying only on one radio for contact discovery. Also, our approximation algorithms help to save considerable energy when compared to the case without power management. However, our simulation results show that while an additional low power radio does reduce the energy consumption needed for discovery, the improvement could be negligible in mobile DTNs. While this seems to contradict the results of previous studies, the differences lie in the density and mobility in the network. In particular, previous research has targeted nomadic computing scenarios where the density of nodes is much higher. In these situations the low-power radio can discover contact opportunities as well as the high-power radio. In DTNs, the low-power radio may discover much fewer contacts than the high-power radio, thus lowering the overall efficiency. Also, if additional information about contacts and traffic load is available, a node even with one radio could save considerable energy. As a result, the overall benefits of the hierarchal radio architecture becomes to a level similar to using one radio alone.

5. Message Ferrying in Clustered Wireless Networks

In this work, we studied data routing in clustered DTNs where nodes are partitioned into clusters. We focus on the case of stationary nodes with message ferrying. We develop a hybrid routing approach in which both MANET routing and message ferrying are used to explore available connectivity in clusters via gateway nodes. We investigate the interaction between gateway nodes and the ferry. Different data aggregation as well as transmission scheduling algorithms are proposed. To achieve better performance, we also study the ferry route design problem in the clustered DTNs. Via extensive simulations, we evaluate our hybrid routing approach, which provides the following guidelines for data forwarding in clustered DTNs.

1- Data aggregation schemes and transmission scheduling algorithms are shown to have significant effects on data delivery performance. In particular, we find that simple schemes that choose gateways randomly and give equal priority to traffic in both directions between gateways and the ferry achieve good delivery ratio while utilizing more information about contacts and traffic load provides certain improvement.

2- While connectivity among clusters allows the ferry to visit a subset of nodes in clusters, customized ferry routes can lead to substantial improvement in performance, e.g., higher delivery ratio and lower delay. However, intra-cluster traffic can cause severe interference to data exchange between the ferry and gateways. In certain cases, this may degrade the performance gain from ferry route optimization.

6. Use of Rateless Coding in DTNs

We evaluated our rateless encoding scheme according to the following metrics 1) latency, reliability, and probability of success. We compared our scheme against the following schemes:

1. Uncoded Unpacketized Scheme,
2. Uncoded Packetized Scheme
3. Erasure-coding based Scheme

We found that our scheme's performance, while dependent on various parameters (such as mobility patterns), it generally outperforms the other schemes we considered.

7. Roadside-to-Roadside Data Transfer Using Vehicular Traffic

In this work we provided further elaboration on the idea of using vehicular traffic as data transfer channel between roadside stations. We first described the protocol features and mechanisms that can be used to transfer data between roadside stations and moving vehicles. We then discussed and evaluated several approaches that can be used to achieve end-to-end data transfer reliability between two roadside stations. We find that a rateless coding approach is best but presents a practical limitation on the size of the file that can be transferred. A hybrid data replication and ARQ scheme works best when transferring larger files.

There are many remaining issues that need to be resolved before this particular form of data transfer can become a reality. Clearly further elaboration on the protocol details leading to a prototype implementation will go a long way to provide a proof of concept.

There are also issues of data security and privacy as well as insuring vehicles have the appropriate incentive structure to participate in this form of data transfer.

8. Understanding the Wireless and Mobile Network Space

In this work we have proposed a framework for classifying wireless and mobile networks, with the goal of having the classification inform the design of routing for the network. Our approach is based on the theory of evolving graphs and provides for three classes of networks (SPN, U-DTN and A-DTN), each derived from our understanding of routing approaches within such networks. We develop formal idealized and practical classifications. The former is based on infinite-duration evolving graphs, while the latter consider finite duration graphs. Our practical classification is based on parameters derived from the constraints imposed by routing protocols. We also develop a methodology that can be applied to given mobility models and traces to obtain the classification for a given network scenario. Finally, we illustrated the use of our classification approach using example network scenarios and mobility models.

We view this as the beginning of an examination of the important question of how one can classify networks with the goal of understanding their design and operation. While we believe that our work so far has touched upon most aspects of this problem, there are many important issues that require further consideration. These include:

- Further formulation of the process of network transformation that can be used to change one network class into another.
- Extensions of the classification formalisms to allow for *partial* classification that may for example include only a specified subset of node pairs in a classification scheme.
- A more in-depth investigation of how to devise parametric classification based on various routing protocols.
- More experience in using the classification approach for other mobility models and network scenarios with possibly a specific application to routing design exercise.

9. A Framework for Characterizing the Wireless and Mobile Network Continuum

Our results provide methods for describing and classifying: 1) node pair connectivity properties, 2) whole network connectivity properties and 3) node pair and network energy sufficiency. We illustrate some of these results here.

First we need to provide the following definitions where we assume that an evolving graph G describes the WAM under consideration.

Definition: Space-Path Pair (S-Pair): A pair of nodes i and j is called an S-Pair if a space path can always be found between them in G .

Data can still be transferred between node pairs which are not S-Pairs with the store-carry-forward paradigm. To capture the character of those node pairs, we have the following node pair definition:

Definition: Space-Time-Path Pair (ST-Pair): A pair of nodes i and j is an ST-Pair if:

- For all t , a journey exists in G between i and j at t .
- this pair is not an S-Pair.

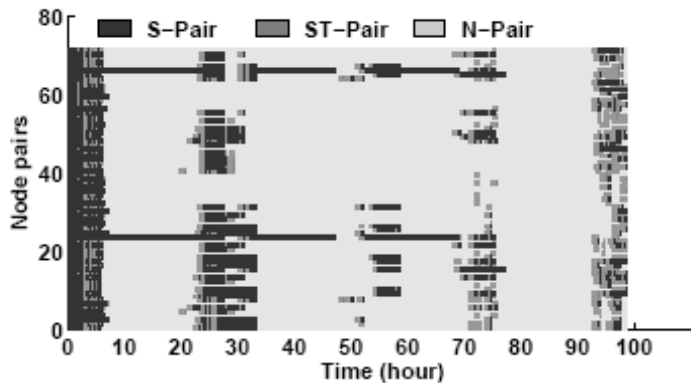
It is possible that no path can be found between particular node pairs after certain time point. Therefore, we introduce the last node pair class.

Definition: No-Path Pair (N-Pair): A pair of nodes i and j is an N-Pair if $\exists t$ such that no journey can be found from i to j after t in G .

We augment our node-pair classification above to incorporate some degree of practical routing consideration. To understand the motivation, consider a network where connectivity changes rapidly but where a particular node pair is always connected through a space path. While a space path exists all the time, the actual links forming the path change in such a manner that no specific path persists for a long time. According to our definitions above, this pair will be classified as an S-Pair and one may reach the conclusion that a MANET routing protocol may be suitable for routing between the two nodes. However, such routing protocols need some time to discover a route and therefore require a certain amount of *route persistence*. In that case we need to modify our S-Pair definition to take this into account.

The previous classification provides a rather strict categorization of node pairs. For example, if a node-pair that is connected through a space path, losing connectivity, even for a brief period of time, makes it not an S-Pair. To address this issue we propose to classify network over time. Time is broken up into *epochs* during which the network connectivity (or graph in an evolving graph) does not change. Our goal is to provide a classification *for each node pair in each epoch*.

The figure below illustrates our results where the node pairs in a trace taken from the Huggle project are classified over time.



We now show how the per-node-pair classification described above can be aggregated to classify whole networks.

Three categories of network are proposed: *Space Path Network (SPN)* in which MANET routing protocols can be applied among all nodes, *Unassisted- DTN (U-DTN)* in which DTN routing protocols (such as epidemic routing or probabilistic routing) are able to deliver data between all node pairs; and *Assistance-needed- DTN (A-DTN)* in which extra assistance (such as message ferrying) is needed because no space/space-time path can be found between some node pairs.

Our approach for whole network classification builds the network classification from components of node-pair classifications. In addition, we allow some flexibility in the definition where for example a network can be classified as an SPN even if some of the node-pairs are ST- or N-pairs.

Specifically we define the following:

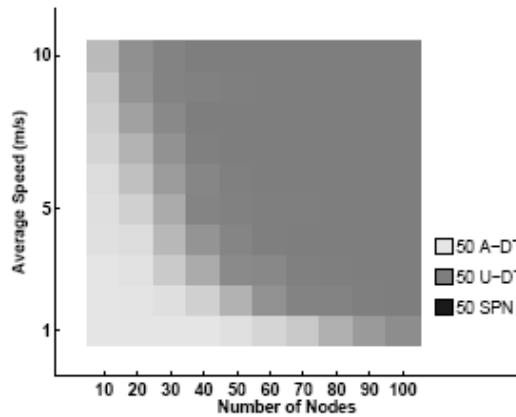
Definition: x-SPN: A network is classified as an x-SPN in an epoch if at least x% of the node pairs are classified as S-Pairs during that epoch.

Definition: x-U-DTN: A network is classified as an x-U-DTN in an epoch if at least x% of the node pairs are classified as either S-Pairs or ST-Pairs and the network is not classified as x-SPN during that epoch.

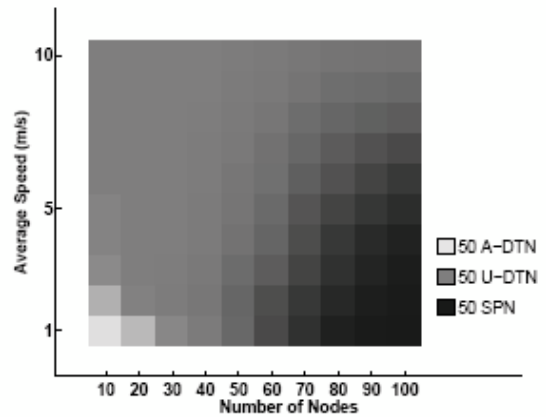
Definition: x-A-DTN: A network is classified as an x-A-DTN in an epoch if it is neither an x-SPN nor an x-U-DTN.

Similarly as in node pair classification, in order to capture the potential network character change over time, we classify the network in each epoch and the network classification over its lifetime is given by the percentage of time that it spends in each class.

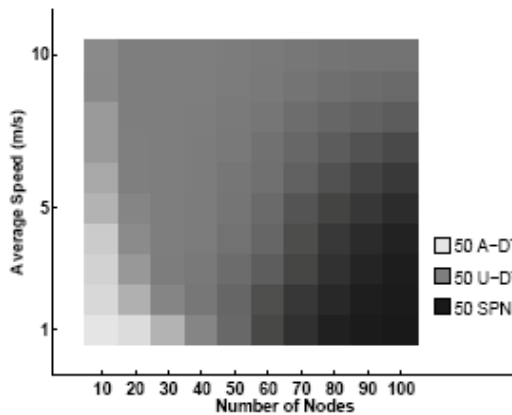
The figures below illustrate the type of result we are able to obtain from this classification framework.



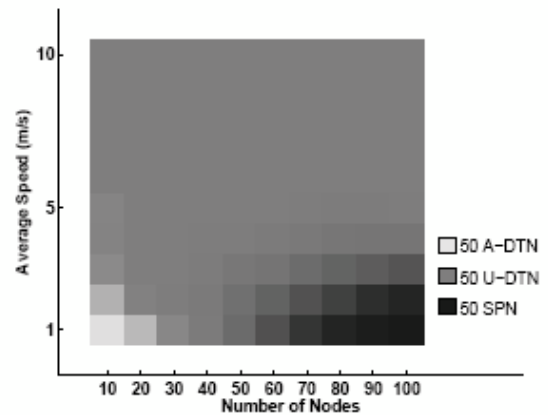
(a) $x = 100; \gamma = 600; \delta = 5$



(b) $x = 50; \gamma = 600; \delta = 5$



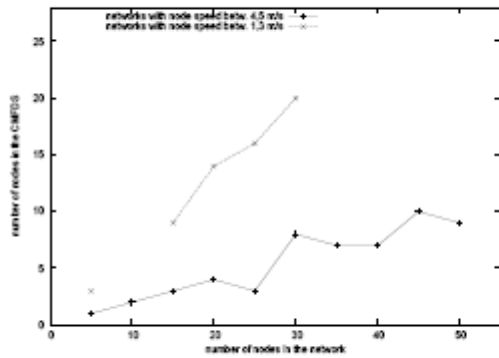
(c) $x = 50; \gamma = 300; \delta = 5$



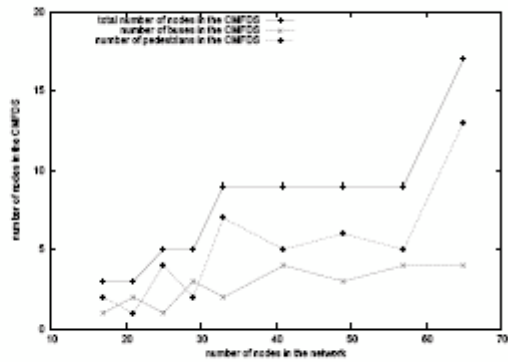
(d) $x = 50; \gamma = 600; \delta = 10$

10. Message Ferries as Generalized Dominating Sets in Intermittently-Connected Networks

Our work here provides definitions of a Connected Message Ferry Dominating Set (CMFDS) and provides algorithms for obtaining the CMFDS for a given definition of a ferry and for a given mobility trace. We have used the algorithm on several mobility traces to determine the types of results that one can expect from our work. Below is an example such result:



(a) The change in the size of the CMFDS as the number of nodes in a network increases (RWP model)



(b) The change in the size of the CMFDS as the number of nodes in a network increases (SPMB model)

11. Anonymous Routing in DTNs

The goal of ARDEN is to provide an efficient anonymous routing mechanism for DTNs. To achieve this, ARDEN builds upon a foundation of onion routing and modifies it for disconnected environments. The lack of long-lived paths between contacts makes the single path of onion routing susceptible to performance degradation in DTNs. Further, the complete knowledge of the path is hard to obtain in advance. ARDEN overcomes these issues by replacing single-node proxies with groups and through the use of multicast. Every node of a group can decrypt the corresponding layer of the wrapped bundle while only the intended destination can ultimately recover the final encrypted message. Through increasing communication redundancy provided by multicast, ARDEN manages to find a short path from senders to receivers. Figure 1 illustrates this strategy at a high level. Note that bundles are forwarded beyond their receivers to make receiver identification difficult for an adversary.

ARDEN relies on the presence of groups capable of de- crypting and forwarding traffic to its receiver. In order for this approach to work, it is necessary to divide nodes into groups. We rely on dynamic group partitioning, as it allows senders to specify groups as they see fit and prevents an adversary from necessarily learning group membership. The use of a public key/certificate-based approach to this problem makes this approach more difficult, as a DTN with n nodes would require the creation of $2n - 1$ groups, with nodes belonging to and therefore storing $2n-1$ certificates.

We instead make use of a group management mechanism based on Attribute-Based Encryption (ABE). ABE allows users to combine a set of semantically expressive attributes to generate public and private keys. In a social network, for instance, such primitives could be used to protect messages between group members (e.g., Above 6' tall, Basketball Player) and prevent users without these attributes from decrypting these messages. Moreover, these primitives prevent colluding adversaries (e.g., one Above 6' tall and one Basketball Player) from meaningfully combining their attributes and decrypting the contents of a message.

We specifically rely on a Ciphertext Policy Attribute-Based Encryption (CP-ABE) scheme for our construction, but refer to this approach simply as ABE throughout the work as other related techniques could provide similar protection with minor modifications to ARDEN. We then represent each user with a unique numerical identifier and assign attributes to that user based on the binary representation of that identifier.

ARDEN achieves sender anonymity in the presence of a passive adversary through three mechanisms. First, as messages are encrypted multiple times, an adversary can not extract the original message and link it to the sender. Second, because all nodes can act as sources, destinations and intermediary proxies, transmitting nodes are equally as likely to be forwarding messages as they are generating them. Finally, the use of anonymous reply paths makes it difficult for a receiver to identify the sender (unless the sender explicitly identifies itself in the message).

Scenarios in which nodes rarely (if ever) transmit improve the odds of a globally passive adversary correctly guessing the source of a message. Our current implementation of ARDEN does not account for this issue; however, it can easily be addressed through the occasional generation of dummy traffic. While we previously noted that the use of dummy traffic is not appropriate in DTNs in the presence of high volumes of legitimate traffic, the creation of dummy packets when ingress throughput drops below a threshold may help to prevent such attacks. For instance, if a node receives only one bundle per quantum, it can generate a garbage message destined for a random receiver. Such reactive cover traffic, which can be generated at different volumes, makes traffic analysis attacks extremely difficult without wasting contact bandwidth.

ARDEN also achieves receiver anonymity through three methods. Layered encryption allows only the ultimate receiver to determine the target of a message. Because the last layer of a message is encrypted using the private key of the destination, only the destination will know that it is the intended receiver. Moreover, multicasting prevents an adversary from determining the intended receiver on a hop-by-hop basis when multiple nodes are within transmission range. Finally, by placing the receiver randomly in the layers of encryption as opposed to the deepest layer as is traditionally done in onion routing, it becomes much more difficult for an adversary to determine when the true destination has actually received a message.

While robust against passive adversaries, it is important to understand how ARDEN withstands more active attacks. We therefore briefly explore a number of different attacks commonly discussed in this space.

As is the case for benign nodes, an adversary controlling a single randomly placed node will be unable to extract the source and destination of a message because of the use of multiple layers of encryption. An adversary able to compromise multiple nodes may not necessarily be more successful. For instance, compromising the nodes surrounding an arbitrary node will not reveal the source or destination of a transmitted message. An adversary will instead need to control nodes within every group along the path between a suspected source and destination in order to identify the source correctly – a physically far more demanding task. Note that because the destination node will continue to forward the bundle (potentially over multiple hops), such an attack is unlikely to successfully identify the destination of a message. An active adversary may also attempt to manipulate the bundles exchanged between legitimate nodes. For instance, an adversary may replay specific bundles to determine when that message is retransmitted by the intermediary node. By checking to see if an incoming message matches the stored HMAC of previous messages, such attacks are easily defeated by ARDEN. An adversary attempting to watermark specific

flows through the use of temporal perturbation attacks will also fail due to both the shuffling of message within ARDEN intermediary nodes and the store-and-forward nature of DTNs. Deducing the origin of a message through dropping attacks is similarly difficult given both the long retransmission timeouts and that such retransmissions will appear different from the original message.

12. iDTT: Delay Tolerant Data Transfer for P2P File Sharing Systems

iDTT is an overlay network made up of a set of dedicated servers (iServers) that are deployed to transfer delay tolerant data (called *bulks*). Each iServer has a large storage capacity and maintains connections to several other iServers. Bulks are transferred between connected iServers in a store-and-forward manner. Before being forwarded, data can reside in an iServer for periods that can range from several minutes to several hours. iDTT provides a delay tolerant data transfer service to users who connect to iServers over the Internet. Each user runs local client software called *iProxy* to establish control channels with the other end points, to exchange (send and receive) bulks with its nearby iServer, and to handle delay tolerant data transfer for user applications. We assume that the iProxy knows how to contact its local iServer with a hard-coded address or with an address obtained through other means (DNS for example).

When iDTT users exchange delay tolerant data through iDTT, the sender's iProxy first establishes a control channel with the receiver's iProxy to exchange some meta-data and then constructs and sends bulks to its connected iServer. The receiver's iProxy will receive the bulks from its local iServer and deliver them to the corresponding user applications.

We evaluate iDTT using two real network topologies, the GEANT and Abilene networks, and their corresponding traffic traces. The GEANT network has 23 routers and 74 directed links, while Abilene has 10 routers and 30 directed links. We use Emulab as the implementation testbed. The traffic traces are used to generate the background (non-iDTT) traffic. They are equally divided into time periods: the first period is used as historical information, while the second period is used to emulate current networks.

Applications: We report the results for two P2P applications: eMule and BitTorrent. We implement simplified versions of each of these applications. The first version uses direct end-to-end native network transmission while the second ones use the iDTT overlay as their data plane. No change is made to other mechanisms of these applications. Note that this may mean that they are not necessarily optimized for use with iDTT. For instance, some BitTorrent peers may be unfairly treated by the tit-for-tat mechanism because of the extra delay introduced by iDTT. We will investigate efficient iDTT-based P2P systems as part of our future work.

Performance Metrics: We use the following metrics:

- *Downloading time:* It measures the performance of P2P file sharing systems. It is the total time that a peer finishes downloading a file.
- *Link's 95-percentile traffic:* It evaluates the cost caused by the peak-time traffic on a link. Since the prices for incoming and outgoing traffic are different for inter-domain traffic, we treat the two directions of a link as two directed links.
- *The maximum storage usage:* Every iServer has temporary storage to store data. This metric assesses the resource required by iDTT.

Through these emulations, we demonstrate that this architecture can drastically reduce the peak-time usage of network links, while incurring low additional downloading time when compared with traditional E2E transmission.

Note: Some work reported in this report overlaps with work reported for NSF Award 0831714: NETS-NECO: The WAM Continuum: Unified Design and Operation for Wireless and Mobile Networks

ACTIVITIES:

Traditional networks, both wired and wireless, have the property that end-to-end paths between nodes are relatively stable. Furthermore, in the event of link or node problems, paths can be restored by the routing protocols on fairly short time scales. However, not all environments that require communication will allow the creation of stable end-to-end paths. For example, the aftermath of a severe earthquake disaster will include collapsed buildings, persons trapped in debris, damaged utilities and roads, as well as fires and secondary explosions. Under this situation, the ability to communicate, even at low rates, is extremely valuable for sharing vital information, such as the number and location of survivors and the activities of rescue workers. Alternative plausible scenarios can be constructed for similarly fragile environments: for example, a peace-keeping operation in a third-world country that has no fixed communication infrastructure and adversaries interested in disrupting communication.

While not the common case for networking, these environments do represent some of the most critical cases, where the ability to communicate can make a huge difference for human lives. To provide communication in these challenged environments, the network protocols must be explicitly designed to perform despite frequent disruptions in the availability and performance of network components (i.e., links and nodes). The resulting networks are termed *disruption tolerant*. (Some of the early literature used the term “delay tolerant”, perhaps reflecting an origin in interplanetary networking; we prefer the more general term “disruption tolerant”.) A typical Disruption Tolerant Network (DTN) consists of a set wireless nodes some or all of which are mobile. The nodes can range from small, sensor-sized nodes, to mid-sized devices carried by people, robots or vehicles, to larger, long-lasting, kiosk-style installations. The node mobility, placement and/or radio ranges are such that end-to-end paths do not exist between some or all the nodes. This may be an intrinsic feature of the networks due to terrain limitations or the difficulty in providing enough nodes, or it could be by design—to save battery power, for example. While DTNs may lack end-to-end paths at any instant in time, node mobility results in data *transfer opportunities* between pairs of nodes. The network uses the transfer opportunities non-contemporaneously to construct end-to-end paths using a *store-carry-and-forward* switching paradigm, where node mobility is leveraged to provide connectivity.

Fundamental research in Disruption Tolerant Networks (DTNs) is fairly new, and many efforts to date have primarily focused on the problems of finding point-to-point routes with varying levels of network predictability. In addition, there have been proposals for generalized DTN architectures. While network architecture and point-to-point routing are certainly an interesting and logical starting point, they are only a first step towards developing and understanding how to best use these new types of networks.

In this project, we focus on the construction of DTNs that go far beyond the task of finding unicast paths. Instead, our DTNs are robust under uncertainty and attack, and they are highly efficient in their use of the node and link resources, which will often be extremely limited.

Our activities during the first year of the project consisted of 1) design of techniques for providing novel functions within the message ferrying paradigm, 2) an initial investigation of the use of DTN technology to provide communication facilities in vehicle to vehicle networks, 3) use of “throwboxes” to enhance the capacity of DTNs, and 4) development of a framework and methodology for power management in DTNs. These last two activities were carried out in close collaboration with Brian Levine and Mark Corner at the University of Massachusetts at Amherst.

Our activities for the second year consisted of 1) development of a novel approach hybrid routing in clustered wireless networks, 2) a technique for incorporating rateless coding into routing in DTN networks, 3) the design and implementation of a DTN bundle specification-compliant system in C-Sharp which makes it suitable for smart-phones and PDAs.

This last activity is the beginning of an effort aimed at building a DTN testbed at Georgia Tech. The testbed will be based on the DTNrg DTN2 architecture and will be used to test and verify the schemes we develop in this project. This will give us testing capability that complements the Dieselnets testbed developed at UMass.

The activities are summarized below. The details can be found in the publications listed in this report:

1- Novel schemes using message ferrying

Disruption tolerant networks (DTNs) can remain partitioned for extended periods of time. Several schemes for routing in DTNs exist. A common theme across all of these schemes is that they use a *store-carry-and-forward* paradigm, where existing nodes in the network relay the data from source to destination nodes, in one or more hops, such that each node along the path receives the data from previous node and buffers (or stores) it. This node then carries the data for a while, and upon contact with other nodes, forwards the data. Schemes that rely on the intrinsic movement of nodes in the network for routing are referred to as “passive” schemes, yet others where the nodes move proactively to bridge network partitions are referred to as “active” schemes.

The Message Ferrying (MF) proposed by our group initially in 2003 and refined in subsequent efforts is an active scheme in which special mobile nodes called Message Ferries, are used for facilitating connectivity between nodes. The role of the message ferries (or simply ferries), is to visit the nodes in the network, and deliver data among them. This is an attractive model because of the versatility that it affords in terms of the kinds of networks that can be served, and the end-to-end message delivery characteristics that can be achieved. Additionally, it takes the burden of message routing away from the nodes, which may be desirable when nodes have limited energy and storage resources.

Our work in this project continues to consider novel aspects of the message ferrying paradigm as described below.

Multicast communication using message ferrying

In this work, we design a multicast architecture based on message ferrying for DTNs, aimed at achieving both timely and efficient delivery. We propose a set of multicast protocols combining message ferrying with different epidemic routing options. In particular, we develop adaptive protocols which adjust the epidemic routing options according to the number of group members. If the number of group members is small, we use only the ferry to deliver messages to the group members. When the number of group members increases, we enlarge the flooding scope accordingly. Using simulations, we evaluate our proposed protocols under various application/network conditions. We show that a single cure-all multicast solution for sparse MANETs is highly unlikely. In addition, the adaptive protocols which are able to adapt their behaviors according to the scenarios generally have good performance.

By combining the message ferrying and epidemic routing with different flood options, we propose a series of multicast protocols. The protocols enhance the straightforward multicast approach of Epidemic Routing (ER) (aka flooding):

1) *MFER*: This protocol is similar to ER except that the ferry helps to disseminate messages around the network. If the source, either a group member or not, wants to send a multicast message to the group, it passes the message to the ferry. Then the ferry “starts” the multicast session and every node can now participate in this dissemination process.

2) *MFGR*: In MFER, a certain level of overhead is inevitable due to the data exchange among non-members. If we limit the participants to be only the ferry and group members, we have the MF Group Routing (MFGR). In MFGR, message exchanges can only happen among the ferry and group members so that all the data transmissions are necessary. If the group size is large, frequent member-member contacts in addition to ferry-member contact can speed up the data delivery.

3) *Adaptive Schemes*: Our adaptive schemes rely on the fact that the performance of ER or MF depends on the group size. When the group size is small, the message ferrying scheme provides both high efficiency and reasonable message delivery latency. On the other hand, when the group size becomes large, the epidemic routing scheme is reasonably efficient and provides better message delivery latency performance.

Message ferry routing for mobile nodes

A key problem under the Message Ferrying paradigm is to design routes for the message ferries such that certain properties of end-to-end message delivery, such as, delay, and

loss, can be assured. This is a difficult problem when the nodes in the network move arbitrarily. The difficulty arises from the fact that as the nodes move in arbitrary fashion, we cannot be certain about their precise location at any given time, and consequently, we cannot be certain about their precise location at any given time, and consequently, we cannot be certain about their precise location at any given time. In previous work we devised a method for ferry route design when the nodes are mobile. However, the scheme requires that the nodes or the ferry move proactively to meet each other and in some cases require the use of long-distance radio to agree on a meeting location. Such communication may not always be feasible or desirable. Besides, such collaboration may disrupt the node mobility, which is undesirable as it may be dictated by non-communication needs.

Our key contribution through this work is an algorithm for designing message ferry routes in sparse networks where nodes have arbitrary movement, that does not require any online collaboration between the ferry and the nodes, and does not disrupt the mobility of the nodes. We call this method the Optimized Way-points (OPWP) ferry routing method. In this method we determine message ferry routes that comprise an ordered set of way-points and waiting times at these way-points, which are chosen carefully based on the mobility model of the mobile nodes in the network. Every time that the ferry traverses this route, it contacts every node in the network with a certain minimum probability. This node-ferry contact probability in-turn determines the distribution of node-ferry contact times, and ultimately the properties of end-to-end message delivery. The ferry can repeatedly follow the same OPWP route, without any changes, as long as the system remains in steady-state, i.e, the nodes continue to use the same mobility model, and the traffic demands do not change. Thus we do not require any kind of adaptability from the ferry once we program it to follow the route.

The main difficulty in designing message ferry routes for arbitrarily moving nodes is that we cannot correctly predict the location of these nodes, and hence it may not be possible to correctly and deterministically position the ferry to contact these nodes. Observe that if there exists a steady-state in the mobility model, such that the steady-state probability of presence of a mobile node in a specific sub-region (such as one defined by radio coverage area around a ferry), is nonzero, then we can contact the node with certainty if we wait for the node in that sub-region for long enough. However, for most mobility models, this probability approaches 1, only as the waiting time approaches infinity. Clearly, we cannot afford to wait for infinitely long, and hence, we cannot afford to contact the nodes with certainty.

However, as we show in this work that it is possible to meet the nodes with desired probability by waiting a finite amount of time at a sub-region, as long as the desired meeting probability is modest (i.e., sufficiently large but not approaching 1), and the steady state probability of node presence in that sub-region is substantial (i.e., not approaching zero). If the ferry is able to contact a particular node with some probability $0 < p \leq 1$, in every tour, then in expectation the ferry will meet the node in every $1/p$ tours. More specifically, the number of tours that the ferry may make between successive node-ferry contacts is a geometric random variable with probability p . This observation is key to the ferry route design algorithm that we develop in this work. The ferry route design process can be thought of as consisting of two steps. In the first step we choose way-points and the waiting times at these way-points so that we can meet each node with certain probability. In the second step we order these way-points together to form a

tour that is of minimum length. The time that it would take the ferry to go through the route is the sum of time that it takes to travel between each of the way-points, the waiting time at each way-point, and the time that the ferry spends to service the nodes. For simplicity, let's assume that it takes some constant T units of time to complete the tour. Then the expected time between successive contacts is T/p . In order to reduce the message delay and loss, we should try to minimize the ratio T/p , i.e, reduce T , or increase p , or do both. However, this is not straight forward. First note that in order to minimize T , we not only need to choose way-points that would require minimum waiting time, but also pick points that would yield shorter overall tour. So we have to solve the problem in a way that we optimize the two steps in ferry route design together. Secondly, note that reducing T/p presents a trade-off. Since the contact probability p increases only as waiting time increases, any attempt to increase p may inadvertently result in increasing the overall tour duration T . The end-to-end delay for a message has two components; (i) time that the message spends in the send buffer at the source waiting to be uploaded to the ferry, and (ii) the time that the message spends on the ferry until the ferry makes a successful contact with the destination node and downloads the message at the destination. Both of these components depend on the T/p ratio.

2- Using DTN schemes in V2V networks

Recently, vehicular networks (VANETs) have attracted much attention due to the emergence of Intelligent Transportation Systems (ITS). The various applications of ITS include automated toll-booths, safety messaging, roadside entertainment services and drive-thru internet. In the context of VANETs, vehicular communication is broadly classified into two classes: *vehicle-to-vehicle (v2v)* **and** *vehicle-to-roadside (v2r)* communication. In v2v communication, the vehicles relay data packets in a multi-hop manner e.g., in a safety messaging application. In the case of v2r communication, the vehicles communicate with base-stations along the roadway possibly to get pertinent information (e.g., conditions ahead on the roadway) or to avail of any number of services that are advertised by the base-station.

In this work, we consider the problem of roadside-to-roadside communication (r2r) communication. Specifically, we investigate the problem of communicating between two road-side entities (roadside sensors and cameras, base stations or stationary vehicles) that have no wired or wireless connectivity between them. The main approach is to use the mobility of vehicles between these two roadside entities to relay data between them. We consider a roadside *source* desiring to send data to a roadside *sink*. This technique is precisely the DTN store-carry-and-forward paradigm. This work, thus considers the use of DTN techniques in the context of v2v networks. Our aim is to explore the achievable throughput between the roadside entities as a function of the vehicle arrival rates and their velocities. In addition, we will also present protocols that can be employed at the source and sink to facilitate the data relaying.

The motivation for transferring data between points along the roadway are manifold. For example, consider an accident that occurs at some point on the roadway. Then, a data relaying scheme would facilitate transfer of data between a vehicle located at the site of the emergency and another (stationary) vehicle or base-station located further down the roadway. Data-relaying over vehicular networks via v2r communication can also reduce the cost incurred in setting up a backhaul network to connect roadside components of an ITS network. The data relaying via the vehicles also provides an alternate channel in case of failure of the links of the backhaul network.

We consider two roadside entities on the roadway referred to as *source* and *sink*. We wish to provide connectivity between the source and the sink by employing vehicles as data relays. The data transfer between the vehicles and the source/sink is accomplished via vehicle-to-roadside communication. The vehicles express their intent of acting as data relays to the source (or of uploading data to the sink) by sending *beacons* to the source and the sink. The beacons containing information such as the current velocity of the vehicle, the transmit and receive power and receive sensitivity, if such information is required by the protocol. The duration of time during which the vehicle and the source/sink are within transmission range is referred to as the *contact time*. The vehicles whose position relative to the source/sink is such that the contact time is sufficient to initiate a transfer of a reasonable amount of data are referred to as *eligible* vehicles.

The time spent by the source/sink in activities other than transfer of data (for relaying) with the vehicles is referred to as the *idle time* (refer Figure 3). The *control channel* is used by the vehicles to send beacons to the base-stations and receive protocol messages from the base-stations, e.g., the base-station's beacon timer or the SOLICIT message. The *data channel* is used for the actual transfer of data between vehicles and the base stations.

In this work we propose and evaluate three variations on this basic idea for relaying. The protocols differ in how they use the available channels and in whether the vehicles or the road-side entities initiate communication.

3- Use of throwboxes to enhance DTN data carrying capacity

DTNs rely on the inherent mobility in the network to deliver packets around frequent and extended network partitions using the store-carry-and-forward paradigm. While such networks can only support delay-insensitive applications, such as messaging, file transfer, and data dissemination, they enable communication where otherwise there may be none. Even though DTNs are highly robust to poor connectivity, their performance is highly dependent on chance encounters between nodes. For the network to route a packet between two nodes that never meet, a transitive series of meetings, each of sufficient duration, must occur. When opportunities are missed, there is a decrease in throughput and an increase in delay in the network. Consequently, if a network designer can engineer a greater number of contact opportunities then the performance of the network can be greatly enhanced. In this work we propose the use of *throwboxes* to accomplish this goal. Throwboxes are small and inexpensive devices equipped with wireless interfaces and storage. Throwboxes are stationary, thus when two nodes pass by the same

location at different times, the throwbox acts as a relay, creating a contact opportunity where none existed before.

Throwboxes must be designed to have high availability, be able to transmit data at high data rates, have sufficient processing power to handle such data rates, and be energy efficient. Several platforms exist that meet some or all of these requirements. Commercially, the Intel Stargate is a reasonably efficient and powerful device. However, prior work at UMass created a prototype device that is closer to our assumptions. The Triage platform and software consist of a coupled Stargate board and MicaZ mote. The Triage software reduces the amount of time that the device must spend operating the high-power Stargate board, running tasks on the MicaZ whenever possible. Using a hailing radio, passing nodes can trigger the always-on MicaZ to power on the Stargate board and 802.11 CF network interface. In this way, Triage never misses a passing node but remains energy efficient. Solar panels recharge batteries to ensure long-term use. Given the current technological advances in processor and storage, we expect the cost and size of such devices, as well as their energy efficiency, to continually decrease.

This work investigates algorithms for adding throwboxes to a running DTN. Placing throwboxes into an operational network also creates an opportunity to modify the routing to utilize the throwboxes effectively. However, the addition of the throwboxes affects the flow of data through the network, which consequently affects the placement of throwboxes. While placement can be considered in isolation, we show that it is most effective to consider the routing algorithm simultaneously with the placement of throwboxes, thus maximizing the overall effectiveness of the throwboxes after they are deployed.

A defining characteristic of DTNs is that it is difficult to gather data about the performance of the network itself. Information about contact opportunities and traffic load may not be available as input to deployment algorithms. Similarly, algorithms for routing can vary in sophistication from simple epidemic replication to more efficient strategies that do not replicate messages and balance congestion across multiple paths. Specifically, we consider three different deployment scenarios that differ by what information is available, and for each we evaluate three different routing scenarios that differ in their sophistication. Fig. 4 illustrates these nine cases. For deployment, the three cases are:

- Contact and traffic-based (CTB): both contact opportunity and traffic demand information are input to the deployment algorithm,
- Contact-based: information about contact opportunities is input to the deployment algorithm;
- Oblivious: no information about contact opportunities or traffic matrix is available for the deployment algorithm.

In the simplest case, routing in DTNs is based on epidemic style replication of data that is oblivious to information about traffic load and contact opportunities. With more sophistication, an algorithm can take these data as input and compute the optimal **single**

path from source to destination without using replication. Improved performance is possible when different messages are routed each along different single paths (without replication) to make use of more resources in the network, which we call multipath routing. These routing algorithms are optimal as they have complete knowledge of the contact opportunities and traffic demand. We prefer these solutions because they show the upper bound on throwbox performance. This is independent of any particular routing algorithm and only dependent on the single versus multi path assumption.

4- Power management in DTNs

DTNs have as an important component untethered devices with limited energy supplies. Without careful management, therefore, depleted energy supplies will degrade network connectivity and counteract the robustness gained by mobility. While energy savings is necessary, mobile devices exhibit a tension between saving energy and providing connectivity through opportunistic encounters. In order to pass messages from node to node, the device must discover other nodes .typically the discovery is done using the same wireless interface used for message transfer. At the same time, energy savings necessitates disabling (i.e., sleeping) the wireless device.the wireless interface is one of the largest energy consumers in mobile devices whether they are actively communicating or just listening. If the wireless interface is asleep, the node cannot discover other nodes for communication and other nodes cannot discover it, either. The time periods when two nodes can communicate with each other are called *contacts*. When networks are partitioned most of the time, it is not trivial to discover contacts while also saving energy. Therefore, designing power management for DTNs in this way is challenging since a node needs to detect when it can communicate with other nodes while aggressively disabling its radio during the remaining periods.

In previous work (published in IEEE SECON 2005), we have shown how nodes can effectively use statistical information about network connectivity to predict when to enable their wireless interfaces and search for contacts. However, such prediction could save considerable amount of energy only when the network connectivity has a certain degree of regularity. Thus, a network with significant randomness in node mobility requires better mechanisms to save energy while delivering messages.

In this work, we examine the possibility of using a hierarchical radio architecture in mobile DTNs, in which nodes are equipped with two complementary radios: a long-range, high-power radio and a short-range, low-power radio. In this architecture, energy can be conserved by using the low-power radio to discover contacts with other nodes and then wake up the high-power radio to undertake the data transmission. Previous studies using this hierarchical radio architecture have considered only densely deployed networks, in which the short range of the low power radio is sufficient to discover each other. However, a node in DTNs is often far away from the rest of the network. Therefore, if a node relies only on the low power radio to discover contacts, it may miss them due to the shorter range. To avoid missing too many contacts, we propose a generalized power management scheme that allows the main high power radio to participate in contact discovery, but searches for contacts less frequently than the low power radio. Each radio controls its frequency by a time interval, called *wake-up interval*,

to wake up and search for contacts. This wake-up interval can be used as a tuning parameter to trade between energy savings and the performance of message delivery. In addition, for the case when traffic load can be predicted, we devise approximation algorithms to determine the optimal wake-up intervals that discover enough contacts to handle a given traffic load, while minimizing the overall energy consumption.

In a mobile DTN, if the devices are equipped with multiple radios, the nodes may or may not discover a particular contact opportunity depending on the range of the radios, which radios are active, and the movement trajectory of the two nodes. While the short range radio may discover less contact opportunities than the long range radio, it does so at substantially reduced energy costs. The energy consumption of each wireless interface depends on whether it is *transmitting*, *receiving*, *idling* (when listening to the wireless medium without transmitting nor receiving), or *sleeping* (when the wireless interface is disabled). When sleeping, a node consumes an order of magnitude less energy than when idling, while an idling node consumes energy at the same order of magnitude as a receiving or transmitting node. In addition, the low power radio consumes an order of magnitude less energy than the high power radio for each activity. Thus, it can discover contacts using substantially less power. However, its outdoor range is limited relative to the high power radio.

In this work, we consider a DTN consisting of mobile nodes as well as stationary nodes, which have two radio interfaces: one with a long radio range and high-power, e.g., a 802.11 wireless card, and the other with a short radio range and low-power, e.g., a CC1000 radio. We only account for the communication energy consumption of a wireless interface and do not consider other sources such as computation or mobility. Also, we assume that nodes have no a-priori knowledge about other nodes' mobility and contacts must be discovered by one or both radios of the nodes. To discover contacts, a radio broadcasts messages called *beacons* periodically. To save energy while discovering contacts, a radio has three power management modes: *search*, *contact*, and *dormant modes*. In the search mode, the radio wakes up periodically to discover a contact. This period is called a *wake-up interval*. In the contact mode, the radio stays awake to exchange messages with other nodes that it previously discovered in the search mode. In the dormant mode, the radio is not used and remains asleep.

Given that the high-power radio is always used, we develop and evaluate four possible variations of this general framework. The *Continuous Aware Mechanism (CAM)* uses only the high-power radio of a node. In CAM, the high power radio always stays awake to search for other nodes. The *Power Saving Mechanism (PSM)* also uses only the high-power radio, however it alternates between asleep and awake while discovering contacts. While similar, note that this PSM is not the same as 802.11's PSM mode. The *Short-rangeradio-dependent Power Saving Mechanism (SPSM)* uses both lowpower and high-power radios of a node. In this mechanism, the low-power radio alternates between sleeping and waking up to discover contacts, while the high-power radio sleeps and is awakened by the low-power radio only after discovering a contact. Finally, the *Generalized Power Saving Mechanism (GPSM)* uses both the low power and high-power radios of a node. In this mechanism, both radios alternate between sleeping and waking up to discover contacts. If a contact is discovered by the low-power radio, the high-power

radio is awakened. If a contact is discovered by the high-power radio, the radio stays awake as long as it has contact with the other node.

The critical issue in all of the power management mechanisms is determining the proper wake-up interval. In each mechanism, the wake-up intervals can be used to trade between energy and delivery performance. Specifically, when the wake-up intervals are long, energy can be saved at the cost of missing contacts, which in turn results in poor delivery performance. When the wake-up intervals are short, more energy will be expended while discovering more contacts and improving delivery performance. The key lies in discovering just enough contacts to deliver a node's messages.

We state the problem as follows: *For each node, find the wake-up intervals that leads to discover enough contacts to deliver the expected traffic load while minimizing energy consumption.*

To formulate the problem, we assume that statistical information about contacts and traffic load between each pair of nodes is available. This information is often already available in a DTN: nodes observe and exchange the history of contacts and traffic load to make efficient routing decisions. To address the problem, we define the *contact arrival rate* as the number of contacts between two nodes over a unit time and *expected bandwidth* as the maximum amount of messages that can be delivered by the discovered contacts between two nodes over a unit time. With these definitions, we can estimate the expected bandwidth for given wake-up intervals using contact duration and contact arrival rate per pair of nodes. Unfortunately, the distribution of inter-arrival times and contact durations is generally not known, especially for many common mobility models. Without this, it is not possible to develop general, optimal algorithms. Therefore, we devise an approximation by *measuring* contact durations from a mobility model and assume that contacts arrive according to a Poisson process. We validate this approximation and show that it works well in practice.

5. Message Ferrying in Clustered Wireless Networks

Both MANET and DTN routing protocols focus on cases in which nodes' spatial distribution is homogeneous, i.e., either an end-to-end path can always be found or nodes are completely isolated from others. In this work we study routing in clustered DTNs where nodes are partitioned into clusters. Clustered DTNs may arise in a variety of situations. For example, a sensor network can be deployed around hot spots. Although sensors at the same spot are connected and collaborate in sensing and data collection, there is a lack for direct communication between sensors in different spots. In addition, heterogeneous spatial node distribution was observed in real vehicular network traces. For these scenarios, MANET routing alone would fail to deliver data between different clusters. On the other hand, existing DTN routing protocols only consider direct contacts between nodes and is not able to exploit multi-hop connectivity within each cluster.

In this work, we consider networks with a single ferry moving at a constant speed. The ferry has sufficient storage so that no message is dropped due to buffer overflows. While the store-carry-and-forward scheme is utilized among clusters, regular MANET routing can be used within each cluster to exploit local connectivity. In such a hybrid routing

scenario, traffic may be carried via multiple routing paradigms. This raises a question of how message ferrying and MANET routing interact with each other. We consider a hybrid routing approach in which the ferry only communicates with gateway nodes in each cluster, which aggregate traffic from nodes in the cluster and represent them when communicating with the ferry. As we will point out in the next section, this approach reduces wireless interference by using shorter hops and fewer number of concurrent transmissions towards the ferry.

Ferry routes may be fixed due to application requirements. For example, the ferry can be a public bus whose movement is determined by passenger transportation considerations. In such cases, contacts between gateways and the ferry can be limited. To best utilize gateways, we study the issues of data aggregation at gateways and transmission scheduling between gateways and the ferry, both can significantly affect data delivery performance. We develop various data aggregation and scheduling algorithms and evaluate them in scenarios when ferry routes are fixed. Alternatively, ferry movement can be adjusted for communication purposes, e.g., mobile robots. To achieve further performance improvement, we continue to study the problem of customizing ferry movement according to node location and traffic conditions. Using extensive simulations, we evaluate various hybrid routing schemes in clustered DTNs.

6. Use of Rateless Coding in DTNs

In this work, we investigate the problem of reliable and low-latency multiple single-source single-destination (uni-cast) message delivery in DTNs. Epidemic routing, aimed at minimizing latency, has been suggested as a viable solution to the problem of message delivery in DTNs. Here, multiple identical copies of messages are injected into the network, and node mobility is relied upon to transfer all the requisite data packets to the destination with high probability. An intermediate node (other than the source and destination, called a relay node) transfers a copy of its packets to a node that it is in contact with if the latter does not already have a copy of the same. Although considerable research is available for efficient routing of messages in DTNs, most of them employ simple replication of packets and multiple transmissions to ensure higher reliability and lower latency. Recently, hybrid schemes, wherein both replication of messages and simple erasure-coding are employed, were shown to have great potential as efficient solutions for DTNs. It was shown in that such hybrid routing strategies employing both coding and replication are more robust than simple replication. However, these works are either very simplistic or do not have realistic assumptions. For example, in recent work considered schemes where data is first encoded with a replication factor of r and then packetized into $s*r$ chunks for some integer s . They are then relayed in a two-hop fashion. Simulations reveal the superiority of this scheme over other schemes such as simple replication.

There are several drawbacks with this approach. First, although packetizing the replicated data ensures that the chunks have a smaller size than the original message, in order to ensure that all possible opportunities to disperse data are used, these chunks must be

fairly small, equivalently, s must be large. However, this was not considered. Second, employing a fixed-rate erasure coding scheme makes it natural to ask, “what rate is optimal?”. Clearly, the higher the rate, the better, but it comes with the cost of using more network resources such as bandwidth. Moreover, realistic assumptions such as finite packet expiry and time-varying channel losses make the scheme practically unusable since they would require the scheme to be rate-adaptable.

In this work, we show that a new class of packet-level codes called rateless codes can be employed for unicast communication in DTNs in order to obtain a significantly better reliability and delivery delay performance compared to existing schemes. Several aspects of rateless codes make them apt for such applications. First, their rateless nature does away with issues regarding good choice of rates even in the presence of varying channel loss conditions. Second, they are packet-level codes that have low complexity of encoding and decoding and require very low coding overhead to re-cover the message. We show using extensive simulations under both real-world trace data such as UMassDieselNet testbed and simulated trace data from the area-based random waypoint model that our coding scheme offers greater reliability and lower latency than other coded and uncoded schemes. We see from simulations that our scheme suffers far less degradation in performance for the same decrease in packet expiry times or increase in message sizes when compared to others.

In this work, we concern ourselves with the benefits offered by rateless coding in DTNs. We do not assume any knowledge of the paths between nodes, or any statistics of the duration and inter-contact waiting times between any pair of nodes. However, during contacts between nodes, it is assumed that channel losses are absent. It is certain that coding would further enhance the performance of our proposed scheme relative to other alternatives when channel losses are present. The main aim of our approach is to increase reliability and reduce latency even in lossless channel conditions.

7. DTN sharp: A .Net implementation of the DTN bundle spec

As a first step to our DTN testbed goal, we have undertaken is an implementation of the DTN Research Group’s bundle forwarding architecture in C# on top of Microsoft’s .NET platform. The DTNRG bundle forwarding architecture consists of three major components: routing, convergence, and storage facilities. These cover, respectively, deciding when and how bundles are to be transmitted between nodes, actually transmitting bundles from one hop to the next over whatever underlying network infrastructure exists, and storing bundles for forwarding at some point in the future (possibly with a rich set of metadata to allow for highly specialized routing schemes).

The implementation was designed with flexibility as its primary design constraint. The goal is to provide a system that would allow researchers to easily add new storage, routing, and convergence facilities without requiring a thorough understanding of the particulars of the implementation (which are in many cases largely not relevant to the research question under consideration). Currently it consists of a functional DTN

forwarder with a set of primitive data stores and a growing set of routing and convergence facilities. At present work is focused on development of additional convergence and routing facilities as well as development of applications that fully demonstrate the enhancements that DTN offers over conventional network deployments.

In line with the goal of designing toward flexibility, the system consists of a relatively small set of core functionality that provides structure, organization, and data flow. This core does is not sufficient to function as a DTN forwarder, however it is supported by a set of plug-in modules that provide the actual routing, convergence, and storage functions. This modular design allows researchers working in the DTN space to test ideas quickly and easily by implementing them as new plug-ins with no modification to the stable core classes. A researcher who is only concerned with routing can avoid a dependence on data storage or bundle transmission, and can instead work in the abstracted space of a contact graph while still producing functional code. Additionally, the project employs an XML based protocol for delay tolerant applications. This means that developers can work in virtually any available programming language and leverage language features and libraries that may not have larger availability. By using Microsoft's .NET framework, the code is able to run (with no modification) on desktop and server Windows systems (XP, Server 2003, Vista), Windows Mobile systems (Pocket PC, Smartphone, Windows CE), and Unix-like systems (by way of the Mono Project).

Interoperability was last thoroughly examined in November 2006 at the IETF meeting in San Diego. As of that writing DTN Sharp was able to communicate via UDP with all other implementations present (including the DTNRG reference implementation). We support the generation and processing of status reports in accordance with the Bundle Protocol Specification and are able to participate as a first class node as part of a mixed-implementation DTN installation. As of this writing DTNSharp is compliant with the latest revisions to the draft Bundle Protocol Specification, and work is essentially completed on an implementation of a TCP convergence layer outlined in IETF Internet Draft draft-demmer-dtnrg-tcp-clayer-00.

8. Roadside-to-Roadside Data Transfer Using Vehicular Traffic

Vehicular networks can be designed with many objectives in mind. Two primary varieties have been explored in the literature: 1) vehicle-to-vehicle (v2v) communication in which vehicles communicate with each other in support of safety or information dissemination applications, and 2) vehicle-to-roadside applications in which vehicles exchange data with roadside kiosks in support of information or data gathering applications. In this work we consider how vehicular networking technology can be used to support *roadside-to-roadside* (r2r) communications. In this form of communication vehicles traveling along a road or highway are used to transport data between two fixed roadside locations that are too far apart to be connected. The basic idea is to have a roadside station give data to a moving vehicle as it gets close and then have the vehicle

carry and then deliver the data to the other roadside station. This same idea is the basis of many data delivery protocols in Disruption Tolerant Networks (DTNs). Here we explore its use in the mobility context of vehicular networks.

Our problem can be stated as follows: given two roadside stations R1 and R2 with a distance between them that can reach to tens of Kilometers, and given that at least one of the two stations does not have either a wired or a wireless connection to the Internet, how can we make use of both the wireless and storage capabilities in the vehicles traveling on the road to provide a reliable data transfer service between R1 and R2. For the purposes of this work we assume that both roadside stations as well as some of the vehicles have 802.11 radios.

There are many possible applications that could make use of such roadside-to-roadside service. One example) is where an ambulance at a remote accident site uses the vehicles to send and receive data to and from an access point that is connected to the Internet. Another possible use of an r2r service could be the collection of sensor data or images from a roadside station to a delivery point.

We first considered the feasibility of such a service in previous effort in this project where our goal was to understand the throughput achievable in such an environment and to consider the design of specific MAC protocols in that environment. In this work we aim to take concrete steps towards making the deployment of such a service real and useable. We, therefore, focus on two main aspects of such a service:

- The design of the data transfer mechanisms between the roadside stations and the vehicles assuming a standard 802.11 MAC protocol is in use (as opposed to redesigning the MAC protocol).
- The design of schemes to insure reliable data transfer between the two roadside stations over the particularly challenging channel provided by the moving vehicles.

9. Understanding the Wireless and Mobile Network Space

Wireless data networks with mobile nodes have been the subject of extensive research for at least three decades now. Research into such networks has mostly considered networks called Mobile Ad Hoc Networks (or MANETs). While the nodes in such networks are mobile, it is generally assumed that end-to-end, possibly multi-hop paths between node pairs exist most of the time. Routing protocols designed to operate in MANETs assume that these paths are formed by a set of wireless links that exist contemporaneously. It is also assumed that if these paths are disrupted because of node mobility, then this disruption is only temporary and the same or alternate paths are restored relatively quickly.

Disruption or delay tolerant networks (DTNs) are a form of wireless and mobile networks that has received significant attention recently. Their primary distinction from MANETs

is the fact that in DTNs links on an end-to-end path may not exist contemporaneously and intermediate nodes may need to store data waiting for opportunities to transfer data towards its destination. We call such paths *space-time paths* to distinguish them from contemporaneous *space paths* used in MANETs. To deliver data in DTNs new routing protocols that are quite different from those used in MANETs have been developed.

For any particular network, the question of whether the network is a MANET or a DTN is important to answer as it will influence its design and operation. In reality such a question is hard to formulate and even harder to answer as many networks will not fit neatly within a simple classification scheme. How a network is classified depends on several factors. Most important are the size of the network, the geographical area covered by the network, the node mobility pattern, and the range of wireless radios. Except for some extreme cases, it is in general not obvious, given these network parameters, as to which class a particular network belongs to.

This work is concerned with developing a formal classification of mobile and wireless networks. The goal is to have the classification be usable to determine the most appropriate routing strategy for a network. We call this a *routing-centered classification*. We also aim to develop a methodology that allows us to perform this classification given network characteristics. Note that our objective is to have the network classification provide guidance regarding which *class* of routing protocol (e.g., MANET, or DTN) is feasible. Further specification of the routing protocol would be needed within the specific class indicated but beyond what our classification informs. This will typically require additional information that is beyond the scope of our classification such as traffic and reliability requirements.

7.1 The basic idea

We already mentioned two main classes of wireless and mobile networks, namely MANETs and DTNs. MANETs are characterized by the availability of space paths and DTNs by the availability of space-time paths. Space paths are actually a special case of space-time paths in which all the links exist simultaneously. Because of this, it can be argued that MANETs are actually a special case of DTNs. In fact, it is easy to see that DTN routing protocols are perfectly usable in MANETs.

DTNs are, in turn, actually a special case of a more general class of networks in which space-time paths may not exist. For example, a network with nodes that are sparsely deployed and move in limited regions does not provide end-to-end space-time paths. In such networks data delivery is simply not possible between node pairs. Networks of this type require additional *assistance* in order to enable paths (space or space-time) for data delivery. Proposals for the use of message ferries or throwboxes] are motivated by this type of network. It should be noted, however, that message ferrying and throwboxes while initially motivated by this type of sparse network are perfectly usable in regular DTNs or MANETs.

To describe the network classes above we will first of all use the term *Space-Path Networks (SPNs)* to denote what we have been calling MANETs so far. We do this because the term “MANET” is currently overloaded in the literature to indicate both a network path characterization as well as the type of routing protocols used. Our

terminology emphasizes the path behavior of MANETs that we are interested in without implying the use of any particular routing protocol.

We use the term *unassisted DTN or U-DTN* to describe networks which provide space-time paths between all node pairs. Note that the U-DTN class includes the SPN class. We use the term *strict U-DTN* to describe networks in the U-DTN class but not in the SPN class. Networks that do not provide space-time paths between some or all the nodes (or alternatively whose space-time paths take an infinite amount of time to complete) are called *assistance-needed DTNs or A-DTNs*. The A-DTN class includes the U-DTN class. Here again we use the term *strict A-DTN* to describe networks in the A-DTN class that are not in the U-DTN class.

Note that while the network classification above is based on path properties it also is intended to inform routing protocol design. Traditional MANET protocols are usable in networks belonging to the SPN class and perform poorly for networks outside the class. Of course, exactly which MANET protocol is best cannot be specified with this type of classification. DTN routing protocols like epidemic routing are usable in the entire U-DTN class (including the SPN class).

Assistance (like Message Ferrying) is required in the strict A-DTN class but is usable and sometimes beneficial in the entire A-DTN class (including networks in U-DTN and SPN classes). Again exactly which form of assistance or how it should be designed (e.g., how a ferry route should be designed) is not informed by our classification and requires additional information beyond what we use in our classification.

10. A Framework for Characterizing the Wireless and Mobile Network Continuum

Wireless data networks with mobile nodes have been the subject of extensive research for at least three decades. Interest in such networks is almost as old as the Internet itself. Early efforts focused on networks frequently called Mobile Ad Hoc Networks (MANETs). While the nodes in such networks are mobile, it is generally assumed that end-to-end, multi-hop paths between node pairs exist most of the time. That is, end-to-end connectivity is the norm. Routing protocols designed to operate in MANETs assume that these paths are formed by a set of wireless links that exist contemporaneously. If MANET paths are disrupted because of node mobility, then this disruption is only temporary, and identical or alternate paths are restored relatively quickly.

Recent research has focused on a different paradigm for wireless and mobile networks. This paradigm goes by various names: disruption-tolerant, delay-tolerant, opportunistic or intermittently-connected networks. We will use the “DTN” acronym to refer to this class of networks. DTNs differ from MANETs in end-to-end connectivity. In a DTN, the links on an end-to-end path may not exist contemporaneously and intermediate nodes may need to store data waiting for opportunities to transfer data towards its destination. That is, end-to-end connectivity is the exception. We call end-to-end DTN paths *space-time paths* to distinguish them from contemporaneous *space paths* used in MANETs.

For any particular network, the question of whether it is a MANET or a DTN (or equivalently a network with space paths or space-time paths) is critical to answer. Protocols developed for MANETs generally do not work in DTNs and vice versa, since the connectivity assumptions are so different; hence categorizing a network is critical to its effective operation. In practice, the question of network category is challenging to formulate and answer. Many networks will not fit neatly within a simple classification scheme and/or they may change their classification over time. Informally, how a network is classified depends on network *features* such as the size of the network, the geographical area covered by the network, and the node mobility pattern. Additional factors such as traffic pattern and application requirements, also often affect how a network is classified. Except for some extreme cases, it is generally not *a priori* obvious as to which class a particular network belongs.

This work aims to take a fresh perspective on classifying wireless and mobile networks (WAMs) networks. We develop the WAM continuum framework which provides a unified treatment of wireless and mobile networks. Our main premise is that it is best to consider the space of WAMs as a *continuum*. A particular network is characterized by its position in this continuum, which may change over time. Our characterization can apply to an entire network or can be different for different parts of the network (e.g., different node pairs). Certain *network equivalence classes* can be defined over subsets of this WAM continuum and such network classification can be used to help guide WAM design and operation.

The unified view provided by our framework has several advantages.

- 1) it formalizes the widely held informal notion that wireless and mobile network classes such as MANETs and DTNs that have often received distinct treatment in the literature are “related”.
- 2) it provides a framework for understanding networks that either don’t fit neatly into one category or can change classification over time.
- 3) it can form the core of a model of operation in which networks can adapt over time based on changes in their position in the WAM continuum.
- 4) a carefully constructed classification framework provides insights into network performance.

Our framework provides the means by which one can describe connectivity properties in a WAM. In networks where data is routed on contemporaneous end-to-end paths, it is simple to discuss connectivity properties. The network (or more specifically a node pair) is either connected or not. In DTNs, the concept of space-time path connectivity is more complex and needs to incorporate notions of connectivity over time. As a result, the problem of developing a unifying framework for WAMs presents many challenges, including developing a formalism for specifying network classes, incorporating

parameters that reflect some level of abstraction of protocols within the classification framework, and developing procedures and algorithms for performing classification based on network and protocol parameters and features. We tackle each of these challenges in this work.

In this paper we present a specific instantiation of our framework which provides guidance on what type of routing protocol is feasible for WAM networks. The goal is to have the classification be usable to determine the most appropriate routing strategy for a network. This new work admits granularity at the level of node-pairs, thus accommodating classification of portions of the network as well as the network as a whole. We also develop a methodology that allows us to perform and analyze this classification given network characteristics.

We then extend this classification to consider the issue of energy availability, clearly important in the design and operation of many WAM networks. Using the WAM framework, we are able to classify a WAM's energy "sufficiency" depending on a combination of the network connectivity properties, available energy, and power management scheme.

We illustrate our approach by applying it to networks described by traces from real platforms and mobility models. Our results demonstrate the power of our unifying approach in providing significant insight into the effect of network parameters and features on network character and performance. They also show how instances of our framework can be used as a systematic and formal descriptive and evaluative tool.

11. Message Ferries as Generalized Dominating Sets in Intermittently-Connected Networks

Message ferrying is a technique used for routing data in wireless and mobile networks in which one or more (usually) mobile nodes are tasked with delivering data between sources and destinations. In general, multiple ferries may be used to achieve connectivity. Each ferry provides connectivity for a subset of the nodes and communicates with other ferries through ferry-to-ferry contacts. One can see that if the ferries are regularly in proximity to one another, then the overall network can provide end-to-end paths between all nodes by traversing one or more ferries. We say this network is a "connected message ferry network", understanding that the connectivity is not instantaneous in time, but requires the message ferries to carry messages and wait for the necessary proximity.

While useful as a routing technique for wireless and mobile networks in general, message ferrying is particularly useful in intermittently connected networks where the links on an end-to-end path may not exist contemporaneously and intermediate nodes may need to store data waiting for opportunities to transfer it towards its destination. Message ferrying is among a set of routing techniques that have been developed for such networks. Intermittently connected networks are representable by *evolving graphs* which provide so-called *space-time paths* between sources and destinations

Message ferrying is a rich design space. Message ferrying can be performed by nodes already in the network (*intrinsic message ferries*) or by nodes added explicitly for such a task. Additionally, a message ferry's mobility may be controlled to improve its ferrying performance or may be uncontrolled, allowing the message ferrying functions to be performed through the natural mobility of a node. We say that a wireless and mobile network possesses *intrinsic message ferry capability* if a subset of the nodes can act as message ferries by virtue of their own mobility pattern, without introducing additional nodes or modifying existing node mobility. Our work in is concerned with characterizing such intrinsic capability for a given wireless and mobile network. In particular we are interested in determining for a given network's mobility pattern which subset of nodes, if any, can act as intrinsic message ferries, individually or collectively. Answering such a question for a particular network allows the design of a routing strategy for the network using the set of message ferries that has been identified. On a longer time scale, the lack of sufficient intrinsic message ferries could be used to trigger the dispatch of additional nodes to boost connectivity.

Interestingly, it turns out that identifying sufficient intrinsic message ferries is a generalization of the well-known connected dominating set discovery problem. In effect, a set of intrinsic message ferries that can be used to provide end-to-end connectivity in an intermittently connected network is analogous to a connected dominating set (CDS) in a connected (non time-varying) graph. Since a connected network is a special case of an intermittently connected network, finding a CDS is also a special case of finding an intrinsic message ferry set. Thus our work fits within a broader theme of unifying how researchers think about mobile and non-mobile networks.

In general, a dominating set (DS) of a graph $G = (V, E)$ is a subset $V' \subseteq V$ such that each node in $V - V'$ is adjacent to some node in V' . A *connected dominating set* (CDS) is a dominating set which also induces a connected subgraph. It has been shown that the presence of a connected dominating set in an ad hoc network can provide simplified *backbone-based routing* and *spine-based routing*. Just as the nodes in a CDS form a connected backbone subgraph that can be used for routing, a set of intrinsic message ferries with sufficient connectivity over time can be used as a mobile routing backbone.

In this work we define the connected message ferry dominating set (CMFDS) problem and develop a heuristic to find a minimum-size CMFDS, given a model for the connectivity between nodes over time. Specifically we provide a detailed formalization of message ferrying concepts. We give a formal definition of a message ferry, then we present the concept of a message ferry dominating set (MFDS) — a *space-time* dominating set constituting nodes that behave as intrinsic message ferries. We further define a connected message ferry dominating set (CMFDS), that can be used to provide message ferrying connectivity between all sources and destinations in a mobile ad hoc network. Finding a CMFDS in an intermittently connected network is analogous to finding a CDS in a stationary network, a well known NP-complete problem. We present an algorithm that uses a heuristic approach to determine a CMFDS for the given network

and the application of our algorithm to a stationary network, followed by showing examples of how network and ferry parameters may impact the message ferrying capabilities of the nodes in the network.

12. Anonymous Routing in DTNs

While significant research efforts have focused on improving networking (e.g., latency, packet loss) and security (e.g., confidentiality, integrity) issues for DTNs, anonymous communication has received relatively little attention. Mechanisms providing anonymity are crucial given that knowledge of two nodes having exchanged messages may allow an adversary to recover sensitive information, including the structure of an organization or changes in that organization's behavior. For dissidents attempting to communicate in the presence of an oppressive government or a deployed army that worries about exposing the location of its officers, such traffic analysis attacks may be as damaging as a loss of confidentiality. While a range of mature anonymous communication methods exist for Internet-based communications, none of these techniques are immediately applicable to DTNs given their often unpredictable patterns of connectivity.

In this work, we present ARDEN, an anonymous routing protocol adapted to the unique operating conditions of DTNs. ARDEN builds upon traditional onion routing techniques by wrapping messages in multiple layers of encryption in order to hide their sources and destinations from intermediary nodes. However, whereas traditional onion routing requires that messages be encrypted using the keys of specific proxy nodes, ARDEN uses Attribute-Based Encryption (ABE) and multicast communications to allow multiple nodes to forward messages toward their intended destinations.

ARDEN uses the combination of these mechanisms to take advantage of DTNs in two critical ways. First, ARDEN is able to perform relatively expensive but semantically expressive cryptographic operations without reducing throughput, because the time between communication opportunities is often long enough to allow for complex computational tasks. Second, ARDEN can better respond to unexpected changes in topology by allowing specific small groups of nodes to act as proxies. Accordingly, ARDEN is able to provide DTNs with the same guarantees offered by onion routing on the Internet while drastically reducing timing correlation attacks.

In so doing, we make the following contributions:

- **Design and Implement a robust anonymous communication protocol for DTNs:** We develop ARDEN, an anonymous routing protocol for DTNs. ARDEN builds on a traditional onion routing architecture but incorporates Attribute-Based Encryption (ABE) to simplify management and drastically reduce latency while providing strong anonymity guarantees.
- **Measure and Compare Performance:** We perform a number of extensive simulations to accurately characterize ARDEN using the Huggle and RollerNet datasets. We also compare ARDEN against traditional onion routing approaches and against routing mechanisms without anonymity guarantees to understand its impact on latency and loss. Our experiments demonstrate that ARDEN adds very low communications overhead, decreasing delivery rates as little as 1.1% in some cases.

- Discuss robustness to more advanced attacks: We also investigate the robustness of our scheme to a number of passive and active attacks. We show that breaking ARDEN requires extraordinary effort on the part of the adversary.

13. iDTT: Delay Tolerant Data Transfer for P2P File Sharing Systems

Peer-to-Peer(P2P) file sharing systems have been the most popular Internet application for years. According to a recent Internet study, they account for 43% - 70% of the world- wide Internet. Unfortunately, due to the wide adoption of flat- rate broadband pricing, the large amount of P2P traffic merely causes a significant burden on ISPs instead of leading to extra revenue. As a direct reaction, many ISPs have attempted to discriminate against P2P traffic , which in turn causes heated controversy on network neutrality. Alleviating the conflict between P2P systems and ISPs is, therefore, important.

Recently proposed solutions primarily focus on localizing P2P traffic to reduce inter-domain traffic. The P4P project enables ISPs to help peers find preferred neighbors. The insight from these methods is that downloading from local sources effectively reduces both the network traffic and downloading time. However, the effectiveness of such techniques in a real Internet environment is still controversial because of the sparse distribution of peers .

In this work, we propose a scheme for resolving the P2P/ISP conflict that relies on the delay-tolerance of P2P applications. Data transmission among peers is shifted to times when ISPs have idle transmission capacity. Diurnal variation is a universal phenomenon of Internet traffic. The traffic at peak times can be several times larger than that at off-peak times. If a portion of Internet traffic can be shifted from peak time to off-peak time, the ISP burden can be proportionally relieved. In fact, some ISPs encourage such shift by offering a free download window at night. This time-shifting of traffic can also reduce ISP operating costs as discussed in more detail in the next section. This time-shift is feasible for video file sharing which represents a large portion of P2P traffic. Video is the most popular type of content shared in a P2P system, accounting for 55%-78% of P2P traffic in various regions. Additionally, the size of high-resolution video files is usually several gigabytes. Video downloading is delay tolerant for two reasons: the downloading time is usually long and varies from time to time, making additional delay less noticeable, and users will probably watch the downloaded video when they have enough time, leading to sometimes significant time gaps between downloading and watching a video.

To use the underutilized network capacity at off-peak time,we propose a novel architecture for Internet delay tolerant data transfer (iDTT). It is designed as an overlay network providing delay tolerant data transfer service to applications. It borrows the basic idea of delay tolerant networks in which nodes use the intermittently available connectivity to transfer data in a store-and-forward manner. Data is transferred in iDTT when the background Internet traffic is low, and is temporarily stored at iDTT nodes (called *iServers*) when the traffic is high. With the low price of storage today, iServer storage can be very large, supporting the time-shifting of a large amount of traffic. Even though iServers are always able to transfer data over the iDTT overlay, we develop an analogy with intermittently connected DTNs by allowing data transfer only when the overlay links between iServers have low utilization and disallowing it otherwise. Thus one can say that iServers are “in contact” when data transfer is allowed, and “out of contact” otherwise. This analogy allows us to develop routing protocols for the iDTT overlay that are

inspired by DTN routing protocols. Note that while we focus here on the use of iDTT for P2P systems, the architecture we propose is also suitable for other data intensive applications that can tolerate delay.

iDTT, despite its extra delays, can also be beneficial to end users. First, ISPs have no motivation to discriminate against iDTT traffic. On the contrary, they will probably encourage users to use this new service. Second, the delay tolerant communication supported by iDTT represents a new paradigm that may inspire the development of new applications exploiting its unique properties. To assess iDTT's practicality and utility, we implement a user-space prototype, running on Emulab. The GÉANT and Abilene networks with their corresponding traffic matrices are used to emulate a real network environment. Two representative P2P systems, eMule and BitTorrent, are adapted to use the data transfer service of iDTT. For these applications, we compare iDTT with traditional end-to-end data transfer. The results demonstrate that iDTT reduces the 95-percentile traffic while increasing the downloading time only slightly.