

## Scholars' Mine

**Masters Theses** 

Student Theses and Dissertations

Spring 2010

## Data aggregation in wireless sensor networks

Priya Kasirajan

Follow this and additional works at: https://scholarsmine.mst.edu/masters\_theses



Part of the Electrical and Computer Engineering Commons

Department:

#### **Recommended Citation**

Kasirajan, Priya, "Data aggregation in wireless sensor networks" (2010). Masters Theses. 5002. https://scholarsmine.mst.edu/masters\_theses/5002

This thesis is brought to you by Scholars' Mine, a service of the Missouri S&T Library and Learning Resources. This work is protected by U. S. Copyright Law. Unauthorized use including reproduction for redistribution requires the permission of the copyright holder. For more information, please contact scholarsmine@mst.edu.

# DATA AGGREGATION IN WIRELESS SENSOR NETWORKS

by

#### PRIYA KASIRAJAN

#### A THESIS

Presented to the Faculty of the Graduate School of the

MISSOURI UNIVERSITY OF SCIENCE AND TECHNOLOGY

In Partial Fulfillment of the Requirements for the Degree

#### MASTER OF SCIENCE IN ELECTRICAL ENGINEERING

2010

Approved by

Dr. J. Sarangapani, Advisor Dr. M. Zawodniok Dr. S. Madria

#### **PUBLICATION THESIS OPTION**

This thesis consists of the following two articles that have been submitted for publication:

Paper 1, pages 10 – 52, Priya Kasirajan, Carl Larsen and S. Jagannathan, "A New Data Aggregation Scheme via Adaptive Compression for Wireless Sensor Networks," has been submitted to <u>ACM Transactions on Sensor Networks</u>.

Paper 2, pages 53 – 93, Priya Kasirajan, Maciej Zawodniok and S. Jagannathan, "Hardware Verification of Data aggregation and Multi-interface Multi-channel Routing Protocol," intended for submission to <u>International Journal of Distributed Sensor Networks</u>.

#### **ABSTRACT**

Energy efficiency is an important metric in resource constrained wireless sensor networks (WSN). Multiple approaches such as duty cycling, energy optimal scheduling, energy aware routing and data aggregation can be availed to reduce energy consumption throughout the network. This thesis addresses the data aggregation during routing since the energy expended in transmitting a single data bit is several orders of magnitude higher than it is required for a single 32 bit computation. Therefore, in the first paper, a novel nonlinear adaptive pulse coded modulation-based compression (NADPCMC) scheme is proposed for data aggregation. A rigorous analytical development of the proposed scheme is presented by using Lyapunov theory. Satisfactory performance of the proposed scheme is demonstrated when compared to the available compression schemes in NS-2 environment through several data sets. Data aggregation is achieved by iteratively applying the proposed compression scheme at the cluster heads.

The second paper on the other hand deals with the hardware verification of the proposed data aggregation scheme in the presence of a Multi-interface Multi-Channel Routing Protocol (MMCR). Since sensor nodes are equipped with radios that can operate on multiple non-interfering channels, bandwidth availability on each channel is used to determine the appropriate channel for data transmission, thus increasing the throughput. MMCR uses a metric defined by throughput, end-to-end delay and energy utilization to select Multi-Point Relay (MPR) nodes to forward data packets in each channel while minimizing packet losses due to interference. Further, the proposed compression and aggregation are performed to further improve the energy savings and network lifetime.

#### **ACKNOWLEDGMENTS**

I would like to add a few heartfelt words for the people who were part of my thesis in numerous ways, people who gave unending support right from the beginning. I would like to express my sincere gratitude to my advisor, Dr. Jagannathan Sarangapani, for his continuous guidance and support. I would also like to thank Dr. Maciej Zawodniok for his invaluable contributions and suggestions that helped me complete this work. I would also like to thank Dr. Sanjay Madria for his kind support. Financial assistance from the Air Force Research Laboratory and Army's Leonard Wood Institute in the form of research assistantship is thankfully acknowledged. Lastly, I would like to thank my parents and friends for their constant encouragement and support throughout my education.

## TABLE OF CONTENTS

	Page
PUBLICATION THESIS OPTION	iii
ABSTRACT	iv
ACKNOWLEDGMENTS	v
LIST OF ILLUSTRATIONS	viii
LIST OF TABLES	x
SECTION	
1. INTRODUCTION	1
1.1 MOTIVATION	2
1.2 ORGANIZATION OF THE THESIS	5
1.3 CONTRIBUTIONS OF THE THESIS	6
REFERENCES	8
PAPER	
A NEW DATA AGGREGATION SCHEME VIA ADAPTIVE     COMPRESSION FOR WIRELESS SENSOR NETWORKS	10
ABSTRACT	10
I. INTRODUCTION	11
II. BACKGROUND	14
A. Quantization	14
B. ADPCM	15
III. PROPOSED METHODOLOGY	16
A. Adaptive estimation	16
B. Analytical results	18
C. Algorithm	26
IV. RESULTS AND DISCUSSION	27
A. Synthetic data	31
B. River discharge data	35
C. Audio data	38
D. Geophysical data	40

E. Performance as an aggregation scheme	43
F. Scalability	46
V. CONCLUSIONS	49
REFERENCES	50
2. HARDWARE VERIFICATION OF DATA AGGREGATION AND M INTERFACE MULTI-CHANNEL ROUTING PROTOCOL	
ABSTRACT	53
I. INTRODUCTION	54
II. BACKGROUND	57
A. Neighbor discovery	58
B. MPR selection	62
C. Topology discovery	64
D. Route selection and data transmission using the selected routes	65
III. IMPLEMENTATION OF THE ROUTING PROTOCOL	66
A. Hardware description and limitations	67
B. Implementation details	68
C. Node functions	70
D. Protocol verification	72
E. Real-time voice with MMCR	76
IV. IMPLEMENTATION OF DATA AGGREGATION	79
A. NADPCMC	79
B. Zeolite sensors for explosive detection	81
C. Transmission of sensor data	83
D. Energy consumption	89
V. CONCLUSIONS	91
REFERENCES	91
SECTION	
2. CONCLUSIONS AND FUTURE WORK	94
APPENDIX - SOURCE CODE ON CD-ROM	96
VITA	07

## LIST OF ILLUSTRATIONS

Fig	ure	Page
INT	TRODUCTION	
1.1	. Architecture of a sensor node	1
1.2	. CPU compute cycles versus transmission energy of one byte over three radios	3
1.3	. Thesis outline	6
PA	PER 1	
1.	Proposed architecture	16
2.	Network topology	29
3.	Hardware architecture	31
4.	Estimator output	32
5.	Reconstruction with 8 bit encoded error	33
6.	Total reconstruction error with different error encodings	33
7.	Output of estimator	36
8.	Reconstruction with 8 bit encoded error	36
9.	Total reconstruction error with different error encodings	37
10.	Total reconstruction error with 8 bit encoded error	39
11.	Total reconstruction error with 6 bit encoded error	39
12.	Total reconstruction error with 8 bit encoded error	41
13.	Total reconstruction error with 6 bit encoded error	42
14.	Hardware architecture	43
15.	Dependency on number of flows	47
16.	Average compression ratio at first cluster-head level	48
17.	Average compression ratio at second cluster-head level	49
PA	PER 2	
1.	Neighbor discovery	59
2.	Handling BEAM packets	59
3.	Sending ACK packets	60
4.	Handling of ACK packets	61
5	Pseudo-code for MPR selection	63

6.	MPRs selected by the algorithm	. 63
7.	Sending TC packets	. 64
8.	Handling of TC packets	. 65
9.	Handling of SWITCH packets	. 66
10.	G4 mote	. 68
11.	Packet structure	. 69
12.	Application specific header	. 70
13.	Activities of sender	. 71
14.	Activities of an intermediate node	. 71
15.	Activities of a destination node	. 72
16.	Demonstration of MMCR	. 73
17.	Performance metrics.	. 74
18.	Demonstration of channel switching	. 75
19.	Channel switching	. 76
20.	Real-time voice over MMCR	. 77
21.	Performance metrics for real-time voice	. 78
22.	NADPCMC flowchart	. 81
23.	M2 mote	. 82
24.	Prototyped sensor circuit response	. 83
25.	8 bit NADPCMC in the presence of packet losses	. 84
26.	Modified 8 bit NADPCMC in the presence of packet losses	. 85
27.	Demonstration of data aggregation	. 86
28.	Performance metrics with 4 bit data aggregation	. 87
29.	Reconstructed explosive sensor data	. 88
30.	Reconstructed river discharge data	. 89

## LIST OF TABLES

Table		Page	
PA	APER 1		
1.	Performance metrics for synthetic data	34	
2.	Performance metrics for river-discharge data	38	
3.	Performance metrics for audio data	40	
4.	Performance metrics for geophysical data	42	
PA	APER 2		
1.	Specifications of G4 mote	68	
2.	Average performance metrics for raw data	74	
3.	Average performance metrics for real-time voice	78	
4.	Effect of data aggregation	88	
5.	Energy expenditure	90	

#### **SECTION**

#### 1. INTRODUCTION

Wireless Sensor Networks (WSN) are one of the first practical real-world examples of the pervasive computing paradigm - the concept of small, inexpensive, robust, networked processing devices eventually permeating the environment. In 2003, MIT's Technology Review magazine [1] described sensor networks as "One of the ten technologies that will change the world." Though sensors have been available for decades, the application of the technology was hampered until recently owing to the high cost. The advances in semiconductor technology finally brought smaller and cheaper sensors to life. The same semiconductor manufacturing techniques miniaturized radios and processors. The system-on-a-chip (SoC) technology integrated microsensors, onboard processing and wireless interfaces which is now referred to as a sensor node or a mote. A sensor node with several features is shown in Figure 1.1.

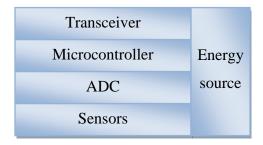


Figure 1.1. Architecture of a sensor node

Once networked, deeply embeddable sensor nodes can reveal phenomena that were previously unobservable. Existing and potential applications of WSNs include, among others, radiation detection, habitat sensing, seismic monitoring, video surveillance, traffic surveillance, environment monitoring, weather sensing, homeland security, forest fire detection and chemical attack detection.

#### 1.1 MOTIVATION

Energy constraints dominate algorithm and system design trade-offs for small embedded sensor devices. The lifetime of a WSN depends on the energy that can be stored or harvested by individual sensor nodes. WSNs are meant to be deployed in large numbers in various environments, including remote and hostile regions, with ad-hoc network communications as key way of connecting nodes. In most situations, replacement of dead batteries is expensive. Hence lifetime maximization through energy efficiency becomes an important issue. The following are a few ways to address the power consumption issue.

- **1. Duty cycling** The energy consumption for idle listening, which is needed to keep the receiving circuitry awake for possible packet reception, is a major source of current drain. Duty cycling schemes [2] [3] define coordinated sleep/wakeup schedules consisting of short active durations and long inactive ones.
- **2. Adaptive sampling** This method improves the network efficiency and the data accuracy by dynamically changing the sampling rate of a node in response to its data characteristics [4] or available resources [5]. Consider the example of a fire detection

sensor. If the temperature is nearly constant for a long time, the sampling rate of the sensor node can be decreased to reduce the data transmission without affecting the data quality. On the other hand, if the temperature increases rapidly above a threshold, the base-station has to be informed impromptu and sampling rate of the sensor node should be increased to improve the accuracy [6].

**3. In-network processing** –Data transmission is probably the most energy-intensive operation performed by a sensor node. Figure 1.2 displays the number of TIMSP430F1611 machine cycles equivalent in energy to the transmission of a single byte over the CC2420, CC1000 and MaxStream XTend radios. This figure indicates that any additional processing to reduce at least a single data bit might still be advantageous in terms of energy efficiency.

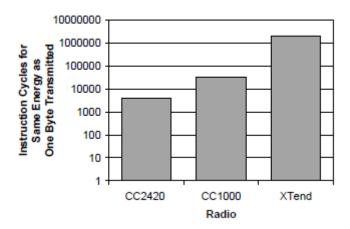


Figure 1.2. CPU compute cycles versus transmission energy of one byte over three radios (Reprint from [9])

In-network processing involves reducing the amount of data to be transmitted by means of compression [7] and/or aggregation [8] techniques. As an example, consider a Cluster-head receiving two packets from two different sources containing the locally measured temperatures. Instead of forwarding the two packets, the sensor may compute a statistical aggregate such as AVERAGE or MAX or MIN of the two readings and send it in a single packet [6]. The tradeoff is that though this approach reduces the amount of data to be sent over the network but it may also reduce the accuracy with which the gathered information can be recovered at the sink. After aggregation, it is usually not possible to perfectly reconstruct all of the original data.

4. Energy aware routing – While data aggregation is intended to reduce the number of transmissions, routing is intended to ensure reliable packet delivery and minimize the number of retransmissions. With the exponential growth in the energy-cost of radio transmission with respect to the distance transmitted, it is very unlikely that every node will reach the base station. Thus, multi-hop routing is mandatory. The basic idea for multi-hop routing then is to route the packet through the minimum energy paths so as to minimize the overall energy consumption for delivering the packet from the source to the destination. Routing protocols use multiple paths rather than a single path in order to both enable load balancing and increase fault tolerance capabilities. Moreover, some sensor nodes have radios with multiple interfaces and can handle many non-overlapping channels. This instigates the use of a multi-channel routing protocol that would balance the load evenly on multiple channels and ensure reliable packet delivery with minimal packet losses due to interference.

Thus, a variety of ways are available for reducing energy consumption in energy and resource constrained WSNs. Moreover, as mentioned before, the key lies in the minimization of the number of transmissions and retransmissions of sensor data. This thesis explores the last two options for energy efficiency improvement in WSNs—Data aggregation and multi-channel routing. Data aggregation reduces the number of transmissions while efficient routing reduces the number of retransmissions. Consequently, these two schemes can be applied together to achieve a significant amount of energy efficiency improvement.

#### 1.2 ORGANIZATION OF THE THESIS

This thesis is presented in two papers. Their relationship is shown in Figure 1.3. Both the papers address energy efficiency improvement in WSNs through data aggregation. In paper 1, a new compression scheme based on adaptive estimation and quantization is proposed. Convergence is proved and analytical bounds on the distortion are derived using Lyapunov theory. The proposed scheme is then tested on multiple datasets and topologies with MATLAB and the Network Simulator NS-2. Then data aggregation through iterative application of compression is analyzed. This is followed by scalability tests to verify protocol performance in large WSNs.

Paper 2 deals with the hardware implementation of a proactive routing protocol for WSNs - Multi-interface Multi-Channel Routing (MMCR) protocol. This protocol is evaluated for different data flow cases. This is followed by hardware verification of the aggregation scheme developed in paper 1 in conjunction with the routing protocol and

experimental results show that the proposed aggregation scheme indeed results in energy efficiency beyond the energy efficient routing protocol can offer.

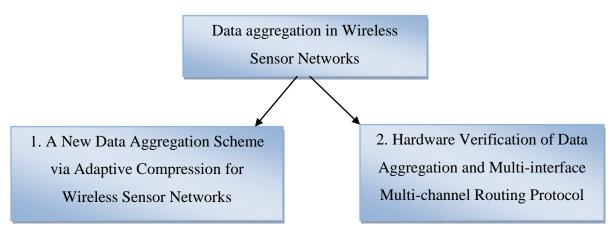


Figure 1.3. Thesis outline

#### 1.3 CONTRIBUTIONS OF THE THESIS

Data aggregation reduces energy consumption by combining data from different sensors and eliminates unnecessary packet transmission by filtering out redundant sensor data. Most of the existing compression/aggregation methods [12] [13] operate well on specific types of data while their performance on the others are unsatisfactory. This calls for a compression scheme that would perform fairly well on various types of data. In the first paper, a new compression scheme based on adaptive nonlinear estimation and quantization is developed. Lyapunov theory is used to derive theoretical bounds on performance in the presence of approximation errors. The scheme is then verified on multiple data sets and on networks of varying sizes.

The second paper deals with the hardware verification of the MMCR routing protocol and data aggregation using NADPCMC on the Missouri S&T mote network. Many existing routing methods [14] [15] do not exploit the possibility of using multiple radio channels for routing. However, the MMCR protocol utilizes multiple channels for transmission and improves quality of service by using a routing metric that involves throughput, delay and energy utilization. The addition of in-network aggregation using NADPCMC further improves the network utilization and energy efficiency.

#### **REFERENCES**

- [1] MIT Technology Review available online <a href="http://www.techreview.com">http://www.techreview.com</a> accessed on Nov 2009.
- [2] W. Ye, J. Heidemann, and D. Estrin, "An energy-efficient MAC protocol for wireless sensor networks," *Proc. of the IEEE INFOCOM*, Vol. 3, pp. 1567 1576, Jun 2002.
- [3] T.R. Park, K. Park, M.J. Lee, "Design and analysis of asynchronous wakeup for wireless sensor networks," *IEEE Transactions on Wireless Communications*, Vol. 8, pp. 5530 5541, Nov 2009.
- [4] R. Willett, A. Martin, R. Nowak, "Backcasting Adaptive sampling for sensor networks," *IPSN*, pp. 124 133, Apr 2004.
- [5] A. Jain, E. Chang, "Adaptive sampling for sensor networks," *Proc. of the 1st international workshop on Data management for sensor networks*, pp. 10 14, 2004.
- [6] H. Wu, Q. Luo, "Supporting adaptive sampling in wireless sensor networks," *IEEE WCNC*, pp. 3442 3447, Mar 2007.
- [7] F. Marcelloni, M. Vecchio, "A simple algorithm for data compression in wireless sensor networks," *IEEE Communications Letters*, Vol. 12, pp. 411 413, Jun 2008.
- [8] E. Fasolo, M. Rossi, J. Widmerand M. Zorz, "In-network aggregation techniques for wireless sensor networks: A survey," *IEEE Transactions on Wireless Communications*, Vol. 14, pp. 70-87, Apr 2007.
- [9] C.M. Sadler and M. Martonosi, "Data compression algorithms for energy-constrained devices in delay tolerant networks," *Proc. of the 4th Int'l conference on Embedded networked sensor systems*, pp. 265-278, 2006.
- [10] S. Kee-Young, S. Junkeun, K. JinWon, Y. Misun, S.M. Pyeong, "REAR: Reliable energy aware routing protocol for wireless sensor networks", 9<sup>th</sup> Int'l Conference on Advanced Communication Technology, Vol. 1, pp. 525 530, Feb 2007.
- [11] T. Stathopoulos, M. Lukac, D. McIntire, J. Heidemann, D. Estrin, W.J. Kaiser, "End-to-end routing for dual-radio sensor networks", *IEEE INFOCOMM*, pp. 2252 2260, May 2007.
- [12] C. Alippi, R. Camplani, and C. Galperti, "Lossless compression techniques in wireless sensor networks: Monitoring Microacoustic Emissions," *Int'l Workshop on Robotic and Sensors Environments*, pp. 1-5, Oct 2007.

- [13] P. Cummiskey, N.S. Jayant, and J.L. Flanagan, "Adaptive quantization in differential PCM coding of speech," *Bell Syst. Tech. J.*, vol. 52, pp. 1105-1118, Sept 1973.
- [14] C.E. Perkins and P. Bhagwat, "Highly dynamic destination-sequenced distance vector routing (dsdv) for mobile computers," *Proc. of SIGCOMM*, pp. 234 244, Sept 1994.
- [15] C.E. Perkins and E. M. Royer, "Ad-hoc on-demand distance vector routing," Proc. of WMCSA, pp. 90 100, Feb 1999.

#### **PAPER**

## 1. A NEW DATA AGGREGATION SCHEME VIA ADAPTIVE COMPRESSION FOR WIRELESS SENSOR NETWORKS

Priya Kasirajan, Carl Larsen and S. Jagannathan

ABSTRACT — Data aggregation should be performed to extend network lifetime for wireless sensor nodes with limited processing and power capabilities since energy expended in transmitting a single data bit would be at least several orders of magnitude higher when compared to that needed for a 32 bit computation. Therefore, in this paper, a novel nonlinear adaptive pulse coded modulation-based compression (NADPCMC) scheme is proposed for data aggregation. Satisfactory performance of the proposed compression scheme in terms of distortion, compression ratio, energy efficiency and in the presence of estimation and quantization errors is demonstrated using Lyapunov approach. Then the performance of the proposed scheme is contrasted with the available compression schemes in NS-2 environment through several data sets. Simulation and hardware experimental results demonstrate that almost 50% energy savings with low distortion levels less than 5% and low overhead are observed. Iteratively applying the proposed compression scheme at the cluster head nodes over the network yields an improvement of 20% in energy savings per aggregation with overall distortion below 8%.

Keywords: Compression, Data Aggregation, Energy Efficiency, Wireless Sensor Networks

#### I. INTRODUCTION

Recent advancements in embedded processing and wireless networking have led to the development of wireless sensor networks (WSN). A WSN is a multi-hop network of nodes each with a short-range radio, limited sensing and on-board processing capability. Sensor nodes are powered by small batteries, which determine their lifetime. This necessitates network protocols with energy efficiency as a critical design goal. Some popular tailored applications for fulfilling this goal include adaptive sampling [1], energy-aware routing [2], energy-efficient data processing [3], and energy-optimal topology construction [4].

In this paper, we focus on designing techniques to conserve energy by reducing amount of data transmitted while still delivering all the information which is referred to as data aggregation. This process usually involves data at select nodes, called Clusterheads, being combined by computing statistical aggregates such as COUNT, SUM, AVERAGE, MAX or MIN and then sending this data to the observer at the base-station node [3] [5]. In [6], a comprehensive survey of data aggregation schemes applicable for different topologies such as flat, hierarchical, cluster-based and grid networks is presented. In the literature [6], data aggregation methods focus only on reducing the overall amount of data by combining data from geospatially located sensors. In many applications such as monitoring of forest fire, humidity in a building, water level etc., sensors repeatedly report data values, and therefore the amount of data transmitted onto the network can be further reduced if we combine multiple data values from a single sensor over time. This task can be achieved through data compression, whereby a large number of bits of data, in this case multiple sensor data values, are "compressed" and

represented by a smaller number of bits in such a way that we can recreate the original data at the base station from those bits. Since more data is represented using fewer bits, energy required to transmit this compressed data is significantly less by every node that forwards the data. Though these methods seem computationally intensive, the energy required to transmit an extra bit is at least several orders of magnitude higher than the energy required for a single 32 bit computation [7]. Thus, compression algorithms with a reasonable level of complexity are certainly worth exploring for data aggregation.

While there are many compression algorithms [8], not many [9] [10] are currently used in WSNs. Though audio and video data may tolerate some degradation in quality, sensor data must be relayed faithfully without loss of vital information. Therefore, the performance of popular entropy encoding schemes such as Huffman coding [11], Adaptive Huffman coding and Delta coding are studied and compared in [12] for a micro-acoustic emissions sensor network. In [7], the authors propose and evaluate a variant of the famous LZW algorithm called S-LZW, specifically tailored for sensor nodes. All these algorithms are lossless and provide light compression since they use a heavy codebook.

When a certain amount of data loss can be tolerated, better compression can be achieved using lossy compression algorithms. In [13], a combination of regression and model based compression is suggested. A base signal is constructed from a set of values that capture the most prominent features of the data. Then, the collected data is partitioned into intervals that can be efficiently approximated as linear projections of some part of the base signal. This method promises high accuracy and a satisfactory reduction in bandwidth consumption for linearly varying data. There are a number of

methods [14] [15] based on regression to compress data with a certain percentage of distortion. A new model based compression technique called adaptive learning vector quantization (ALVQ) is proposed in [16] to compress historical data. A codebook is created from training data at the sensor. This codebook is used to encode the samples in real-time. When the buffer is full, the codebook is updated and a 2-level piece-wise linear regression technique is applied to compress the codebook update.

The entropy encoding schemes [11] [12] work best on correlated data. By contrast, the regression techniques [13] [14] [15] [16] perform well when the data is linearly varying. Our objective is to develop a compression scheme that could be applied on any form of data provided it is deterministic. Our motivation comes from the adaptive differential pulse code modulation (ADPCM) [17] scheme wherein a linear estimate of the sample is generated at every instant, compared with the original sample and only the difference is quantized resulting in good compression. By contrast, we propose to represent the data as a nonlinear relationship and use techniques from adaptive estimation theory to obtain an accurate estimate. A novel nonlinear discrete-time estimator is proposed and its performance is demonstrated using Lyapunov theory. It will be shown that the nonlinear adaptive pulse coded modulation-based compression (NADPCMC) indeed results in better compression ratio, reduced distortion, and higher energy efficiency analytically when compared to its linear counterpart and other lossless compression schemes. Subsequently, the performance of the NADPCMC is demonstrated in NS-2 environment using various data sets and on hardware using multiple levels of aggregation.

The rest of this paper is organized as follows. Section 2 outlines the necessary background. Section 3 deals with the proposed methodology, discusses theoretical bounds on the stability of the proposed scheme and presents the detailed algorithm. The algorithm is tested with MATLAB and NS2, and results from the simulations and hardware implementation are detailed in Section 4. Section 5 contains the concluding remarks.

#### II. BACKGROUND

In this section, some background on quantization and linear ADPCM is briefly revisited since quantization is still applied in the proposed approach as well.

#### A. Quantization

Quantization is a process by which a continuous signal is approximated and mapped into a finite set of values. This mapping process invariably results in loss of some information in the presence of quantization noise.

A uniform quantizer approximates the signal into equally spaced quantization levels. In other words, the quantizer step size is typically held as a constant. For a b bit quantization of a signal that has a dynamic range of  $(E_{min}, E_{max})$ , the required step size is given by

$$step\_size = (E_{max} - E_{min})/2^b \tag{1}$$

The quantization noise depends on the reconstruction levels which in turn depend on the step size. The maximum error in quantization is *step\_size/2* [8]. Thus, the quantization noise depends on the dynamic range of the data and the resolution of the

encoder. In other words, for a given bit size, a signal with smaller dynamic range can be approximated with less errors than one with a larger dynamic range. Thus, some preprocessing has to be done on the data to reduce its dynamic range, so that the bit rate can be reduced to achieve compression.

#### B. ADPCM

The ADPCM scheme is used widely in speech coding. It uses the correlation between adjacent samples to reduce bit rate and achieve compression. Instead of quantizing the speech signal directly, only the difference between the actual sample and the predicted sample is quantized. If the prediction is accurate then the difference between the real and predicted speech samples will have a lower variance than the real speech samples, and will be accurately quantized with fewer bits than what would be needed to quantize the original speech samples. At the decoder the quantized difference signal is added to the predicted signal to give the reconstructed speech signal.

The amount of compression achieved depends on the performance of the predictor. Real world sensor data does not always show good correlation as speech signals. A linear predictor will not always be able to handle fast changing data. We propose the use of an adaptive nonlinear estimator to get a better prediction of the sample and reduce the bit rate. The idea of estimation is shown subsequently.

#### III. PROPOSED METHODOLOGY

Figure 1 depicts the proposed compression-based data aggregation approach. Two stages are involved at the source node – estimation and quantization. In the first stage, nonlinear adaptive estimation is performed to obtain a close estimate of the current sample based on a few previous samples. In the second stage, the difference between the actual value and the estimated value is quantized. This quantized value is sent to the next node or to the destination. At the destination end, a similar estimator is used. A few initial samples are fed directly to the estimator to help it get started. Subsequently, the estimation errors from the first encoder are added to the estimate obtained from the second to reconstruct the signal.

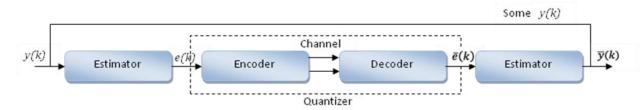


Figure 1. Proposed architecture

#### A. Adaptive estimation

Adaptive estimation of the data sequence is performed by representing the data as a nonlinear autoregressive moving average sequence as

$$y(k+1) = \theta^{T}(k)\emptyset(k) + \varepsilon(y)$$
(2)

where  $\emptyset(k) \in \mathbb{R}^{n+1}$  is the basis function,  $\theta(k)$  is the unknown parameter vector, and  $\varepsilon(y) \in \mathbb{R}$  is the reconstruction error such that it is bounded above by  $\|\varepsilon(y)\| \leq \varepsilon_N$ . The estimated signal is given by

$$\hat{y}(k+1) = \hat{\theta}(k)\emptyset(k) - k_{,,e}(k) \tag{3}$$

where  $\hat{\theta}(k)$  is the estimated parameter estimate vector and e(k) is the estimation error.

The estimation error is then given by

$$e(k+1) = y(k+1) - \hat{y}(k+1) \tag{4}$$

Substituting for y and  $\hat{y}$  in (3) renders

$$e(k+1) = k_{v}e(k) + \tilde{\theta}^{T}(k)\varphi(k) + \varepsilon$$
(5)

where the parameter estimation error is defined by

$$\tilde{\theta}^T(k) = \theta(k) - \hat{\theta}(k) \tag{6}$$

Now consider the parameter update as

$$\widehat{\theta}(k+1) = \widehat{\theta}(k) + \alpha \emptyset(k) e^{T}(k+1)$$
(7)

Here  $\alpha$  is called the adaptation gain.

It will be shown by using the estimation error (5) and parameter update (7) that the estimation error is bounded in the general case when there are reconstruction and quantization errors and convergence of the estimation error to zero in the ideal case when there are no reconstruction and quantization errors. Since the estimation error is related to distortion, subsequently, it will be shown that the distortion is also bounded.

#### B. Analytical results

The following theorems examine the stability of the estimator and the performance of the method. In the ideal case, when the estimation error  $\varepsilon = 0$ , equation (5) reduces to

$$e(k+1) = k_{,,}e(k) + \tilde{\theta}^{T}(k)\varphi(k)$$
(8)

where  $\bar{e}_i(k) = \tilde{\theta}^T(k)\varphi(k)$ . Next the following theorem can be stated in the absence of approximation errors.

**Theorem 1 (Estimator-Ideal Performance):** Let the proposed nonlinear estimator given in (3) be utilized with the parameter vector be tuned by (7). In the ideal case with no reconstruction errors and noise present, the estimation error e(k) approaches to zero asymptotically while the parameter estimation error vector is bounded.

**Proof:** Select the Lyapunov candidate as

$$J(k) = e^{T}(k)e(k) + \frac{1}{\alpha}\tilde{\theta}^{T}(k)\tilde{\theta}(k)$$
(9)

The first difference is given by

$$\Delta J = e^{T}(k+1)e(k+1) - e^{T}(k)e(k)$$

$$+\frac{1}{\alpha} \left[ \tilde{\theta}^{T}(k+1)\tilde{\theta}(k+1) - \tilde{\theta}^{T}(k)\tilde{\theta}(k) \right]$$
 (10)

Let  $\Delta J = \Delta J_1 + \Delta J_2$ , where

$$\Delta J_1 = e^T (k+1) e(k+1) - e^T (k) e(k)$$
(11)

$$\Delta J_2 = \frac{1}{\alpha} [\tilde{\theta}^T(k+1)\tilde{\theta}(k+1) - \tilde{\theta}^T(k)\tilde{\theta}(k)]$$
(12)

Substituting (6) and (7) into (12) results in

$$\Delta J_2 = -\bar{e}_i^T(k)\bar{e}(k+1) - \bar{e}_i(k)e^T(k+1)$$

$$+\alpha \emptyset^T(k)\emptyset(k)e(k+1)e^T(k+1)$$
(13)

Therefore (11) becomes

$$\Delta J = [1 + \alpha \emptyset^T(k)\emptyset(k)]e^T(k+1)e(k+1)$$

$$-\bar{e}_i^T(k)\bar{e}(k+1) - \bar{e}_i(k)e^T(k+1) - e^T(k)e(k)$$
(14)

Thus,

$$\Delta J \leq -e^{T}(k)[I - k_{v}^{T}k_{v}]e(k)$$

$$+\alpha \parallel \emptyset(k) \parallel^{2} (k_{v}e(k))^{T}k_{v}e(k)$$

$$+2\alpha \parallel \emptyset(k) \parallel^{2} (K_{v}e(k))^{T}\bar{e}_{i}(k)$$

$$-[1 - \alpha \parallel \emptyset(k) \parallel^{2}]\bar{e}_{i}(k)^{T}\bar{e}_{i}(k)$$
(15)

$$\Delta J \leq \; -(1 - \; \eta k^{\,2}_{\;vmax}) \parallel e(k) \parallel^{2} - \; \left[1 - \alpha \parallel \, \emptyset(k) \parallel^{2}\right]$$

$$\| \bar{e}_i(k) - \frac{\alpha \| \emptyset(k) \|^2}{1 - \alpha \| \emptyset(k) \|^2} k_v e(k) \|^2$$
(16)

where

$$\eta = \frac{1}{1 - \alpha \|\emptyset(k)\|^2} \tag{17}$$

and the maximum singular value of the gain matrix is given by

$$k_{vmax} < \frac{1}{\sqrt{\eta}} \tag{18}$$

Since J > 0 and  $\Delta J \le 0$ , this shows stability in the sense of Lyapunov. Since  $\Delta J$  is negative semidefinite [18] and according to Lyapunov theory, summing both sides of (16) and taking limits, it is easy to show that the estimation error approaches to zero asymptotically i.e.  $\| e(k) \| \to 0$  as  $k \to \infty$  and the parameter estimation errors are bounded. Thus, the estimation error tends asymptotically to zero in the absence of reconstruction errors.

In the general non-ideal case, when the reconstruction error is nonzero, the estimation error is as defined in (5) and the following theorem can be stated.

Theorem 2 (Estimator Performance—General Case): Let the hypothesis presented in Theorem 1 hold and if the functional reconstruction error is bounded with  $\|\varepsilon(y)\| \le \varepsilon_N$ , then estimation error  $\varepsilon(k)$  is bounded while the parameter errors are also bounded.

**Proof:** Select the Lyapunov candidate as

$$J(k) = e^{T}(k)e(k) + \frac{1}{\alpha}\tilde{\theta}^{T}(k)\tilde{\theta}(k)$$
(19)

Using (10), the first difference is given by

$$\Delta J = e^{T}(k+1)e(k+1) - e^{T}(k)e(k)$$

$$+\frac{1}{\alpha} \left[ \tilde{\theta}^T(k+1)\tilde{\theta}(k+1) - \tilde{\theta}^T(k)\tilde{\theta}(k) \right] \tag{20}$$

Let  $\Delta J = \Delta J_1 + \Delta J_2$  where

$$\Delta J_1 = e^T (k+1) e(k+1) - e^T (k) e(k) \tag{21}$$

$$\Delta J_2 = \frac{1}{\alpha} \left[ \tilde{\theta}^T (k+1) \tilde{\theta}(k+1) - \tilde{\theta}^T (k) \tilde{\theta}(k) \right]$$
 (22)

Substituting (6) and (7) into (22) results in

$$\Delta J_2 = -\bar{e}_i^T(k)\bar{e}(k+1) - \bar{e}_i(k)e^T(k+1)$$

$$+\alpha \emptyset^T(k)\emptyset(k)e(k+1)e^T(k+1)$$
(23)

Now (11) can be written as,

$$\Delta J = [1 + \alpha \emptyset^{T}(k)\emptyset(k)]e^{T}(k+1)e(k+1)$$

$$-\bar{e}_{i}^{T}(k)\bar{e}(k+1) - \bar{e}_{i}(k)e^{T}(k+1) - e^{T}(k)e(k)$$
(24)

Substituting for e(k+1) and simplifying,

$$\Delta J \leq -e^{T}(k)[I - k_{v}^{T}k_{v}]e(k) 
+\alpha \parallel \emptyset(k) \parallel^{2} (k_{v}e(k))^{T}k_{v}e(k) 
+ [1 + \alpha \parallel \emptyset(k) \parallel^{2}]\varepsilon^{T}(k)\varepsilon(k) 
+2[1 + \alpha \parallel \emptyset(k) \parallel^{2}](K_{v}e(k))^{T}\varepsilon(k) 
-[1 - \alpha \parallel \emptyset(k) \parallel^{2}] * 
\left(\bar{e}_{i}(k)^{T}\bar{e}_{i}(k) - \frac{2\alpha \parallel \emptyset(k) \parallel^{2}}{1 - \alpha \parallel \emptyset(k) \parallel^{2}}(\varepsilon(k) + K_{v}e(k))\bar{e}_{i}^{T}(k)\right)$$
(25)

From (17) and (18),  $\eta = \frac{1}{1-\alpha\|\phi(k)\|^2}$  and  $k_{vmax} < \frac{1}{\sqrt{\eta}}$ 

Then, 
$$\Delta J \leq -(1 - \eta k^2_{vmax})$$

$$\left( \parallel e(k) \parallel^2 - \frac{2\eta k_{vmax}}{1 - \eta k^2_{vmax}} \varepsilon_N \parallel e(k) \parallel - \frac{\eta}{1 - \eta k^2_{vmax}} \varepsilon^2_N \right)$$

$$- \left[ 1 - \alpha \parallel \emptyset(k) \parallel^2 \right]$$

$$\| \bar{e}_i(k) - \frac{\alpha \| \emptyset(k) \|^2}{1 - \alpha \| \emptyset(k) \|^2} (k_v e(k) + \varepsilon(k)) \|^2$$
(26)

Thus,  $\Delta J \leq 0$  as long as

$$\parallel e(k) \parallel > \frac{1}{1-\eta k^2_{vmax}} \varepsilon_N \left( \eta k_{vmax} + \sqrt{\eta} \right)$$
 (27)

This demonstrates that  $\Delta J \leq 0$  outside a compact set U. Thus, the estimation error is bounded. By applying the persistency of excitation condition [18], it is easy to show that the parameter estimates are bounded as long as the above equations (17), (18) and (27) are satisfied.

The above theorems demonstrated the performance of the estimator. Let us now analyze the overall approach. The proposed scheme (as shown in Fig. 1) involves 2 estimators – one at the transmitter and one at the receiver. The error in estimation from the first is quantized and fed to the second.

The entire NADPCMC scheme can be expressed mathematically as follows: The first estimator continuously produces an estimate  $\hat{y}_T$ . From (3), the estimated signal can be represented as

$$\widehat{y}_{\tau}(k+1) = \widehat{\theta}_{\tau}(k) \emptyset_{\tau}(k) - k_{v} e(k)$$
(28)

The error in estimation is obtained from (4) as

$$e(k+1) = y(k+1) - \hat{y}_T(k+1)$$
(29)

The parameter  $\hat{\theta}_R$  is continuously updated such that the error e which is given by  $\hat{y}_T - y$  is minimized. From (7), we have

$$\widehat{\theta}_T(k+1) = \widehat{\theta}_T(k) + \alpha \emptyset_T(k) e^T(k+1)$$
(30)

Then, e(k) is quantized. This stage adds a quantization error  $\varepsilon_Q$ 

$$\bar{e}(k) = Q(e(k)) = e(k) + \varepsilon_Q \tag{31}$$

The first few samples of y(k) are sent to the receiver side estimator to initialize  $\emptyset_R(k)$ . This is sufficient for it to start estimating  $\widehat{y}_R$ . As in (3), the estimated signal is given by,

$$\widehat{y}_R(k+1) = \widehat{\theta}_R(k) \emptyset_R(k) - k_v e_R(k)$$
(32)

Now, to obtain the original signal, we simply add the error offset to the estimate.

Thus, the recovered signal can be expressed as

$$\bar{y}(k+1) = \hat{y}_{R}(k+1) + \bar{e}(k+1)$$
 (33)

The error in estimation is obtained from (4) as

$$e_R(k+1) = \bar{y}(k+1) - \hat{y}_R(k+1) = \bar{e}(k+1)$$
(34)

The parameter  $\hat{\theta}_R$  is updated to account for the error  $\bar{e}$  that was incurred at the transmitter side estimator. As in (7), we have

$$\widehat{\theta}_R(k+1) = \widehat{\theta}_R(k) + \alpha \emptyset_R(k) e_R^T(k+1)$$
(35)

Loss of data can occur at both the estimation and quantization stages. The quantization error  $\varepsilon_Q$  is bounded by  $step\_size/2$  and thus, resolution of the quantizer has to be chosen based on the permissible level of distortion. Let us now proceed to analyze the maximum distortion introduced by our scheme.

In the proposed scheme, the amount of data lost is dependent on the total error in reconstructing the data at the receiver. The total error after reconstruction is  $|y(k) - \overline{y}(k)|$ . Now the following theorem can be stated.

**Theorem 3 (NADPCMC Distortion):** Consider the NADPCMC scheme presented in (3) through (7). If the estimator reconstruction and quantization errors are considered bounded, then the distortion at the destination is bounded. On the other hand in the absence of estimator reconstruction and quantization errors, the distortion is zero.

**Proof:** Let us consider the case where there are no reconstruction and quantization errors. The total reconstruction error after substituting from (33) is

$$|y(k) - \overline{y}(k)| \le |y(k) - \hat{y}_{R}(k) - \overline{e}(k)| \tag{36}$$

Substituting from (29) and (31)

$$|y(k) - \overline{y}(k)| \le |y(k) - \hat{y}_R(k) - y(k) + \hat{y}_T(k) - \varepsilon_Q|$$
(37)

Simplifying (37), we get

$$|y(k) - \overline{y}(k)| \le |\hat{y}_T(k) - \hat{y}_R(k) - \varepsilon_Q|$$
(38)

Substituting (28) and (32) in (38) to get

$$|y(k) - \overline{y}(k)| \le |k_v e_R(k) - k_v e(k) - \varepsilon_0| \tag{39}$$

Substituting from (34) and (31) to get

$$|y(k) - \overline{y}(k)| \le |(k_{\parallel} - 1) \varepsilon_{Q}| \tag{40}$$

25

In the ideal case with zero quantization and reconstruction errors,  $\varepsilon_Q$  is zero. Thus,

distortion is zero in an ideal case. In the non-ideal case, the quantization error  $\varepsilon_Q$  is

nonzero but is bounded by step\_size/2. Then (40) can be written as,

$$|y(k) - \overline{y}(k)| \le (k_{vmax} - 1) \mid step\_size/2| \tag{41}$$

Since the maximum singular gain matrix is normally selected less than one as

discussed in the previous theorem, the distortion (41) should be very small.

**Remark:** From (41), the following conclusions can be deduced:

The total distortion introduced by the proposed scheme is bounded and

made small by appropriately selecting the maximum singular value of the

gain matrix.

The distortion is dependent mainly on the quantization errors  $\varepsilon_Q$ .

**Theorem 4 (NADPCMC Performance):** Consider the NADPCMC scheme presented in

(28) through (35). Let us consider y(k) be a N sample vector of x bits each and that the

receiver side estimator requires the first K samples to initialize the regression vector.

Then the compression ratio, defined as the ratio of the amount of uncompressed data to

the amount of compressed data, is greater than one. Moreover, the proposed scheme will

render energy savings.

**Proof:** From (1), the resolution of the quantizer is given by

$$b = \log_2\left(\frac{E_{max} - E_{min}}{sten \ size}\right) \tag{42}$$

The estimation error has a smaller dynamic range compared to the original data. In other words,  $(E_{max} - E_{min}) \ll (Y_{max} - Y_{min})$ . Thus, b < x. The compression ratio is then given by

$$Compression\ ratio = \frac{Nx}{Kx + (N - K)b} \tag{43}$$

Since N >> K and x > b, the numerator in (43) is greater than the denominator and hence the compression ratio is greater than one.

This metric can be used to calculate the amount of energy savings that can be achieved. Assuming that each bit requires the same amount of energy  $E_b$  to be transmitted, the amount of energy required to send the uncompressed data is  $NxE_b$  and that required to send the uncompressed data is  $(Kx + (N - K)b)E_b$ . The total energy savings is given by

Energy savings (%) = 
$$\frac{(N-K)(x-b)}{Nx} * 100$$
 (44)

Again, since N >> K and x > b, a finite positive energy saving is achieved.

# C. Algorithm

The proposed algorithm for the data compression using the nonlinear adaptive estimator can be summarized as follows:

#### At the Transmitter:

Step 1: Initialize  $\emptyset_T(k)$  with first few data points

Step 2: Calculate estimate  $\hat{y}_T(k)$  from (28)

Step 3: Calculate estimation error e(k) from (29)

Step 4: Calculate parameter  $\hat{\theta}_T(k)$  update from (30)

Step 5: Quantize e(k) and send to receiver

Step 6: Update  $\emptyset_T(k)$  and repeat from step 2

# At the Destination:

Step 1: Initialize  $\emptyset_R(k)$  with first few data points

Step 2: Calculate estimate  $\hat{y}_{R}(k)$  from (32)

Step 3: Add  $\bar{e}(k)$  and  $\hat{y}_{R}(k)$  to reconstruct data as in (33)

Step 4: Calculate estimation error as difference between reconstructed and estimated signals as in (34)

Step 5: Calculate parameter  $\hat{\theta}_R(k)$  update from (35)

Step 6: Update  $\emptyset_R(k)$  and repeat from step 2

# IV. RESULTS AND DISCUSSION

It is important to identify the proper metrics to use for evaluating the performance of the proposed data compression scheme as data aggregator in a WSN environment. The performance of compression algorithms in general can be measured by using the following metrics:

• Quality / Percentage of distortion

- Compression ratio
- Latency Speed of compression and decompression
- Hardware support
- Energy savings

Quality is an important factor for lossy compression algorithms. It is quantified by percentage of distortion which is measured as the absolute difference between the original data and the reconstructed data. We calculate it as

Distortion = 
$$\left| \frac{y(k) - \bar{y}(k)}{y(k)} \right| * 100\%$$

Compression ratio is defined as the ratio of the amount of uncompressed data to the amount of compressed data and the additional overhead needed for reconstruction. For the algorithm to be advantageous, compression ratio has to be greater than one. The compression ratio is defined as

$$Compression \ ratio = \frac{total \ bits \ in \ y(k)}{total \ bits \ in \ e(k) and \ some \ y(k)}$$

Latency of the compression/decompression process also plays a vital role. The number of machine cycles utilized directly impacts the energy expended in computations. Further, applications such as landslide monitoring and fire detection, cannot tolerate delay in the reception of sensor data at the base station. Thus, the computation complexity of the algorithm directly affects the applicability of the algorithm for the sensor network case.

The memory requirement of the algorithm should also be considered while designing or porting compression algorithms for the sensor node. The code footprint and memory usage should be minimal.

However, the most important performance metric in the wireless sensor network case is the energy saving provided by the algorithm. It is calculated as the ratio of difference between the energy required to send the uncompressed data and that required for the compressed data to the energy required to send the uncompressed data and is expressed as a percentage.

Distortion and energy savings are not only important metrics for data compression but are also two most commonly used metrics for the evaluation of data aggregation schemes. As a result, these metrics are utilized to evaluate the proposed scheme both as a data compression algorithm and as a method for data aggregation in WSN.

The algorithm was tested in three levels. It was first implemented in MATLAB and was tested with different data sets. Then it was implemented in C to be tested with the Network Simulator (NS2). The topology is shown in Figure 2. Three clusters of 9 nodes are considered. The empty circles represent sensor nodes. The shaded ones indicate the cluster-heads and the striped one is the Base Station. Each cluster-head (CH) aggregates the data and routes it to the base station (BS). TCP agents were used for reliable packet delivery.

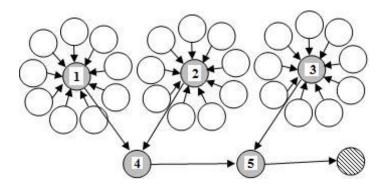


Figure 2. Network topology

The performance is compared with the popular lossless Huffman coding algorithm [11] and its differential variant. Delta coding was performed at the nodes followed by Huffman coding at the cluster-heads. Simulation results with plain quantization of scaled data are also put forth to analyze the advantages of quantizing the estimation error instead of the original data. The G.722 sub band linear ADPCM (8 bit) was also evaluated to highlight the improvement provided by the nonlinear estimation scheme.

The proposed NADPCMC algorithm (with 8 bit error encoding) was also implemented on a low-cost, fan-less single board computer called Beagle Board [19] running Ubuntu Linux and interfaced with Missouri S&T G4 motes. These motes provide a common platform for sensing, networking and data processing. The platform consists of an 8051 processor and an 802.15.4 (XBee) radio with micro Smart Digital (SD<sup>TM</sup>) flash storage, USB and RS-232 connectivity and an assortment of sensors. More information can be found in [20]. The motes form a network and use a static routing protocol to deliver data over multiple hops to a base station (as shown in Figure 3).

Initially, uncompressed data is packetized and sent over the network and the energy expended is calculated. Then the data is compressed online using the proposed algorithm, packetized and then routed to the base station. Once again the expended energy is calculated. The same static multihop routing protocol utilized for no compression case is also used. However, it is important to note that for data aggregation, the type of routing protocol is not relevant since proposed aggregation scheme is independent of routing. The data is recovered at the base station using the static routing protocol and the performance metrics are calculated and averaged over 10 trials.

Simulations and hardware experimental results performed with different data sets are now presented.

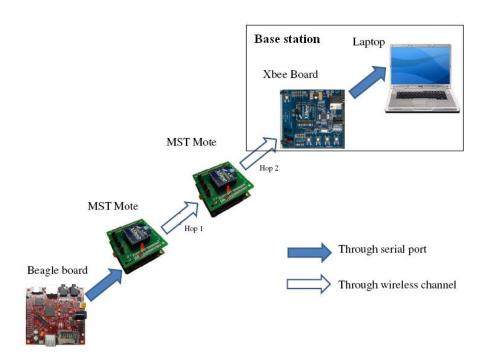


Figure 3. Hardware architecture

# A. Synthetic data

This data was generated in MATLAB to resemble data from an explosive sensor. Figure 4 shows the performance of the estimator. The estimate follows the highly non linear sequence with a minimal delay.

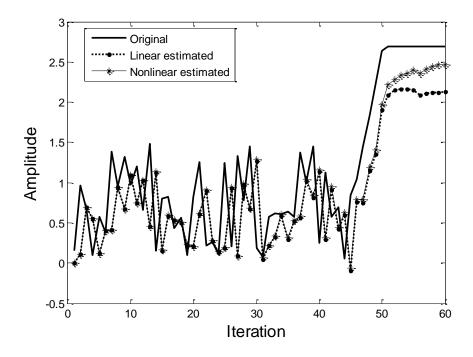


Figure 4. Estimator output

Figure 5 depicts the performance of the proposed NADPCMC scheme with 8 bit error encoding. The reconstructed data very closely resembles the original data. By contrast, Figure 6 illustrates the reconstruction error for different resolutions of the quantizer. The quality improves with the resolution. However, the overhead increases, which in turn causes an increase in the energy expended. These results show that reduced distortion implies higher compression ratio which translates into higher energy efficiency but at the expense of more memory and computation.

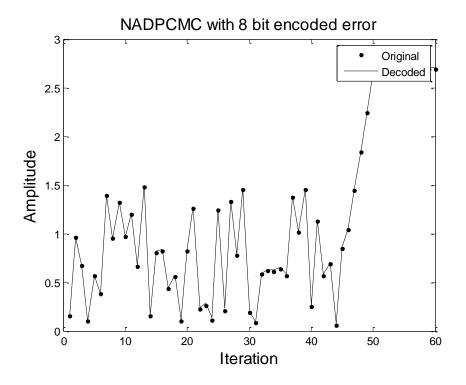


Figure 5. Reconstruction with 8 bit encoded error

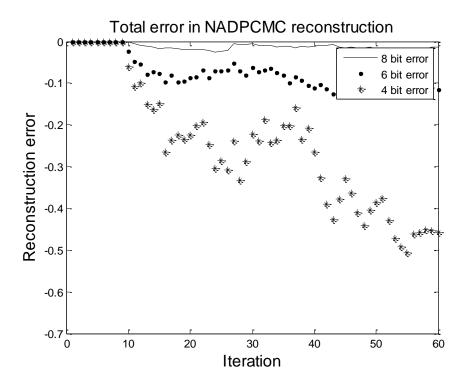


Figure 6. Total reconstruction error with different error encodings

Table 1 shows a comparison of the proposed performance metrics for different compression schemes on the synthetic data. Huffman coding is lossless and provides good compression for correlated data. But this data set is synthetic and mostly nonlinear. Hence Huffman coding does not offer much of an improvement. Further, there is an overhead of 480 bytes per node to send the codebook to the base station. Differential Huffman coding also suffers from the same problem.

Table 1. Performance metrics for synthetic data

Method	Compress- ion ratio	Energy savings at nodes	Energy savings at CH	Distortion	Overhead
Huffman	1.188	NA	15.850%	NA	480 bytes
Differential Huffman	1.086	5.9%	11.375%	NA	480 bytes
Scaling and 9 bit quantization	1.778	43.76%	43.76%	0%	0
Scaling and 8 bit quantization	2	50%	50%	2.111%	0
Scaling and 6 bit quantization	2.667	62.5%	62.5%	13.627%	0
Linear ADPCM	2	50%	50%	18.9%	0
NADPCMC with 8 bit encoding	1.846	45.83%	45.83%	1.67%	20 bytes
NADPCMC with 6 bit encoding	2.342	57.29%	57.29%	3.64%	20 bytes
NADPCMC with 4 bit encoding	2.667	62.50%	62.50%	7.28%	20 bytes

Direct quantization of scaled data at the nodes provides good compression at the expense of distortion. The linear ADPCM standard loses in terms of distortion. Since the data is very coarse, 8 bit quantization of scaled data is slightly better than the encoding of estimation error with 8 bits. However, the proposed scheme offers better performance for

lower resolutions of the quantizer. The overhead of 20 bytes (10 samples of 2 bytes each) corresponds to the first few samples of the original data required by the receiver side estimator. The parameter and regression vectors hold information about 10 previous samples.

The hardware experiments with the proposed NADPCMC scheme using the network illustrated in Fig. 3 with 8 bit error encoding provided an average energy savings of 41.04% at the source nodes when compared with no compression. Packetization added an overhead of 26 bytes per packet for routing purpose. This resulted in an average compression ratio of 1.6961 when compared with no compression.

#### B. River discharge data

River discharge data from the Amazon basin [21] was used to evaluate the algorithms. Figure 7 shows the performance of the proposed adaptive estimator which indicates that the proposed non linear estimator is able to track the data very well compared to its linear counterpart.

Figure 8 shows the performance of the proposed scheme with 8 bit error encoding. The reconstructed data very closely resembles the original data. Figure 9 shows the reconstruction error for different resolutions of the quantizer. As expected, the quality improves with the resolution.

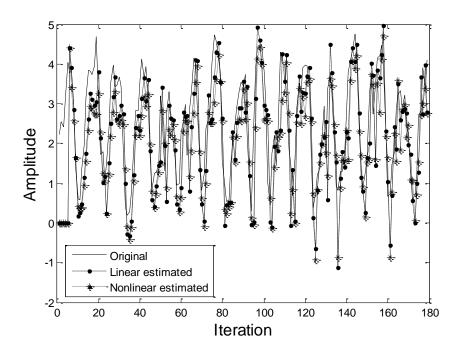


Figure 7. Output of estimator

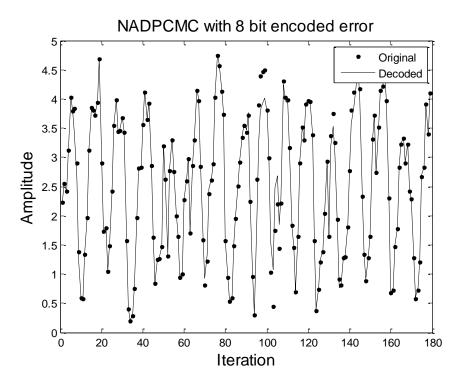


Figure 8. Reconstruction with 8 bit encoded error

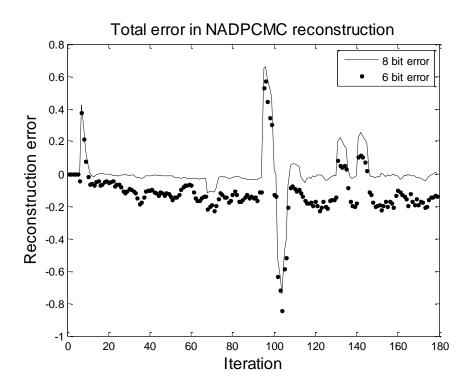


Figure 9. Total reconstruction error with different error encodings

Table 2 shows a comparison of the different performance metrics. ADPCM provides good energy savings at the expense of distortion. Once again 8 bit quantization of scaled data is slightly better than the encoding of estimation error with 8 bits. But in general, the proposed scheme offers better performance for lower resolutions of the quantizer. There is an overhead of 10 bytes corresponding to the 5 samples (2 bytes each) that have to be sent to initialize the receiver side estimator.

Table 2. Performance metrics for river-discharge data

Method	Compression ratio	Energy savings at nodes	Energy savings at CH	Distortion	Overhead
Huffman	1.453	NA	31.177%	NA	480 bytes
Differential Huffman	1.642	21.56%	39.099%	NA	480 bytes
Scaling and approximation	1.137	13.65%	11.65%	0.0657%	0
Scaling and 9 bit quantization	1.778	43.76%	43.76%	0.943%	0
Scaling and 8 bit quantization	2.000	50%	50%	2.0685%	0
Scaling and5 bit quantization	3.200	68.75%	68.75%	16.451%	0
Linear ADPCM	2	50%	50%	13.72%	0
NADPCMC with 8 bit encoding	1.9459	48.61%	48.61%	2.65%	10 bytes
NADPCMC with 6 bit encoding	2.5487	60.76%	60.76%	6.08%	10 bytes

The hardware experiments with the proposed NADPCMC scheme with 8 bit error encoding using the network shown in Figure 3 showed an average energy savings of 47.671% at the source nodes over no compression. The packet headers add an additional 26 bytes per packet for routing and hence a compression ratio of 1.911 was achieved when compared with no compression.

#### C. Audio data

The proposed NADPCMC algorithm was also tested with a wav file. Since audio data can tolerate a higher level of distortion, higher levels of compression can be

achieved at the expense of distortion. Figures 10 and 11 show the total reconstruction error with 8 bit and 6 bit encoding of errors, respectively.

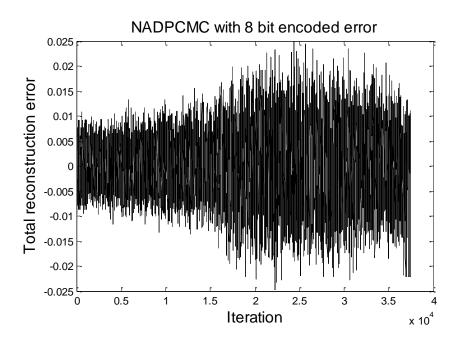


Figure 10. Total reconstruction error with 8 bit encoded error

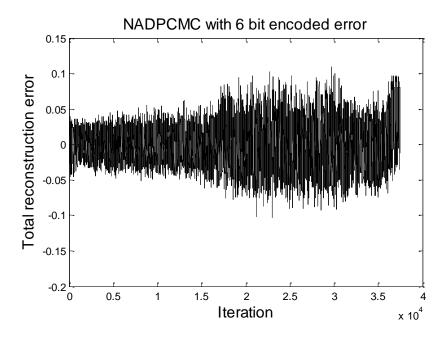


Figure 11. Total reconstruction error with 6 bit encoded error

The performance metrics are tabulated in Table 3. The proposed scheme outperforms the linear ADPCM implementation in terms of distortion. Additionally, the nonlinear ADPCM is better than the scaling and quantization approach in terms of distortion. The parameter and regression vectors were designed to hold 10 previous samples. These 10 samples cause an overhead of 20 bytes.

Table 3. Performance metrics for audio data

Method	Compress -ion ratio	Energy savings at nodes	Distortion	Overhead
Scaling and 8 bit quantization	2	50%	10.59%	NA
Scaling and 6 bit quantization	2.67	62.5%	46.28%	NA
5 bit linear ADPCM	3.199	68.74%	11.37%	NA
4 bit linear ADPCM	4	75%	23.14%	NA
3 bit linear ADPCM	5.332	81.25%	28.45%	NA
2 bit linear ADPCM	8	87.5%	35.86%	NA
NADPCMC with 8 bit encoding	1.9992	49.98%	2.04%	20 bytes
NADPCMC with 6 bit encoding	2.6653	62.48%	6.16%	20 bytes
NADPCMC with 4 bit encoding	3.997	74.98%	14.44%	20 bytes

# D. Geophysical data

The proposed algorithm was applied on geophysical data obtained from the Calgary corpus data set [22]. This dataset is widely used in evaluating compression

algorithms. The authors mention that the geophysical data is particularly difficult to compress because it contains a wide range of data values. Figures 12 and 13 show the total reconstruction error obtained with 8 bit and 6 bit encoding of errors, respectively.

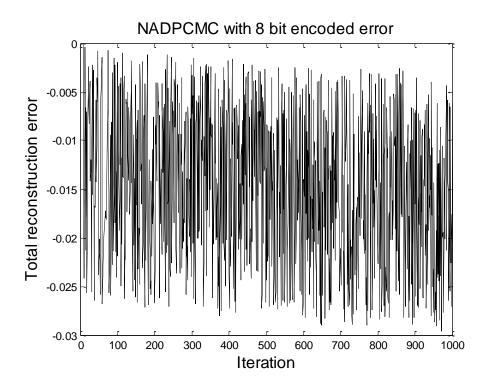


Figure 12. Total reconstruction error with 8 bit encoded error

Table 4 summarizes the performance metrics. The distortion was virtually unnoticeable with 8 bit error encoding unlike in linear ADPCM. 10 samples of 2 bytes each were used to initialize the estimator at the receiver. This adds an overhead of 20 bytes per sensor node.

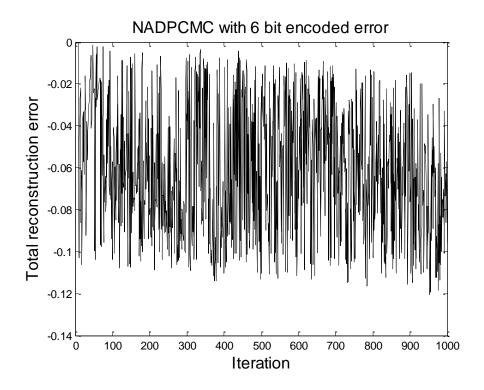


Figure 13. Total reconstruction error with 6 bit encoded error

Table 4. Performance metrics for geophysical data

Method	Compress -ion ratio	Energy savings at nodes	Distortion	Overhead
Scaling and 8 bit quantization	2	50%	4.36%	0
Scaling and 6 bit quantization	2.667	62.5%	13.42%	0
Linear ADPCM	2	50%	35.87%	0
NADPCMC with 8 bit encoding	2	50%	1.02%	20 bytes
NADPCMC with 6 bit encoding	2.667	62.5%	4.22%	20 bytes

#### E. Performance as an aggregation scheme

The results discussed so far dealt with compression performed at the source node only. This section examines the performance of the NADPCMC scheme when it is applied at different cluster-heads showing multiple levels of aggregation.

Figure 14 shows the topology considered. A cluster-head aggregates synthetic data received from three nodes and forwards it to the base station. NADPCMC with 8 bit error encoding was performed at the source nodes and NADPCMC with 6 and 4 bit error encodings were experimented at the Cluster-heads.

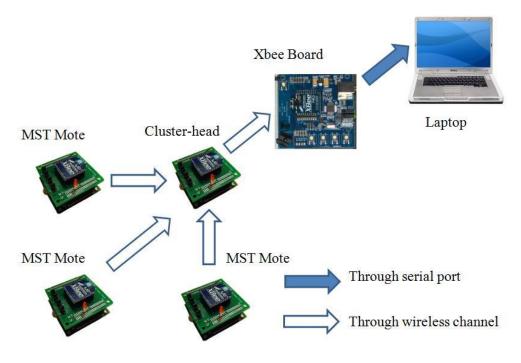


Figure 14. Hardware architecture

With compression only at the source level, all nodes reported an energy savings of 45.83% when compared with no compression. The average compression ratio at the source nodes was still a constant at 1.846. By repeating compression implemented on the

MST Motes at the Cluster-head, the amount of data and the energy expended in transmitting it are reduced further, however at the expense of distortion. When a second level of data aggregation was performed on the already compressed data at the first cluster-head, the compression ratio at the Cluster-head level was 2.526 amounting to an energy savings of 60.42%. There is an overhead of 20 bytes added by the NADPCMC scheme for every level of aggregation. The total distortion has increased from 1.67% to 4.60% with an additional level of aggregation which is considered to be acceptable. These results clearly demonstrate that with repeated compression, the distortion increases while energy savings and compression ratios improve.

In order to understand better the repeated compression along a route, more simulations on aggregation were performed using synthetic data and the topology described in Figure 2. Initially, 8 bit NADPCMC was performed at all source nodes and 6 bit NADPCMC was performed at cluster-heads 1, 2 and 3 respectively in order to evaluate the effect of encoding on energy levels and distortion. The energy savings at the aggregating and forwarding nodes (cluster-heads) increased from 45.83% (with no aggregation at cluster head) to 61.34%. The overall distortion for all the nodes was only 1.90%. Then 4 bit NADPCMC was tried on cluster-heads 1, 2 and 3 with 8 bit at the source nodes. The energy savings improved to 73.61% but the overall distortion summed up to 6.10%.

Next a third level of aggregation was introduced at CH5. 8 bit NADPCMC was performed at all source nodes, 6 bit NADPCMC was performed at cluster-heads 1, 2 and 3 and 4 bit NADPCMC was performed at cluster-head 5. Cluster-head 5 reported a total

energy saving of 74.54%. The overall distortion on the synthetic data was a tolerable at 7.01%.

In short, without aggregation and with compression at only the source nodes, all nodes reported the same energy savings of 45.83%. With one level of aggregation, the energy savings at the aggregating and the forwarding nodes improves to 61.34%. With a second level of aggregation, the energy savings at the aggregating node improves to 74.54%. Based on this study, it can be concluded that compression and data aggregation over multiple levels enhances the energy savings, increases distortion and overhead.

In order to evaluate the repeated compression along a route, a study was conducted with data flows of varying sizes being aggregated at cluster-head 5. Two of these were assumed to generate data from the river discharge data set. The third was made to transmit data from the wav file tested earlier. The other sources remained the same. All source nodes performed 8 bit NADPCMC. Cluster-heads 1, 2 and 3 performed 6 bit NADPCMC and cluster-head 5 performed 4 bit NADPCMC. Synthetic data was observed to have a distortion of 7.01%. River discharge data reported 4.83% distortion while voice degraded by 6.09%. From the aggregation point of view, the distortion levels are dependent on the resolution of the quantizer and the number of aggregations performed. Since each stage of aggregation is lossy, the quality declines with each added level.

Synthetic data propagated through more hops and was compressed thrice while the others were compressed only twice. This led to a higher degradation in synthetic data. This demonstrates that the distortion increases with the number of aggregation levels.

#### *F. Scalability*

Scalability tests are important with any protocol. Real world sensor networks may contain hundreds of nodes and the protocol should not fail during deployment in a dense network. Simple tests were made with NS2 to check the behavior of the aggregation scheme over networks of varying sizes (50 – 250 nodes). Different topologies with varying cluster sizes were created. All source nodes use 8 bit NADPCMC. First level cluster-heads use 6 bit NADPCMC and are one hop away from the sensor nodes. A second level of aggregation with 4 bit NADPCMC is performed at a cluster-head that is closest to the base station. Multiple data flows using synthetic, river discharge and audio data were created for each of these topologies.

Figure 15 shows the compression ratio at the second cluster-head level with increasing number of flows. In spite of increasing the number of flows, the overhead remains constant at 20 bytes for every compression. But with higher flows, the amount of data (N) flowing through the network is higher and the overhead (K) appears smaller. This leads to a slight increase in compression ratio. From (43), the maximum compression ratio that can be obtained with 'b' bit NADPCMC is when  $N \gg K$  and is given by

$$Max.Compression Ratio = \frac{x}{b} \tag{45}$$

Hence, 4 bit NADPCMC on 16 bit data (x = 16) would provide a maximum compression ratio of 4. Thus, the curve tends asymptotically to 4.0 with an increase in number of flows.

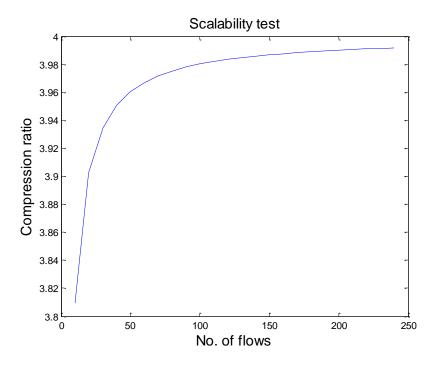


Figure 15. Dependency on number of flows

Figure 16 shows the compression ratio at the first cluster-head level. Each application of the NADPCMC scheme adds an overhead of 20 bytes. With increase in the network size, more clusters were created. This led to an increase in the number of times the compression scheme was applied. Since there is a small overhead associated with the scheme, the compression ratio decreases slightly with increase in network size for the same number of flows. Similarly, the average compression ratio decreases with an increase in the number of flows for a given network size due to added overhead with the number of flows although it is small.

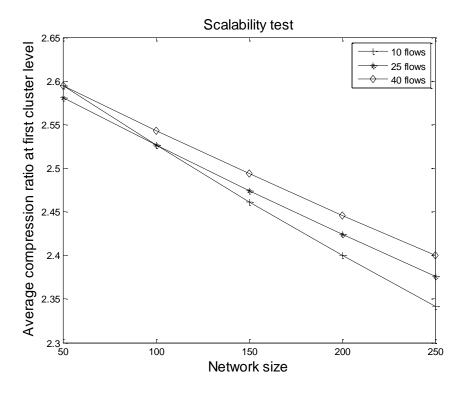


Figure 16. Average compression ratio at first cluster-head level

Figure 17 shows the compression ratio at the second cluster-head level. There is just a single node performing 4 bit NADPCMC in all the topologies considered. With increase in the amount of data flowing, the percentage of overhead associated with NADPCMC decreases. This leads to a slight increase in the compression ratio.

Another important inference can be made from the distortion values. Each data set suffered a constant level of distortion in these different scenarios. Synthetic data was distorted by 7.01%, river discharge data by 5.19% and audio data by 15.35%. This shows that the performance of the NADPCMC scheme is only dependent on the number of aggregation levels and not on the network size.

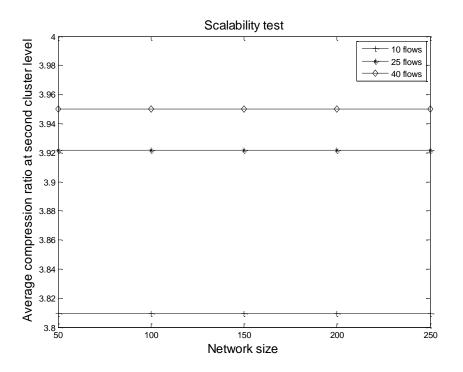


Figure 17. Average compression ratio at second cluster-head level

#### V. CONCLUSIONS

In this paper, a novel compression scheme based on adaptive estimation and quantization is introduced. Given a bounded sensor data, theoretical bounds on estimation error are derived and shown to be bounded when reconstruction error and quantization errors are bounded. Subsequently, distortion when using the proposed scheme has been proven to be bounded and small. The scheme was tested using multiple data sets including synthetic data, real world sensor data, and audio data. This proposed scheme is shown to offer energy savings of approximately 50% at each source node at the cost of around 2-3% distortion. Synthetic and river discharge data sets are coarser whereas the audio and geophysical data sets are very fine. Though direct quantization works well on

coarse data, it fails with data with fine resolution. However the NADPCMC scheme works fairly well on all these data sets. Hardware implementation of the proposed scheme using Missouri S&T motes confirms highly satisfactory performance. Then data aggregation through iterative compression was examined. Simulation results demonstrate that aggregation can improve the over-all energy savings with a small level of distortion. Moreover, the distortion depends mainly on the number of aggregation levels and not on the network size. This indicates that the scheme is scalable and can be deployed for large networks too.

#### **REFERENCES**

- [1] C. Alippi, G. Anastasi, C. Galperti, F. Mancini, M. Roveri, "Adaptive Sampling for Energy Conservation in Wireless Sensor Networks for Snow Monitoring Applications," *Int'l Conference on Mobile Adhoc and Sensor Systems*, pp. 1-6, Oct 2007.
- [2] R. Anguswamy, M. Zawodniok and S. Jagannathan, "A Multi-Interface Multi-Channel Routing (MMCR) Protocol for Wireless Ad Hoc Networks," *Proc. of the IEEE Wireless Communications and Networking Conference*, pp. 1-6, Apr 2009.
- [3] W. Heinzelman, A. Chandrakasan, and H. Balakrishnan, "Energy-Efficient Communication Protocol for Wireless Microsensor Networks," *Proc. Hawaii Intl. Conf. System Sciences*, pp. 3005-3014, Jan 2000.
- [4] P.M. Wightman, M.A. Labrador, "A3: A Topology Construction Algorithm for Wireless Sensor Networks," *Global Telecommunications Conference*, pp. 1-6, Nov-Dec 2008.
- [5] W. Mangione-Smith and P.S. Ghang, "A Low Power Medium Access Control Protocol for Portable Multi-Media Systems," *Proc. of the 3rd Int'l Workshop on Mobile Multimedia Communications*, Sept 1996.
- [6] R. Rajagopalan and P.K. Varshney, "Data-Aggregation Techniques in Sensor Networks A Survey," *IEEE Communications Surveys & Tutorials*, vol. 8, no. 4, pp. 48-63, 2006.

- [7] C.M. Sadler and M. Martonosi, "Data Compression Algorithms for Energy-Constrained Devices in Delay Tolerant Networks," *Proc. of the 4th Int'l conference on Embedded networked sensor systems*, pp. 265-278, 2006.
- [8] K. Sayood, "Introduction to Data compression," Third edition, Morgan Kaufmann Series in Multimedia Information and Systems, Dec 2005.
- [9] N. Kimura, S. Latifi, "A Survey on Data Compression in Wireless Sensor Networks", *Int'l Conference on Information Technology: Coding and Computing*, vol. 2, pp. 8 -13, 2005.
- [10] K. Barr and K. Asanovi'c, "Energy Aware Lossless Data Compression," *ACM Transactions on Computer Systems*, vol. 24, pp. 250-291, 2006.
- [11] D. A. Huffman, "A Method for the Construction of Minimum-Redundancy Codes," *Proc. of the I. R. E.*, vol. 40, pp. 1098-1101, 1952.
- [12] C. Alippi, R. Camplani, C. Galperti, "Lossless Compression Techniques in Wireless Sensor Networks: Monitoring Microacoustic Emissions," *Int'l Workshop on Robotic and Sensors Environments*, pp. 1-5, Oct 2007.
- [13] A. Deligiannakis, Y. Kotidis and N. Roussopoulos, "Compressing Historical Information in Sensor Networks," *Proc. of the 2004 ACM SIGMOD Int'l conference on Management of data*, pp. 527-538, 2004.
- [14] T. Banerjee, K. Chowdhury and D.P. Agrawal, "Distributed Data Aggregation in Sensor Networks by Regression Based Compression," *Int'l conference on mobile adhoc and sensor systems*, pp.-290, 2005.
- [15] C. Guestrin, P. Bodix, R. Thibaux, M. Paskin and S. Madden, "Distributed Regression: an Efficient Framework for Modeling Sensor Network Data," *3rd Int'l Symposium on Information Processing in Sensor Networks*, pp. 1-10, Apr 2004.
- [16] S. Lin, D. Gunopulos, S. Lonardi, V. Kalogeraki, "Applying LVQ Techniques to Compress Historical Information in Sensor Networks," *Proc. of the 2005 Data Compression Conference*, pp. 468-474, 2005.
- [17] P. Cummiskey, N.S. Jayant, and J. L. Flanagan, "Adaptive Quantization in Differential PCM Coding of Speech," *Bell Syst. Tech. J.*, vol. 52, pp. 1105-1118, Sept 1973.
- [18] S. Jagannathan, "Wireless Ad hoc and Sensor Networks: Protocols, Performance and Control," CRC Press, 2007.
- [19] Beagle Board Technical Specifications available online http://beagleboard.org/static/BBSRM\_latest.pdf accessed on Feb 2009.

- [20] J. Fonda, S. Watkins, S. Jagannathan, M. Zawodniok, "Embeddable Sensor Mote for Structural Monitoring," *SPIE 15th Annual Int'l Symposium on Smart Structures/NDE: Sensors and Smart Structures Technologies for Civil, Mechanical, and Aerospace Systems* 2008, vol. 6932, pp. 69322V.1-69322V.11, 2008.
- [21] Macrohydrological dataset for the Amazon basin available online ftp://ftp.ufv.br/dea/macrohydr accessed on Oct 2008.
- [22] Geophysical data from Calgary corpus available online <a href="http://corpus.canterbury.ac.nz/descriptions/#calgary">http://corpus.canterbury.ac.nz/descriptions/#calgary</a> accessed on Oct 2008.

# **PAPER**

# 2. HARDWARE VERIFICATION OF DATA AGGREGATION AND MULTI-INTERFACE MULTI-CHANNEL ROUTING PROTOCOL

Priya Kasirajan, Maciej Zawodniok and S. Jagannathan

**ABSTRACT** — The goal of wireless sensor networks is to gather information with high reliability and low energy. Considering the severe energy constraints of sensor nodes, data aggregation and energy-efficient routing are essential for improving the energy efficiency while maintaining packet delivery ratio. Typically, sensor nodes have radios that can handle many non-overlapping channels. This necessitates the use of a multichannel routing protocol that would balance the load evenly on multiple channels using a metric defined by throughput, end to end delay and energy utilization. We investigate a proactive routing protocol called the Multi-interface Multi-channel Routing (MMCR) protocol that uses Multi-Point Relay (MPR) nodes to forward data through the network, thus reducing the amount of communication overhead. To further improve the energy efficiency, data aggregation is performed along the routing path both at the source and at the cluster head through iterative application of the adaptive pulse coded modulationbased compression (NADPCMC) scheme. With the application of data aggregation, hardware results show that at least a 50% energy saving with less than 5% distortion is observed.

*Keywords:* Compression, Data Aggregation, Multi-channel Routing, Energy Efficiency, Wireless Sensor Networks

#### I. INTRODUCTION

Routing in wireless mesh networks has been explored for a long time since direct communication between all nodes in a network would be very expensive in terms of transmission cost which grows exponentially with distance. This necessitates cooperation between nodes so as to deliver packets over multiple hops from the source to the destination [1]. A common practice is to represent the total cost of a path as the sum of the link costs on the constituting links. Routing, then, aims at finding the path offering the lowest overall cost. Most current ad hoc routing protocols select paths that minimize hop count [2] [3] [4].

In static ad hoc wireless networks, minimal hop count paths can have degraded performance because they tend to include wireless links between distant nodes. These long wireless links can be slow or lossy leading to an unacceptable throughput. A routing algorithm can select better paths by explicitly taking into account the quality of wireless links. This is done in [5] by using a metric referred to as "Expected Transmission Count" (ETX) which is a measure of the loss rate of broadcast packets between pairs of neighboring nodes. In [6], "Per-hop Round Trip Time" (RTT) is utilized to measure the round trip delay seen by unicast probes between neighboring nodes. On the other hand, the work of [7] uses "Per-hop Packet Pair Delay" which is defined as the measured delay between a pair of back-to-back probes to a neighboring node. Expected Transmission Time (ETT) which is a function of the loss rate and the bandwidth of the link is used in [8] as the routing metric. The individual link weights are combined into a path metric called Weighted Cumulative ETT (WCETT) that explicitly accounts for the interference among links that use the same channel.

Wireless technologies, such as IEEE 802.11a [9], provide multiple nonoverlapping channels. Multiple channels have been utilized in infrastructure-based
networks by assigning different channels to adjacent access points, thereby minimizing
interference between them. However, multi-hop wireless networks have typically used a
single channel to avoid the need for co-ordination between adjacent pair of nodes, which
is necessary in a multi-channel network. For meeting the ever-increasing throughput
demands of applications, it is becoming important to utilize the entire available spectrum,
thereby motivating the development of new protocols specifically designed for multichannel operation. Wireless hosts have typically been equipped with one wireless
interface. However, a recent trend of reducing hardware costs [10] has made it feasible to
equip nodes with multiple interfaces and increase the capacity of wireless mesh networks.
With multiple radios, more concurrent communications can be possible in spectrum,
space and time. To maximize the benefit of multiple radios, ideally we should use routes
that contain low interference among the constituting links [11].

A multi-interface multi-channel routing protocol (MMCR) is proposed in [12]. It selects routes that enhance bandwidth utilization while maximizing energy efficiency and minimizing end-to-end delay. This proactive routing protocol operates independently of a particular scheme for receiver-based channel assignment. The protocol utilizes the concept of Multi-Point Relays (MPRs) similar to [13]. The scheme forwards packets using only the MPR nodes that are a fraction of the all one-hop neighbors. Hence, the routing complexity reduces for the same network size when compared with other proactive routing protocols. This paper deals with the hardware verification of the MMCR protocol on the Missouri S&T G4 motes [14].

The WSNs typically generate huge amount of heterogeneous data. The propagation of redundant data is costly in terms of system performance and results in energy depletion, network overloading, and congestion. While effective routing improves the packet delivery ratio, other methods for in-network data processing must be employed to reduce the number of messages relayed without much compromise on the fidelity. With the focus shifting towards multimedia sensor networks for surveillance, compression and aggregation techniques [15] [16] [17] are gaining importance every day. A lot of research has been done on developing tailored compression/aggregation techniques for WSNs. An ant colony approach is used for aggregation in [18]. In [19], wavelets are used to achieve data reduction. In this paper, a Nonlinear Adaptive Differential Pulse Coded Modulation-based Compression (NADPCMC) scheme [20] is used for compression and aggregation in conjunction with the multi-channel routing protocol.

This paper consists of the following sections. Section 2 describes the basic activities of the MMCR scheme. Section 3 illustrates the hardware details, packet types and the detailed algorithm of MMCR. A network of motes is used to relay dummy data and real-time voice and the results are presented. Section 4 outlines the NADPCMC algorithm and presents the results with compression and aggregation. Section 5 contains the concluding remarks.

#### II. BACKGROUND

In general, MMCR routing scheme comprises of three periodic phases:

- selection of MPRs for each node
- selection of routes
- data transfer through the selected MPRs

Simple local broadcast of HELLO messages is performed to discover one and two-hop neighbors and their corresponding costs. Then the MPR nodes are selected. This is followed by route selection globally for the whole network topology. Finally, the data is forwarded through the selected paths. There are five types of control packets in MMCR. They are listed below:

# 1) HELLO packet

Each node periodically broadcasts HELLO packets with information about its energy level to its neighbors until timeout. Every node finds its one hop neighbors and calculates the associated cost based on the received HELLO packets.

# 2) Acknowledgement (ACK) packet

Whenever a node receives a HELLO packet, it responds with an ACK packet with a list of its own one hop neighbors and their link costs. Thus, each node finds its two hop neighbors and calculates their associated cost based on the received ACK packets.

# 3) Topology Control (TC) packet

When the HELLO packet timeout elapses, each node selects MPRs among the one hop neighbors such that all two hop neighbors are covered. The Topology Control packet broadcasts this information to the one hop neighbors to indicate MPR selection information.

# 4) SWITCH packet

This packet is broadcasted by the destination node to all other nodes in the network to prompt them switch from a channel in the presence of interference.

# 5) BEAM packet

A BEAM packet is sent periodically by the destination node to all other nodes in the network. This packet is used to synchronize the real-time clock of all other nodes to facilitate accurate calculation of transmission delays.

The finer details of each phase in routing are presented next.

# A. Neighbor discovery

Nodes broadcast HELLO message locally to learn about their one-hop neighbors and their associated parameters such as energy, bandwidth and transmission delay. The header of the HELLO packet includes the available bandwidth and the transmission time. Figure 1 shows a flowchart of neighbor discovery stage.

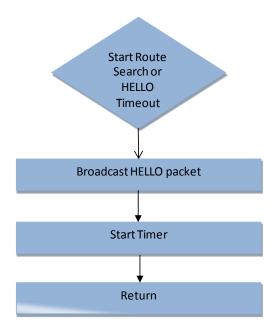


Figure 1. Neighbor discovery

The node receiving the HELLO packet can calculate the delay using the timestamp from the HELLO packet header. However, this requires time synchronization between the nodes. This is established by BEAM packets broadcasted periodically by the Base-station as shown in Figure 2.

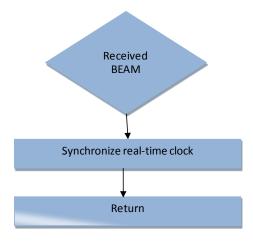


Figure 2. Handling BEAM packets

When a node receives a HELLO packet, it responds with an acknowledgement (ACK) packet as shown in Figure 3. The ACK packets contain the list of its neighbors and the energy utilization for each of these neighbors.

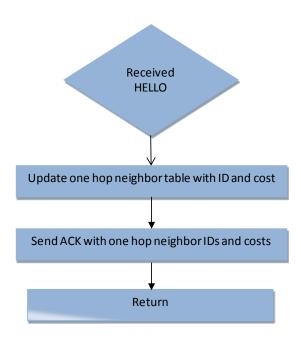


Figure 3. Sending ACK packets

When ACK packets are received, each node updates this information on available bandwidth, energy factor and the delay of the links from their neighbors in the 'neighbor table' as shown in Figure 4.

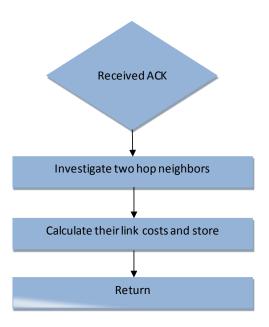


Figure 4. Handling of ACK packets

The utilization metric,  $U_{sn_2}^{MPR}$ , of the path from node s to a two-hop neighbor node  $n_2$  through a relay node  $n_1$  is calculated as follows:

$$U_{sn_2}^{MPR} = (B.F. * E.U.)/D \tag{1}$$

$$B.F. = B_A / B_S \tag{2}$$

$$E.U. = E_A^{n1} / E_{TX}^{n1->n2}$$
 (3)

where B.F. is a bandwidth factor between nodes s and  $n_1$  (MPR),  $B_A$  is an available (free) incoming bandwidth at the  $n_1$ ,  $B_S$  is an expected/requested outgoing bandwidth at the source node s, EU is the energy utilization between nodes  $n_1$  to  $n_2$ ,  $E_A^{n_1}$  is an available

energy at the relay  $n_I$  in Joules,  $E_{TX}^{n_I \to n_2}$  is an energy used to transmit message from  $n_I$  to  $n_2$ , and D is an end to end delay from node s to node  $n_I$  in seconds.

The metric optimization will maximize available bandwidth using bandwidth factor and minimize end-to-end delay using delay factor, *D*. Moreover, the metric will maximize the energy utilization term, which is expressed as energy depletion due to transmissions, thus increasing energy efficiency and lifetime of the nodes and network. The utilization factor given by bits per second is a direct measure of the total throughput of the link. Additionally, a route is selected if and only if the bandwidth factor for all the links on the path is greater than one. Consequently, the route associated with a flow guarantees sufficient bandwidth for the requested service.

### B. MPR selection

Each node in the network uses its 'neighbor table' to select multipoint relay (MPR) nodes from the one-hop neighbors to reach all the two-hop neighbors with minimum cost given by equation (1). The MPR selection metric proposed in [12] ensures that the paths through the MPRs optimize the energy consumption, delay, and bandwidth utilization. Additionally, the MPR selection algorithm ensures that there is sufficient available bandwidth to support the existing and new traffic flows. The optimal set of MPRs varies with traffic and network congestion. Hence, the nodes have to periodically recalculate the set of MPRs using updated data from HELLO and ACK packets. Figure 5 illustrates the MPR selection algorithm.

```
#1_hop_set is a set of one-hop neighbors of source
#2_hop_set is a set of two-hop neighbors of source
mpr_set = { }; # empty set
foreach dest_node IN 2_hop_set DO
  foreach mpr_candidate IN 1_hop_set
     if mpr_candidate connects source and dest_node
     then cost(mpr_candidate) = INFINITY;
     else cost(mpr_condidate) =
             COST (source TO mpr_candidate)
              + COST (mpr_candidate TO dest_node);
  end foreach;
  mpr_node = mpr_candidate with lowest cost;
  add mpr_node TO mpr_set;
  add to a routing table the mpr node as a next hop
  node toward dest_node;
end foreach;
# mpr_set holds the selected MPR nodes for the source
```

Figure 5. Pseudo-code for MPR selection Reprint from [12]

Figure 6 shows the MPR nodes (shaded in the figure) selected by this algorithm.

These nodes would be sufficient relays to reach all nodes in the network.

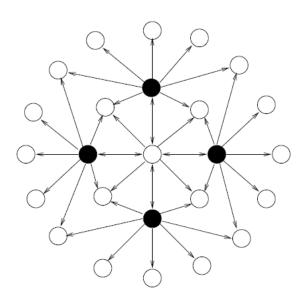


Figure 6. MPRs selected by the algorithm

# C. Topology discovery

The selected MPR nodes periodically transmit Topology Control (TC) messages (as shown in Figure 7) with corresponding link utilization factor data. The updates are propagated to all nodes in the network through the MPRs.

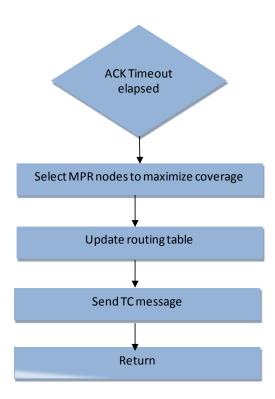


Figure 7. Sending TC packets

Upon receiving the TC messages, each node in the network records the information in the 'topology table' as shown in Figure 8.

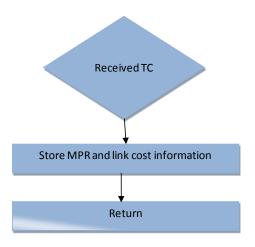


Figure 8. Handling of TC packets

#### D. Route selection and data transmission using the selected routes

Each node in the network uses its 'neighbor table' and 'topology table' to proactively compute the routes to all possible destinations. The protocol selects the path that has the least route cost metric while ensuring that the bandwidth factor is always greater than one for all the links on the path. This way, the algorithm eliminates routes that do not provide sufficient bandwidth to carry the traffic, thus implementing admission control mechanism. It ensures that the required flow data rate is supported throughout the whole route. The cost factor for a route with k intermediate MPRs nodes in the path is given by

$$C_{s,d} = \sum \left( C_{s,n_2}^{n_1}, C_{n_1,n_3}^{n_2}, \dots, C_{n_{k-1},n_k}^{n_{k-1}}, C_{n_k}^{n_k} \right) \tag{4}$$

$$C_{s,n_2}^{MPR} = 1/U_{s,n_2}^{MPR} \tag{5}$$

where  $C_{sn_2}^{MPR}$  is the cost metric between node s and its two-hop neighbor  $n_2 \in N^2(s)$  through the relay node  $n_1$  (MPR).

Once a route is found to the destination, the availability of multiple, independent channels and interfaces are exploited to perform load balancing for a particular link. If a particular link is suffering from interference, the receiver broadcasts a SWITCH packet to indicate a channel switch as shown in Figure 9.

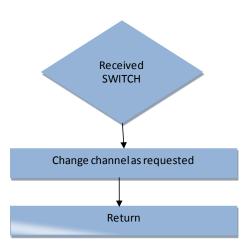


Figure 9. Handling of SWITCH packets

### III. IMPLEMENTATION OF THE ROUTING PROTOCOL

This section presents an overview of the hardware implementation of the MMCR protocol.

### A. Hardware description and limitations

Hardware verification of any algorithm is limited by hardware constraints such as processing capabilities, on-board battery capacity and supported interfaces. Use of specific hardware must be weighed against the precision, speed, and criticality of an algorithm's implementation. Constraints addressed for the implementation of the MMCR were use of low-power, small form-factor, and fast processing hardware. Hence the hardware should be energy conservative; performance oriented and should be of small form factor. Hence the processor architecture that can be deployed should be able to satisfy all these demands. The Silicon Laboratories 8051 variant family was selected for its ability to provide fast 8-bit processing, low-power consumption, and interface compatibility to peripheral hardware components. This provides high-speed processing, interconnectivity with the nodes, and a capable RF communications unit to facilitate a development platform for the ad hoc networks. Limitations that are incurred through the use of these 8051 variant family are a small memory space and limited floating point processing. In the next section, a description of the specifications for the hardware implemented nodes will be given.

The Generation-4 Smart Sensor Nodes (G4-SSN) [14], shown in Figure 10 were used as sensor nodes for implementation of the MMCR routing protocol. These were originally developed at Missouri S&T and subsequently updated at St Louis University (SLU). These motes provide a common platform for sensing, networking and data processing. The platform consists of an 8051 processor and an 802.15.4 (XBee) radio with micro Smart Digital (SD<sup>TM</sup>) flash storage, USB and RS-232 connectivity and an assortment of sensors. These nodes have 8K RAM and 128K flash memory that make it a

suitable choice for the hardware implementation. Table 1 gives a summary of the specifications of the G4-SSN. More information can be found in [14].



Figure 10. G4 mote

Table 1. Specifications of G4 mote

I <sub>c</sub> at 3.3V	35 mA
Flash memory	128 kB
RAM	8448 bytes
Form-Factor	100-pin LQFP
MIPS	100

### B. Implementation details

This section describes the implementation details of the MMCR routing protocol. Six types of packets are created – five of them being control packets and one data packet. The payload of each packet was prefixed with 2 headers – the XBee header and the routing header. The general structure of a packet is shown in Figure 11. The XBee header (shown in light gray) consists of information required by the XBee radios to process the

packets and the routing header (shown in dark gray) comprises of routing related information:

API Start Byte (0x7E)	1 byte	
API length	2 bytes	
API ID	1 byte	
API frame ID	1 byte	
API Destination	2 bytes	
API options	1 byte	
Start Byte (0x42)	1 byte	
Flag byte (0x75)	1 byte	
MAC Destination	2 bytes	
MAC Source	2 bytes	
Packet length	1 byte	
Destination ID	1 byte	
Source ID	1 byte	
Sequence number	1 byte	
Time Stamp	4 bytes	
Payload	Many bytes	
Payload CRC	1 byte	
Packet checksum	1 byte	

Figure 11. Packet structure

In addition, the DATA packet contains the application-specific headers and the payload. The application specific headers are shown in Figure 12. The total number of header bytes is 27 and the packet payload is fixed at 80 bytes. This leads to a 25.23% control overhead per packet.

Module Type	1 byte
Module length	1 byte
Report type	1 byte
Payload	Many bytes

Figure 12. Application specific header

### C. Node functions

Figure 13 shows the activity of a node that wishes to transmit data. A route search is started if no route is available to the required destination node. Otherwise the packet is forwarded to the appropriate MPR.

Figure 14 shows the activities of intermediate nodes. If the destination is a one-hop neighbor, the data packet is sent to it directly. Otherwise, it is relayed to the MPR that is closest to the destination node.

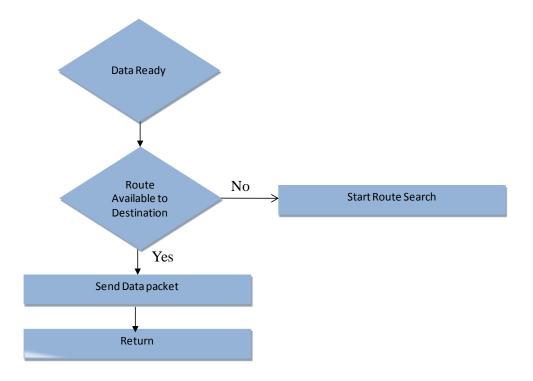


Figure 13. Activities of sender

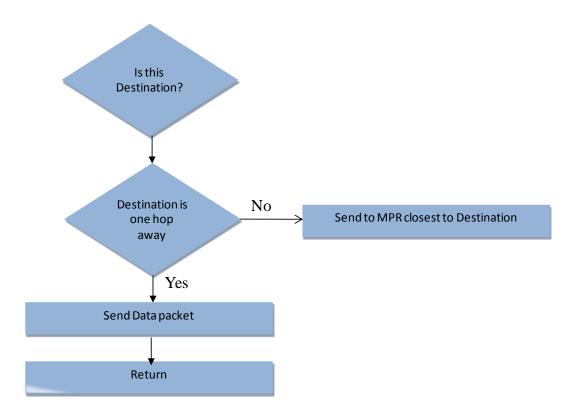


Figure 14. Activities of an intermediate node

Figure 15 shows the activities of a destination node. It broadcasts BEAM packets periodically to synchronize the real-time clock of all nodes in the network. Moreover, if the throughput falls below a certain threshold indicating interference, the receiver broadcasts a BEAM packet instructing the active nodes to switch to a different channel.

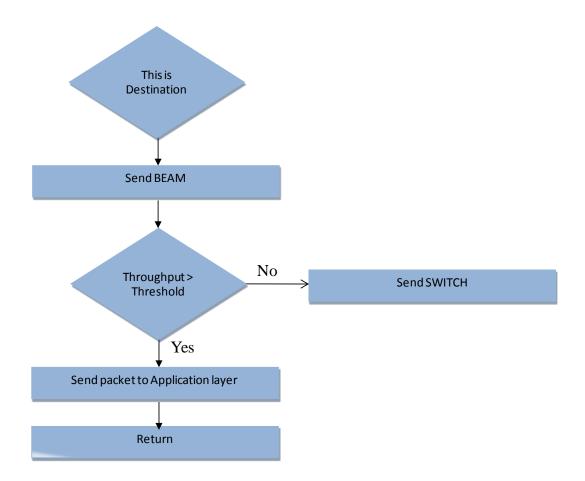


Figure 15. Activities of a destination node

# D. Protocol verification

Figure 16 shows the placement of nodes for demonstrating MMCR. Packets from source 1 were routed through intermediate nodes 1 and 2 to the destination. When intermediate node 2 was turned off, intermediate node 3 was used to relay packets.

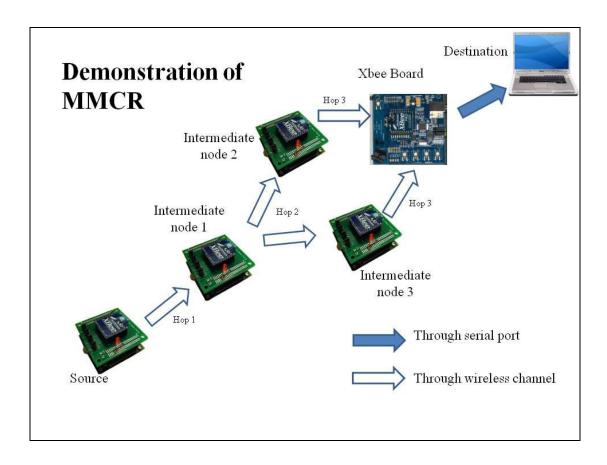


Figure 16. Demonstration of MMCR

The following were the performance metrics used to characterize the performance of the protocol:

- 1. Throughput Number of bits received per second
- 2. Drop rate Number of bits missing per second
- 3. End-to-end delay Total transmission delay in milliseconds
- 4. Jitter Variation in delay in milliseconds

Figure 17 shows the metrics measured when raw uncompressed data was sent over three hops through the network. Data was generated at 2.56 kbps. The total

transmission rate (including the packet headers) amounted to 3.424 kbps. All packets were received in sequence. Thus, the drop rate is zero. The average end to end packet delay over 3 hops was almost 20 ms. Table 2 lists the average values of the performance metrics. All reported values are the average over 10 trials.

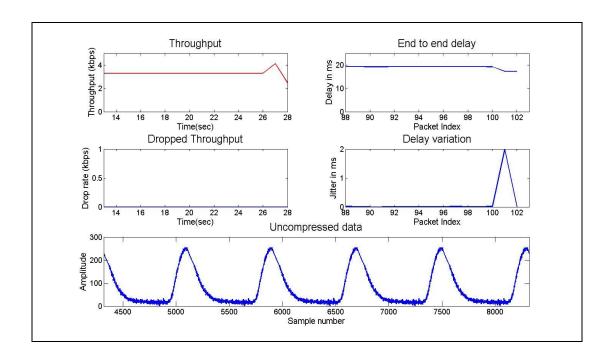


Figure 17. Performance metrics

Table 2. Average performance metrics for raw data

Data generation rate	2.56 kbps
Throughput (data transmission rate)	3.424 kbps
Drop rate	0
Average delay	20.2785 ms
Average jitter	0.7845 ms

Figure 18 shows the experimental setup used to verify the channel switching functionality. Two-hop communication was established using Channel 14 through MMCR as this channel is the default choice.

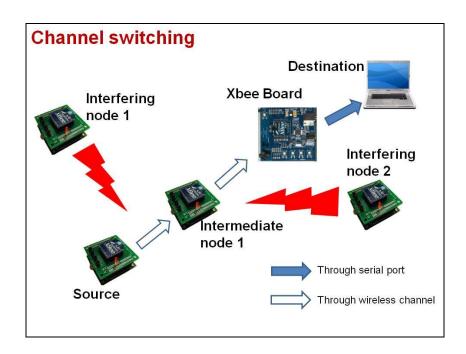


Figure 18. Demonstration of channel switching

When a different node which was broadcasting in the same channel was brought within communication range, a drop in throughput was observed by the Destination node. This triggered the sending of a SWITCH packet by the Destination node and the source and intermediate nodes switched communication to Channel 15 which is one among the list of available channels. Detailed selection of the channels is outside the scope of this

work. Figure 19 shows the switching of channels. The red throughput line indicates the use of Channel 14 and the blue one indicates Channel 15.

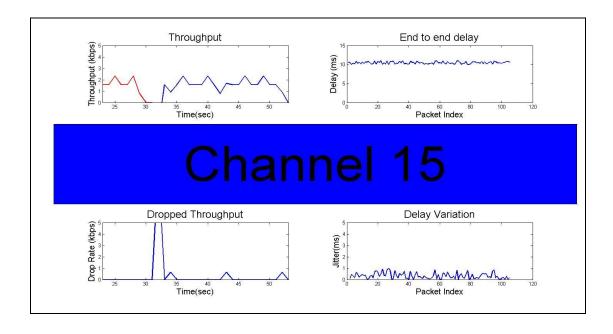


Figure 19. Channel switching

### E. Real-time voice with MMCR

The transmission of real-time voice through a bandwidth limited wireless channel is a very challenging test case. A low-cost, fan-less single board computer called Beagleboard [21] running Ubuntu Linux was interfaced with Missouri S&T motes. The microphone connected to the Beagleboard generated data at 128kbps. A G.721 4 bit ADPCM [22] was implemented to reduce the data rate to 32kbps. This data was packetized and sent over the MMCR network to a destination node connected to a

speaker through another Beagleboard. All communication between the Beagleboard and the motes were achieved through the UART interface. The Advanced Linux Sound Architecture (ALSA) library [23] was used to create the audio interfaces. The experiment setup is shown in Figure 20.

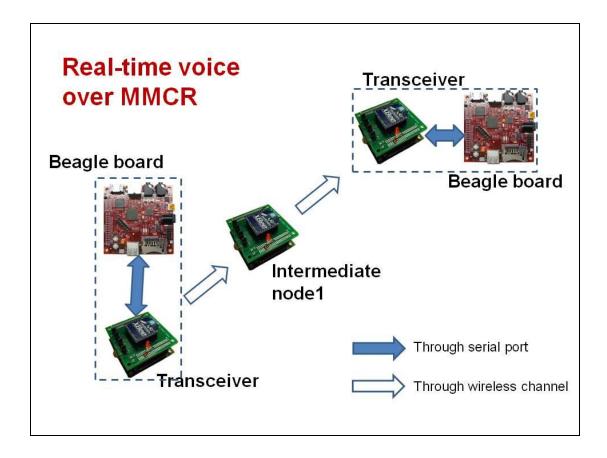


Figure 20. Real-time voice over MMCR

Figure 21 shows the results obtained. The voice data generation rate was 32kbps. Including the packet headers, the generation rate is almost 40kbps. From the figure, it can be observed that MMCR provides the required throughput with a few dropped packets.

The average end to end delay over 2 hops was about 8 ms with very low jitter which is considered to be acceptable for voice transfer. The performance metrics are listed in Table 3.

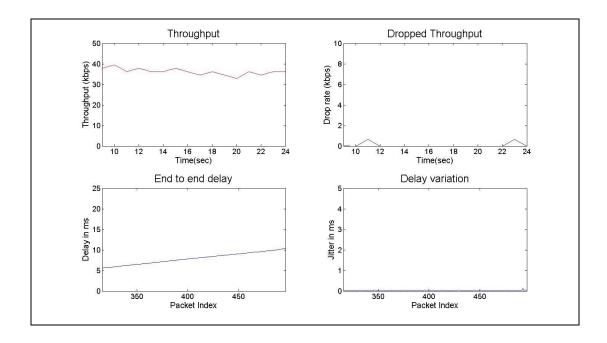


Figure 21. Performance metrics for real-time voice

Table 3. Average performance metrics for real-time voice

Data generation rate	32 kbps
Throughput (data transmission rate)	38.0688 kbps
Maximum drop rate	1.28 kbps
Average delay	8.357 ms
Jitter	0.0449 ms

When the data generation rate is increased further, the number of dropped packets starts increasing rapidly due to congestion in the network. These packet losses can be minimized by compression/aggregation techniques as will be discussed next. The next section describes the implementation of the NADPCMC scheme for this purpose.

#### IV. IMPLEMENTATION OF DATA AGGREGATION

This section provides a brief background of the NADPCMC scheme. This is followed by the hardware implementation details and results.

#### A. NADPCMC

A nonlinear estimator [20] is used to generate the estimate of the data sample at every instant. Adaptive estimation of the data sequence is performed by representing the data as a nonlinear autoregressive moving average sequence as

$$y(k+1) = \theta^{T}(k)\emptyset(k) + \varepsilon(y) \tag{1}$$

where  $\emptyset(k) \in \mathbb{R}^{n+1}$  is the basis function,  $\theta(k)$  is the unknown parameter vector, and  $\varepsilon(y) \in \mathbb{R}$  is the reconstruction error which is bounded by  $\|\varepsilon(y)\| \le \varepsilon_N$ . The estimated signal is represented as

$$\hat{y}(k+1) = \hat{\theta}(k)\emptyset(k) - k_v e(k) \tag{2}$$

where  $\hat{\theta}(k)$  is the estimated parameter estimate vector and e(k) is the estimation error.

The estimation error is then given by

$$e(k+1) = y(k+1) - \hat{y}(k+1)$$
(3)

Substituting for y and  $\hat{y}$  in (3) renders

$$e(k+1) = k_v e(k) + \tilde{\theta}^T(k) \varphi(k) + \varepsilon$$
(4)

where the parameter estimation error is defined by

$$\tilde{\theta}^T(k) = \theta(k) - \hat{\theta}(k) \tag{5}$$

Now consider the parameter update as

$$\hat{\theta}(k+1) = \hat{\theta}(k) + \alpha \emptyset(k) e^{T}(k+1)$$
(6)

Here  $\alpha$  is called the adaptation gain.

The difference e(k) between the actual and the estimated sample is quantized and sent to the destination. If the estimation error is small enough, the error can be represented with smaller number of bits than the original sample. This leads to compression.

At the receiver side, a similar estimator generates sample estimates. This estimator requires synchronization of the basis function  $\varphi(k)$  with a few correct initial samples from the transmitter. The error received from the transmitter is added to the estimate to reconstruct the data. Figure 22 shows the flowchart with the source functionality in blue and receiver functionality in red.

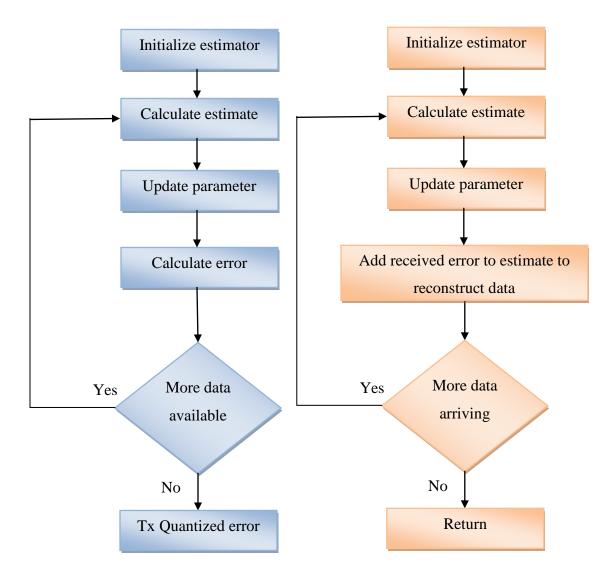


Figure 22. NADPCMC flowchart

# B. Zeolite sensors for explosive detection

Many multi-layer sensor mote architectures consist of a microcontroller (MCU) layer and several slave layers for interfacing with other systems and the application environment. Architectures such as the G4 SSN use direct pin assignment of the MCU to the bus pins and this leads to a limitation of the number of supportable slave devices. The G4 mote supports a single radio interface easily. It had the required UART interfaces for

a secondary radio but the architecture required some manual configuration tweaking and the second radio added to the processor load. These issues call for a new architecture that could support multiple slaves easily. The Missouri S&T M2 mote (shown in Figure 23) addresses these issues through the use of a serial bus to connect the layers.



Figure 23. M2 mote

A network of wireless motes connected to chemical sensors is capable of detecting, locating, and sending warning messages to appropriate decision makers about the presence of explosive threats. In this implementation, we use Missouri S&T motes connected to zeolite chemical sensor technology [24] as an input to the detection system. The zeolite sensors utilize a coated fiber optic tip that measure the reflectivity of the fiber as the chemical is introduced into the environment. An amplifier circuit translates this reflectivity into a 0-3V level and is read by an ADC on the sensor mote which measures the signal against several threshold levels. The sensor response to low concentration of

isopropanol molecule from a prototyped sensor circuit is shown in Figure 24. The first threshold level at 1/3 of the dynamic range is considered to be the Early Warning Level (EWL) and the second threshold at 2/3 the dynamic range is considered to be the Critical Warning Level (CWL).

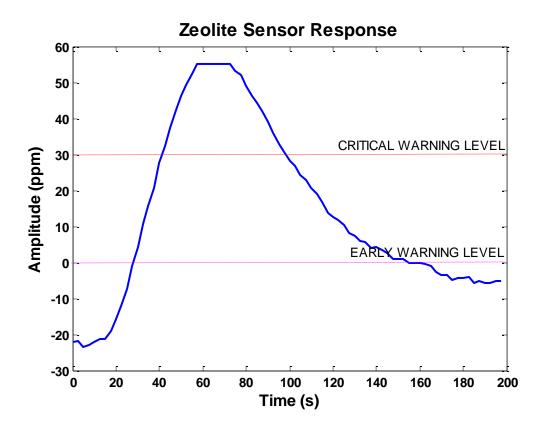


Figure 24. Prototyped sensor circuit response

#### C. Transmission of sensor data

The M2 mote generates sensor data at 2.56 kbps. The 8-bit NADPCMC was applied to reduce the transmission rate by almost half. This led to an energy saving of 46.67% with distortion limited to 0.78%. The transmitter and receiver side estimators used the first 5 data samples for initialization. However, this approach has a major

drawback. When a data packet is dropped, the receiver does not have certain e(k) and cannot reconstruct those samples. Since these samples are required to set the basis function  $\varphi(k)$ , it would be incorrectly set to zero. This leads to error in estimating the forthcoming  $\hat{y}(k)$  samples since  $\hat{y}(k)$  depends  $\varphi(k)$  (from equation (2)). Thus, the estimators on both sides would no longer be synchronized. This situation is shown in Figure 25 where the second packet in the sequence is dropped. The receiver side estimator does not have the appropriate regression vector to generate the next estimates, thus forcing the reconstructed samples deviating from the original data.

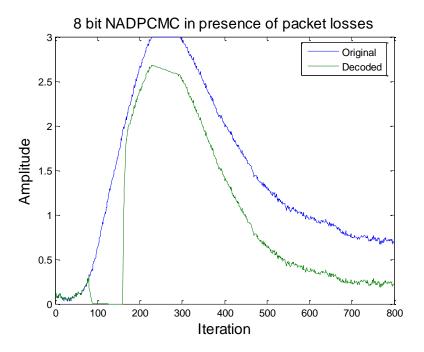


Figure 25. 8 bit NADPCMC in the presence of packet losses

To mitigate this problem, each packet has to be made independent of the previous packet. In other words, the basis function  $\varphi(k)$  of the receiver side estimator has to be reinitialized for each data packet. This way, even if a packet is dropped,  $\varphi(k)$  would be reset with 5 data samples through every other received packet. Thus, the estimated  $\hat{y}(k)$  for the other packets would be same at both transmitter and receiver side estimators. This would add an additional overhead to each packet, reducing the compression ratio from 1.957 to 1.875. In addition, the energy savings reduces from 48.91% to 46.67%. However, it improves the fault tolerance capability of the scheme in a significant manner. Figure 26 shows the performance of the modified NADPCMC scheme when the second packet in the sequence is dropped in the network.

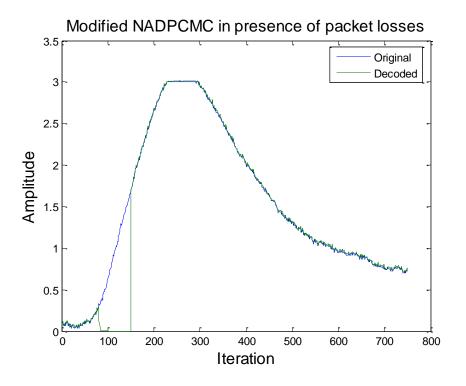


Figure 26. Modified 8 bit NADPCMC in the presence of packet losses

Next, aggregation was performed to further reduce transmission rates since there could be other data flowing through the network causing congestion. Figure 27 shows the experimental setup. Source 1 generates explosive sensor data at 2.56 kbps and Source 2 generated river discharge data from the Amazon basin [25] at 800 bps.

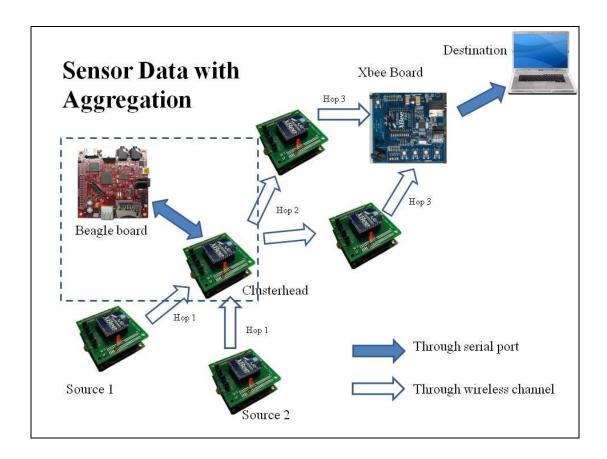


Figure 27. Demonstration of data aggregation

The compression ratio provided by NADPCMC depends, primarily, on the quantizer resolution. However, when the resolution of the quantizer is decreased, the distortion increases. Mathematical proofs of the same are available in [20]. The 8-bit

modified NADPCMC was used for compression at the source level while a 6 bit NADPCMC at the Cluster-head level reduced the transmission rate to 1.284 kbps for the aggregated flow. An energy savings of 56.99% was obtained and the distortion in both data sets was limited to less than 4%. When 4 bit NADPCMC was used at the Cluster-head level, the aggregated transmission rate was reduced to 856 bps. Including the packet headers, the throughput was approximately 1 kbps as shown in Figure 28. Though the energy savings improved to 71.43%, the distortion also increased to 8.21% for explosive sensor data and 10.9% for river discharge data. Table 4 summarizes these results.

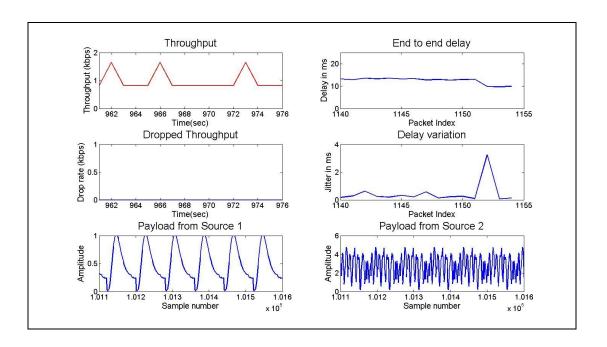


Figure 28. Performance metrics with 4 bit data aggregation

Table 4. Effect of data aggregation

	Data generation rate	Transmissi on rate	Compr ession ratio	Energy savings	Distortion
Uncompressed	2.56 kbps	3.424 kbps	NA	NA	NA
Compressed data	2.56 kbps	1.712 kbps	1.8751	46.67%	0.78% for sensor data 0.81% for river
Compressed and aggregated data – 6 bit NADPCMC	2.56 kbps	1.284 kbps	2.3250	56.99%	discharge data 3.58% for sensor data 2.78% for river
Compressed and aggregated data – 4 bit NADPCMC	2.56 kbps	856 bps	3.5002	71.43%	8.21% for sensor data 10.90% for river discharge data

Figure 29 shows the reconstructed explosive sensor data samples with different level of aggregation.

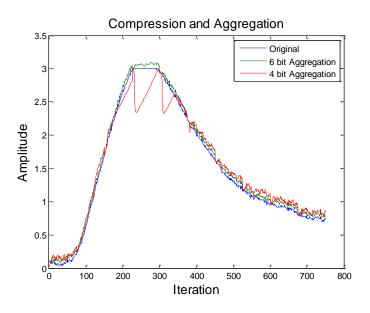


Figure 29. Reconstructed explosive sensor data

Figure 30 shows the reconstructed river discharge data samples with different stages of aggregation.

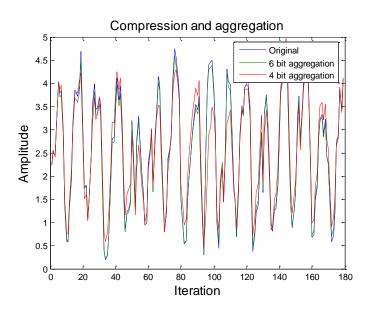


Figure 30. Reconstructed river discharge data

### D. Energy consumption

To achieve true energy efficiency, the energy expended in the compression algorithm must be less than that required to send the extra bits to the destination. Generally a processor consumes less power for such computations than a radio that is required to transmit a packet. In this case, Beagleboard uses the OMAP3530 processor [26] which has a power rating of 250mW. The CPU clock frequency is 720 MHz and the processor is capable of performing two 32-bit operations per cycle. In terms of Floating-point Operations per Second (FLOPS), this amounts to 1.44 gigaFLOPS. Each FLOP expends 0.17361nJ of energy. The NADPCMC encoding scheme consists of 7050 FLOPs

and the decoding scheme consists of 7425 FLOPs. Thus, the OMAP processor expends 1.224 micro Joules for every encoding and 1.289 micro Joules for every decoding.

The XBee radio uses a transmit power of 1mW for a 30m range [27]. This translates to 5 nJ per bit. Sending 160 data samples without compression would require 20.32 µJ of energy. With compression, the number of bytes is reduced by half, thus requiring 10.16 µJ is required for transmission. However, 2.45 µJ would be consumed in computations. Thus, the total energy expenditure with compression is 12.61 µJ. When aggregation is added, the number of bytes is cut down by another half and 5.08 µJ is required for transmission. And 1.224 µJ would be expended in computations. Thus, the total energy consumed is 6.304 µJ with data aggregation including computational overhead when compared to transmission alone. Table 5 lists the total energy consumption and the percentage savings. It indicates that even though some energy is consumed on computations, data aggregation provides significant overall energy savings to improve the network lifetime.

Table 5. Energy expenditure

	Energy consumption	Energy savings
Transmission of uncompressed data	20.32 μJ	NA
Transmission of compressed data using 8 bit NADPCMC	12.61 μJ	37.94%
Transmission of compressed and aggregated data – 4 bit NADPCMC	6.304 µЈ	68.98%

#### V. CONCLUSIONS

This paper dealt with the hardware verification of routing using MMCR protocol and data aggregation using NADPCMC scheme. Usage of multiple channels is shown to help balance the load and reduce packet losses. However, when the data generation rate increases which is normally observed in a WSN, there may be packets dropped due to congestion, thus necessitating compression and aggregation techniques with routing. A 8 bit NADPCMC reduces amount of data at source level by almost half. To make the scheme resilient to a lossy channel, a small overhead is added to each packet to reinitialize the receiver-side estimator. This is shown to improve the performance in a satisfactory manner. In-network aggregation further reduces the amount of data and improves network lifetime. However a small amount of distortion is involved. The aggregation scheme is proven to be truly energy efficient with the computations consuming a very small amount of power compared to the power required for transmission by the radio.

#### REFERENCES

- [1] R. Ramanathan and R. Hain, "Topology control of multihop wireless networks using transmit power adjustment," *Proc. of INFOCOM*, pp. 404 413, Mar 2000.
- [2] V. D. Park and M. S. Corson, "A highly adaptive distributed routing algorithm for mobile wireless networks," *Proc. of INFOCOM*, pp. 1405 1413, Apr 1997.
- [3] C. E. Perkins and P. Bhagwat, "Highly dynamic destination-sequenced distance vector routing (dsdv) for mobile computers," *Proc. of SIGCOMM*, pp. 234 244, Sept 1994.

- [4] C. E. Perkins and E. M. Royer, "Ad-hoc on-demand distance vector routing," *Proc. of WMCSA*, pp. 90 100, Feb 1999.
- [5] D. De Couto, D. Aguayo, J. Bicket, and R. Morris, "High-throughput path metric for multi-hop wireless routing," *Proc. of MOBICOM*, Vol. 11, pp. 419 434, Sept 2003.
- [6] A. Adya, P. Bahl, J. Padhye, A. Wolman, and L. Zhou, "A multiradio unification protocol for IEEE 802.11 wireless networks," *Proc. of BroadNets*, pp. 344 354, 2004.
- [7] S. Keshav, "A Control-theoretic approach to flow control," *Proc. of SIGCOMM*, pp. 3 15, Sept 1991.
- [8] R. Draves, J. Padhye, and B. Zill, "Routing in multi-radio, multihop wireless mesh networks," *Proc. of MobiCom*, pp. 114 128, Sept 2004.
- [9] IEEE Standard for Wireless LAN-Medium Access Control and Physical Layer Specification, P802.11, 1999.
- [10] P. Bahl, A. Adya, J. Padhye, and A. Wolman, "Reconsidering wireless systems with multiple radios," *ACM CCR*, Vol. 34, pp. 39 46, Jul 2004.
- [11] P. Kyasanur, and N.H. Vaidya, Routing and link-layer protocols for multi-channel multi-interface ad hoc wireless networks, ACM SIGMOBILE Mobile Computing and Communications Review, v.10 n.1, p.31-43, Jan 2006.
- [12] R. Anguswamy, M. Zawodniok and S. Jagannathan, "A multi-interface multi-channel routing (MMCR) protocol for wireless ad hoc networks," *Proc. of the IEEE Wireless Communications and Networking Conference*, pp. 1-6, Apr 2009.
- [13] A. Qayyum, L. Viennot and A. Laouiti, "Multipoint relaying for flooding broadcast messages in mobile wireless networks", *Proc. of HICSS*, pp. 3866 3875, Jan 2002.
- [14] J. Fonda, S. Watkins, S. Jagannathan, and M. Zawodniok, "Embeddable sensor mote for structural monitoring," *SPIE 15th Annual Int'l Symposium on Smart Structures/NDE: Sensors and Smart Structures Technologies for Civil, Mechanical, and Aerospace Systems* 2008, vol. 6932, pp. 69322V.1-69322V.11, 2008.
- [15] K. Dasgupta, K. Kalpakis, and P. Namjoshi, "An efficient clustering-based heuristic for data gathering and aggregation in sensor networks," *Proc. Of WCNC*, Vol. 3, 16-20 Mar 2003.
- [16] Krishnamachari, D. Estrin, and S. Wicker, "The impact of data aggregation in wireless sensor networks," *Int'l Workshop on Distributed Event-Based Systems*, pp. 414 415, Jul 2002.
- [17] Boulis, S. Ganeriwal, and M.B. Srivastava. "Aggregation in sensor networks: an energy-accuracy trade-off," *Proc. of IEEE Sensor Network Protocols and Applications*, pp. 128 138, May 2003.

- [18] R. Misra, C. Mandal, "Ant-aggregation: ant colony algorithm for optimal data aggregation in wireless sensor networks," *Proc. of IFIP Int'l Conference on Wireless and Optical Communications Networks*, pp. 1 5, 2006.
- [19] W. Cai, M. Zhang, "Data aggregation mechanism based on wavelet-entropy for wireless sensor networks," *Proc. of WiCOM*, pp. 1 4, Oct 2008.
- [20] K. Priya, C. Larsen, S. Jagannathan, "A new adaptive compression scheme for data aggregation in wireless sensor networks," to appear in *IEEE WCNC 2010*.
- [21] Beagle Board Technical Specifications available online http://beagleboard.org/static/BBSRM\_latest.pdf accessed on Feb 2009.
- [22] G.721 32 kbps ADPCM Specifications available online <a href="http://www.itu.int/rec/T-REC-G.721/en">http://www.itu.int/rec/T-REC-G.721/en</a> accessed on Feb 2009.
- [23] Advanced Linux Sound Architecture Library available online <a href="http://www.alsa-project.org">http://www.alsa-project.org</a> accessed on Feb 2009.
- [24] H. Xiao, Z. Jian, D. Junhang, L. Ming, L. Robert and Van Romero, "Synthesis of MFI zeolite films on optical fibers for detection of chemical vapors," Optics Letters, vol. 30, no. 11, pp. 1270-1272, 2005.
- [25] Macrohydrological dataset for the Amazon basin available online <a href="ftp://ftp.ufv.br/dea/macrohydr">ftp://ftp.ufv.br/dea/macrohydr</a> accessed on Oct 2008.
- [26] OMAP35x Applications Processor Technical Reference Manual available online http://focus.ti.com/lit/ug/spruf98d/spruf98d.pdf accessed on Nov 2009.
- [27] XBee®/XBee-PRO® SE (Smart Energy) RF Modules Specifications available online <a href="http://www.digi.com/standards/smart-energy/assets/90033931\_A.pdf">http://www.digi.com/standards/smart-energy/assets/90033931\_A.pdf</a> accessed on Nov 2009.

#### **SECTION**

#### 2. CONCLUSIONS AND FUTURE WORK

This thesis explores the possibility of energy efficiency improvement in wireless sensor networks through data aggregation and efficient multi-channel routing. Since existing compression schemes are tailored for specific types of data, a novel generic compression scheme called NADPCMC based on adaptive nonlinear estimation and quantization is developed in the first paper. Theoretical bounds on estimation error are derived using Lyapunov theory and the total distortion is shown to be dependent on the quantization error and the maximum singular value of the gain matrix. The scheme is tested using multiple data sets with different resolutions and offers energy savings of approximately 50% at each source node at the cost of around 2-3% distortion. Then data aggregation through iterative compression is examined. Simulation results demonstrate that aggregation can further improve the over-all energy savings with a small level of distortion. Moreover, the distortion depends mainly on the number of aggregation levels and not on the network size. This indicates that the scheme is scalable and is deployable in larger networks.

While data aggregation helps in reducing energy consumption, efficient routing is required to guarantee appropriate quality of service. The second paper deals with the hardware verification of a proactive multi-channel routing protocol. MMCR is implemented on Missouri S&T G4 motes by appropriately weighing the hardware capabilities and limitations in memory size, processing power, energy consumption, form factor and interface compatibility with other hardware components. The protocol uses

multiple channels, thereby providing high data rates and short end-to-end delays with fewer dropped packets. Compression and aggregation through NADPCMC further improve the throughput and energy savings. Moreover, NADPCMC is modified to be made resilient to channel conditions through a minor overhead. It is also observed that the energy consumed in the NADPCMC encoding and decoding algorithms are much lesser than the energy required for transmitting the extra bytes making the compression/aggregation worthwhile.

Future work could be directed towards achieving different quality of service levels for different types of data. NADPCMC based compression/aggregation scheme provides a constant a constant quality of service (i.e., constant distortion) for each flow. In our experiments, the estimation parameters were designed to achieve less than 10% distortion for each flow. In reality, voice and video signals can handle much higher distortion than data. Thus, the estimator parameters can be data dependent. Though such tunable compression/aggregation adds a small overhead in terms of communicating the estimation parameters to the destination, it would be able to satisfy the quality of service requirements of each flow better.

### **APPENDIX**

# **SOURCE CODE ON CD-ROM**

### 1. INTRODUCTION

Included with this Thesis is a CD-ROM, which contains the source code for the NS2 simulations and hardware implementation. The file "Info.txt" contains a short description of the source code. All documents have been prepared as Microsoft Word document files. An outline of the contents of the CD-ROM is as follows.

### 2. CONTENTS

Info.TXT

NS2 simulation:

datacompress.DOC

Hardware implementation:

routing\_mmcr.DOC

## **VITA**

Priya Kasirajan was born on July 18, 1985 in Tamilnadu, India. She received her Bachelor of Technology degree in Electronics and Communication Engineering from Pondicherry Engineering College in May 2006. She subsequently worked as a Software Engineer with Robert Bosch Engineering and Business Solutions Ltd. until July 2008. She began pursuing her graduate studies in Electrical Engineering at Missouri University of Science and Technology in the Fall of 2008 and graduated in May 2010. She is a member of Tau Beta Pi and can be reached at Priya.Kasirajan@gmail.com.

## **Source code of NS2 implementation:**

datacompress.cc – new traffic type to send compressed data.

```
/* -*-
            Mode:C++; c-basic-offset:8; tab-width:8; indent-tabs-mode:t -*-
* /
* Copyright (c) Priya Kasirajan 2009. All rights reserved.
* /
#ifndef lint
static const char rcsid[] =
    "@(#) $Header: /nfs/jade/vint/CVSROOT/ns-2/tools/datacompress.cc,v 1.0
2008/20/12 18:45:32 Priya Exp $ (MST)";
#endif
#include <stdio.h>
#include <math.h>
#include <stdlib.h>
#include <string.h>
#include "datacompress.h"
//#define HUFFMAN
                        /* switch between Huffman coding and NADPCMC */
#define DC DEBUG /* enable debug mode */
#ifdef HUFFMAN
//#include " uffman.h"
#endif /* HUFFMAN */
float data[] =
{2231.6,2554.6,2416.7,3128.3,4031.5,3790.0,3842.4,2904.2,1379.9,597.0,573.7,1
334.8,1970.7,3125.9,3847.3,3813.4,3727.7,3946.9,4681.5,2899.7,1732.5,1784.2,1
048.7,1487.3,2420.9,3539.2,3984.5,3440.1,3460.4,3677.8,3433.6,1576.3,404.9,19
2.5,281.4,755.2,1967.2,2819.5,2827.5,3557.7,4116.4,3647.2,3931.9,2855.3,1626.
7,838.1,1244.8,1271.3,1464.0,3195.6,2624.9,1309.5,2766.3,3300.8,2755.9,1993.0
,1643.5,941.3,1002.3,2267.1,2593.3,2979.6,1704.4,2865.2,3290.2,4144.4,3971.2,
2848.7,1585.9,814.8,1219.7,2369.9,2613.5,2890.5,4027.0,4739.8,4570.3,4125.4,3
731.2,1569.6};
/* multiply two floating point vectors of length 5 */
float mult(float a[], float b[])
 int i;
 float ret = 0;
  for (i = 0; i < 5; i++)
   ret = ret + a[i]*b[i];
  return ret;
/* compute tansig of a float value */
float tansig(float x)
  return (2/(1+\exp(-2*x))-1);
```

```
/* Perform 8 bit NADPCMC */
int encode()
 int i,j;
 float phi[] = \{0, 0, 0, 0, 0\};
  float e[80][5], o[80][5], x[80], err[80];
  float a = 0.0018;
 float temp = 0;
  float k[] = \{0.4, 0.3, 0.2, 0.1, 0.05\};
  unsigned char quant[80];
  for (i = 0; i < 80; i++)
    data[i] = data[i]/1000;
    x[i] = 0;
    for (j = 0; j < 5; j++)
      e[i][j] = 0;
      o[i][j] = 0;
  }
  //iteration 1
  e[0][0] = data[0]-x[0];
  phi[0] = data[0];
  //iteration 2
  temp = mult(phi,e[0]);
  for (i = 0; i < 5; i++)
    o[1][i] = o[0][i] + a * temp;
  x[1] = mult(o[1], phi);
  e[1][0] = data[1] - x[1];
  e[1][1] = data[0] - x[0];
  phi[0] = data[1];
 phi[1] = data[0];
  //iteration 3
  temp = mult(phi, e[1]);
  for (i = 0; i < 5; i++)
    o[2][i] = o[1][i] + a * temp;
  x[2] = mult(o[2], phi);
  e[2][0] = data[2] - x[2];
  e[2][1] = data[1] - x[1];
  e[2][2] = data[0] - x[0];
  phi[0] = data[2];
  phi[1] = data[1];
  phi[2] = data[0];
  //iteration 4
  temp = mult(phi,e[2]);
  for (i = 0; i < 5; i++)
    o[3][i] = o[2][i] + a * temp;
```

```
}
x[3] = mult(o[3], phi);
e[3][0] = data[3] - x[3];
e[3][1] = data[2] - x[2];
e[3][2] = data[1] - x[1];
e[3][3] = data[0] - x[0];
phi[0] = data[3];
phi[1] = data[2];
phi[2] = data[1];
phi[3] = data[0];
//iteration 5
temp = mult(phi, e[3]);
for (i = 0; i < 5; i++)
  o[4][i] = o[3][i] + a * temp;
x[4] = mult(o[4], phi);
e[4][0] = data[4] - x[4];
e[4][1] = data[3] - x[3];
e[4][2] = data[2] - x[2];
e[4][3] = data[1] - x[1];
e[4][4] = data[0] - x[0];
phi[0] = data[4];
phi[1] = data[3];
phi[2] = data[2];
phi[3] = data[1];
phi[4] = data[0];
for (i = 0; i < 5; i++)
  x[i] = 0;
//other iterations
for (i = 5; i < 80; i++)
  x[i] = mult(o[i-1], phi) + mult(k, e[i-1]);
  for (j = 0; j < 5; j++)
    e[i][j] = data[i-j] - x[i-j];
  temp = mult(phi,e[i-1]);
  for (j = 0; j < 5; j++)
    o[i][j] = o[i-1][j] + a*temp;
  phi[0] = tansig(data[i]);
  phi[1] = e[i][0];
  phi[2] = e[i-1][0];
  phi[3] = e[i-2][0];
  phi[4] = e[i-3][0];
}
// estimation over - calc error
//8 bit quantization
for (i = 0; i < 80; i++)
  err[i] = data[i] - x[i];
  if (err[i] > 1.9718)
    quant[i] = 255;
  else if (err[i] < -1)
    quant[i] = 0;
```

```
else
      quant[i] = ((err[i]+1)/0.0117);
#ifdef DC DEBUG
    printf("%d ", quant[i]);
#endif /* DC DEBUG */
#ifdef DC DEBUG
 printf(\overline{``}\n'');
#endif /* DC DEBUG */
return (5*sizeof(data[i]) + (i-5)*sizeof(quant[i]));
}
/* implement a source which takes packets, compresses them and generates
packets of
 * the compressed size. It is parameterized by packet size and interval.
static class DCTrafficClass : public TclClass {
      DCTrafficClass() : TclClass("Application/Traffic/Datacompress") {}
      TclObject* create(int, const char*const*) {
            return (new DC Traffic());
      }
} class dc traffic;
DC Traffic::DC Traffic() : seqno (0), fp1(NULL), compress (0), tsize(0),
isClusterHead (0)
      bind_bw("rate_", &rate_);
bind("maxpkts_", &maxpkts_);
      bind("isClusterHead_", &isClusterHead_);
      bind("compress ", &compress);
}
int DC Traffic::command(int argc, const char*const* argv) {
    if(argc==3){
        if (strcmp(argv[1], "payload") == 0) {
            if ((fp1 = fopen(argv[2],"r")) == NULL) {
                   fprintf(stderr, "File open error");
                   return 1;
            }
            return 0;
    return Application::command(argc, argv);
}
void DC Traffic::init()
      interval_ = (double)(40 << 3)/(double)rate;</pre>
#ifdef DC DEBUG
      printf("Interval is %ld \n", interval );
#endif /* DC DEBUG */
      if (agent_)
            agent ->set pkttype(PT TCP);
```

```
}
void DC Traffic::stop()
#ifdef DC DEBUG
     printf("total size %d %d\n",tsize, seqno );
#endif /* DC DEBUG */
    fclose(fp1);
    running = 0;
void DC Traffic::start()
      init();
      running = 1;
      // Clusterhead has to wait for aggregation
      if (isClusterHead_ == 0) {
    nextPkttime_ = next_interval(size_);
            timer .resched(nextPkttime);
      }
      else
#ifdef DC DEBUG
            printf("Clusterhead here...\n");
#endif /* DC DEBUG */
      // Enable line below if you want to send immediately upon start
      //timeout();
double DC_Traffic::next_interval(int& size) {
      double t = interval_;
      int len = 2000, temp = 0;
#ifdef HUFFMAN
      FILE *fp2, *fp3;
      if (compress == 1) {
            fp2 = fopen("./outputfile","r");
            fp3 = fopen("./dummyfile","w");
             uffman decode file(fp2,fp3);
            fclose(fp2);
            fclose(fp3);
      if (((fp2 = fopen("./temp","w")) != NULL) &&
      ((fp3 = fopen("./outputfile","w")) !=NULL)) {
            char line[2000], newline[2000];
            if (!feof(fp1)) {
                   fgets(line, len, fp1);
                   strncpy(newline, line, strlen(line) -2);
                   newline[strlen(line)-1] = '\0';
                   fputs(newline,fp2);
            fclose(fp2);
            if (compress == 1) {
                   fp2 = fopen("./temp","r");
                   uffman encode file(fp2,fp3);
                   fclose(fp3);
                   fp3 = fopen("./outputfile","r");
```

```
while (fgetc(fp3) != EOF)
                        temp++;
                  size_ = temp;
#ifdef DC DEBUG
                  printf("Compressing %d\n", size );
#endif /* DC DEBUG */
                  fclose(fp2);
            else
                  size_ = strlen(line);
#ifdef DC DEBUG
                 printf("Not compressing %d\n", size );
#endif /* DC DEBUG */
            }
            tsize = tsize + size ;
            fclose(fp3);
      } * /
#else
      if (!isClusterHead )
            if (compress == 1)
                  size = encode();
            else
                  size = sizeof(data);
      tsize = tsize + size_;
#endif /* HUFFMAN */
      if (++seqno_ < maxpkts_)</pre>
#ifdef DC DEBUG
           printf("Schedule packet...");
#endif /* DC DEBUG */
           return(t);
      }
      else
#ifdef DC DEBUG
           printf("Cant schedule packet..");
#endif /* DC DEBUG */
          return(-1);
      }
}
void DC Traffic::generate pkt(int size)
      size = size;
      nextPkttime = next interval(size);
#ifdef DC DEBUG
     printf("Packet of size %d generated...%ld\n", size , nextPkttime );
#endif /* DC DEBUG */
     timer .resched(nextPkttime);
void DC Traffic::timeout()
```

```
{
   if (! Running)
           return;
     /* send a packet */
     send(size);
     if (isClusterHead == 0) {
           /* figure out when to send the next one */
           nextPkttime = next interval(size);
           /* schedule it */
           if (nextPkttime_ > 0)
                 timer .resched(nextPkttime );
           else
                 running_ = 0;
           }
     else
      {
           if (nextPkttime < 0)</pre>
                 running = 0;
     }
}
datacompress.h – corresponding header file
/* -*-
           Mode:C++; c-basic-offset:8; tab-width:8; indent-tabs-mode:t -*-
*/
* Copyright (c) Priya Kasirajan 2008. All rights reserved.
* /
#ifndef NS DATACOMPRESS H
#define NS DATACOMPRESS H
#include "trafgen.h"
/* implement a source which takes packets, compresses them and generates
packets of
* the compressed size. It is parameterized by packet size and interval.
class DC Traffic : public TrafficGenerator {
public:
     DC Traffic();
     virtual double next interval(int&);
     int command(int argc, const char*const* argv);
     void generate pkt(int);
     int isClusterHead; /* is this node a clusterhead */
protected:
     void init();
     void start();
     void stop();
     void timeout();
                   /* send rate during burst (bps) */
     double rate ;
     double interval_; /* inter-packet time at burst rate */
   FILE *fp1; /* pointer to packet payload */
     int maxpkts_;
                      /* max number of packets */
```

```
int compress_;  /* is compression required */
int tsize; /* cumulative packet size */
};
#endif /* NS_DATACOMPRESS_H */
```

## Source code of hardware implementation:

routing\_mmcr.c - implementation of MMCR

```
/***************************
** Copyright (C) 2008-2009 Maciej Zawodniok, Priya Kasirajan. All rights
* *
** This file is part of the documentation of the UMR Mote Toolkit.
* *
** This file may be used under the terms of the GNU General Public
** License version 2.0 as published by the Free Software Foundation
** and appearing in the file LICENSE.GPL included in the packaging of
** this file.
** This file is provided AS IS with NO WARRANTY OF ANY KIND, INCLUDING THE
** WARRANTY OF DESIGN, MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE.
*******************************
#include "esnl pub.h"
#ifdef FEAT ENABLE MMCR
#include "routing.h"
#include "routing MMCR.h"
#include "network hardware ADFS.h"
#include "common.h"
#include "FEAT Xbee API/AT scripts.h"
#include "FEAT Queueing/Queuing.h"
#include "FEAT Networking/tsp common pub.h"
#include "Application.h"
#include "FEAT Networking/Phy layer.h"
#include "packet.h"
#include "FEAT Scheduler/sch basic pub.h"
#ifdef KEIL
     #ifdef ENABLE F1 MOTE TEST 01
          sbit YLED = P2 ^ 5;
     #else // ENABLE F1 MOTE TEST 01
          sbit YLED = P2 ^ 2;
     #endif // else/not _ENABLE_F1_MOTE_TEST_01_
#endif
#ifdef SDCC
     #ifdef ENABLE F1 MOTE TEST 01
          #define \overline{Y}LED P2 \overline{5}
     #else // ENABLE F1 MOTE TEST 01
          #define YLED P2 2
     #endif // else/not ENABLE F1 MOTE TEST 01
#endif // __SDCC__
```

```
uint8 t          mmcr pkt id = 0xFF;
unsigned char xdata mmcr request send TC = 0; // request for sending
Topology control
// Variables to manage )periodic) timeout
uint8 t mmcr hello timeout enabled = 0;
rtc_tick_t mmcr_hello_timeout_ = RTC OVERFLOW TIMER VALUE;
#define RTR MMCR HELLO PKT SIZE
                                  18
#define RTR MMCR TC PKT SIZE 14
#define MMCR BANDWIDTH FACTOR 1
#define MMCR SWITCH CHANNEL(c) (c>=MAX RF CHANNEL)?MIN RF CHANNEL:c+1
uint8 t mmcr state; // state of the routing agent (e.g. IDLE, routing in-
progress)
unsigned char xdata mmcr Hello countdown; // ON - periodically send
HELLO
int xdata mmcr route search BS; // address of the target node (BS)
//Counters for the Routing Energy Analysis
unsigned char xdata mmcr Hello counter;
unsigned char xdata mmcr ACK MMCR counter;
unsigned char xdata mmcr DAT MMCR counter;
uint16 t mmcr len sample;
struct ONEHOP
     char ID;
     bool MPR;
     char Link[10]; // point to index (ID) of the second hop node
     char Links;
     uint32 t LinkCost;
} ;
struct TWOHOP
{
     char ID;
     char Covered;
};
struct
     struct ONEHOP OneHop[10];
                TWOHOP TwoHop[10];
     struct
     int
                OneHopNodes;
                TwoHopNodes;
     int
} Neighborhood;
struct RENTRY
{
     char ID;
     unsigned int NextHopID;
     uint32 t NextHopCost;
```

```
unsigned int AltNextHopID;
     uint32 t AltNextHopCost;
};
struct
     char length;
     struct RENTRY rentry[10];
} rtable;
* routing init - performs initial setup of routing
 * /
void routing init MMCR()
     AODVcounter update = 1;
     mmcr Hello counter = 0;
     mmcr ACK MMCR counter = 0;
     mmcr DAT_MMCR_counter = 0;
     mmcr Hello countdown =0;
     mmcr state =MMCR STATE IDLE;
     mmcr pkt id = 0xFF;
     enableDataTx = 0;
     mmcr_request_send_HELLO_ = 1; // request for sending HELLO
     // Variables to manage (periodic) timeout
     mmcr_hello_timeout_ = RTC_OVERFLOW TIMER VALUE;
     init topology();
     mpr select();
     mmcr startRouteSearch(MY DEST);
#ifdef MZ TEMP TEST SCH HIO
     sch add loop(( sch loop func t )mmcr loop );
#endif // MZ TEMP TEST SCH HIO
}
* init topology - performs initial setup of topology
void init topology()
     int x, y;
     for (x=0; x<10; x++)
          Neighborhood.OneHop[x].ID=0;
          Neighborhood.OneHop[x].MPR=false;
          Neighborhood.OneHop[x].Links=0;
          Neighborhood.OneHop[x].LinkCost=0xFFFFFFF;
          for (y=0; y<10; y++)
               Neighborhood.OneHop[x].Link[y]=0;
```

```
Neighborhood.TwoHop[x].ID=0;
            Neighborhood.TwoHop[x].Covered=0;
      }
#ifdef NODE 1
      Neighborhood.OneHopNodes = 2;
      Neighborhood.TwoHopNodes = 1;
      Neighborhood.OneHop[0].ID = 0 \times 0 C;
     Neighborhood.OneHop[0].Links = 1;
     Neighborhood.OneHop[0].Link[0] = 0x0B;
     Neighborhood.OneHop[1].ID = 0x0D;
      Neighborhood.OneHop[1].Links = 1;
      Neighborhood.OneHop[1].Link[0] = 0x0B;
      Neighborhood. Two Hop [0] . ID = 0 \times 0 B;
#endif
#ifdef NODE 2
      Neighborhood.OneHopNodes = 2;
      Neighborhood.TwoHopNodes = 1;
     Neighborhood.OneHop[0].ID = 0 \times 0 C;
      Neighborhood.OneHop[0].Links = 1;
     Neighborhood.OneHop[0].Link[0] = 0x0A;
     Neighborhood.OneHop[1].ID = 0x0D;
     Neighborhood.OneHop[1].Links = 1;
      Neighborhood.OneHop[1].Link[0] = 0x0A;
      Neighborhood. Two Hop [0] . ID = 0 \times 0 A;
#endif
#ifdef NODE 3
      Neighborhood.OneHopNodes = 2;
      Neighborhood.TwoHopNodes = 0;
     Neighborhood.OneHop[0].ID = 0x0A;
      Neighborhood.OneHop[0].Links = 0;
      Neighborhood.OneHop[1].ID = 0x0B;
      Neighborhood.OneHop[1].Links = 0;
#endif
#ifdef NODE 4
      Neighborhood.OneHopNodes = 2;
      Neighborhood.TwoHopNodes = 0;
      Neighborhood.OneHop[0].ID = 0x0A;
      Neighborhood.OneHop[0].Links = 0;
      Neighborhood.OneHop[1].ID = 0x0B;
      Neighborhood.OneHop[1].Links = 0;
#endif
      /* init routing table */
      rtable.length = Neighborhood.TwoHopNodes;
      for (x = 0; x < rtable.length; x++)
            rtable.rentry[x].ID = Neighborhood.TwoHop[x].ID;
            rtable.rentry[x].NextHopID = 0xFFFF;
            rtable.rentry[x].NextHopCost = 0xFFFFFFFF;
            rtable.rentry[x].AltNextHopID = 0xFFFF;
            rtable.rentry[x].AltNextHopCost = 0xFFFFFFFF;
}
```

```
* mpr select - performs MPR selection
  * /
void mpr select()
      unsigned char mpr[10];
      int x = 0;
      int y = 0;
      int z = 0;
      unsigned char count[10];
      bool uncovered;
      int overcoverage;
      for (x=0; x<10; x++)
            count[x]=0;
            mpr[x]=0;
      }
      /** MPR selection **/
      // count coverage
      for(x=0;x<Neighborhood.OneHopNodes;x++)</pre>
            for(y=0;y<Neighborhood.OneHop[x].Links;y++)</pre>
                   for(z=0;z<Neighborhood.TwoHopNodes;z++)</pre>
                         if(Neighborhood.OneHop[x].Link[y] ==
Neighborhood.TwoHop[z].ID)
                                count[z]++;
                   }
      }
      // 1. select singular neighbors
      for(x=0;x<Neighborhood.OneHopNodes;x++)</pre>
            if(count[x] == 0)
                   Neighborhood.OneHop[x].MPR = true;
                   //TODO: search for single link
      Neighborhood. Two Hop [find two hop node (Neighborhood. One Hop [x]. Link [0])].
Covered ++;
      }
      // 2. selects as MPR neighbors with the largest count of uncovered
twohop nodes
      x=0;
      uncovered = true;
      while(x<Neighborhood.OneHopNodes /*&& uncovered*/)</pre>
             //TODO: sort high-low
            if(count[x]>0)
             {
```

```
if(~Neighborhood.TwoHop[find two hop node(Neighborhood.OneHop[x].Link[0
])].Covered)
                   {
                         Neighborhood.OneHop[x].MPR = true;
      Neighborhood. Two Hop [find two hop node (Neighborhood. One Hop [x]. Link [0])].
Covered ++;
            x++;
      // 3. remove redundant nodes
      for(x=0;x<Neighborhood.OneHopNodes;x++)</pre>
             overcoverage = 0;
             for(y=0;y<Neighborhood.OneHop[x].Links;y++)</pre>
                   //for(z=0;z<10;z++)
      if (Neighborhood. Two Hop [find two hop node (Neighborhood. One Hop [x]. Link [y]
) | .Covered > 1)
                                overcoverage ++;
                   }
            if(overcoverage == Neighborhood.OneHop[x].Links)
                   Neighborhood.OneHop[x].MPR = false;
                   //TODO: reduce Covered
      }
    y = 0;
      for(x=0;x<Neighborhood.OneHopNodes;x++)</pre>
             if(Neighborhood.OneHop[x].MPR)
                   mpr[y++] = Neighborhood.OneHop[x].ID;
      }
char find one hop node(char ID)
      int x;
      for (x=0; x<10; x++)
             if (Neighborhood.OneHop[x].ID == ID)
                   return x;
      return -1;
}
char find two hop node(char ID)
```

```
{
     int x;
     for (x=0; x<10; x++)
          if (Neighborhood. Two Hop [x]. ID == ID)
              return x;
    return -1;
}
^{\star} senddummy data - sends dummy data - test purpose
 */
unsigned int sendDATA MMCR()
     if(enableDataTx )
         mmcr len sample=80;
          tsp reserve packet ( mmcr len sample, &mmcr pkt id, MY DEST );
         tsp send from modules(mmcr pkt id);
         mmcr pkt id = 0xFF;
     return 1;
}
* startRouteSearch - starts the route discovery procedure
void mmcr startRouteSearch ( unsigned int dst )
     if ( ( 0 < mmcr Hello countdown  ) || ( MMCR STATE IDLE != mmcr state
) )
     {
               return;
     // #################################
     // start route Discovery toward BS
    mmcr_route_search_BS_ = dst;
     rtr MMCR Hello Phase();
    mmcr Hello countdown = MMCR DEFAULT ROUTE SEARCH ON REPETITIONS;
    mmcr state = MMCR STATE ROUTE DISCOVERY;
    mmcr set tx timeout ( 10 );
```

```
* rtr MMCR Hello Phase() - handles the routing phase of transmitting HELLO
pkts
 * /
void rtr MMCR Hello Phase()
    int a;
    if ( ROUTING PROTOCOL MMCR == my protocol )
         // Send HELLO routing packet
         if ( 1 == mmcr request send HELLO )
              uint8 t packet[RTR MMCR HELLO PKT SIZE];
              uint16 t len = mmcr sendHELLO ( (hpkt mmcr t xdata*)packet
);
#ifdef ENABLE XBEE_API_
              a = api send packet16( packet, len, 0xFFFF);
#endif // ENABLE XBEE API
              if ( AODVcounter update )
                  mmcr Hello counter++;
              my energy = my energy - HELLO LENGTH;
              mmcr request send HELLO = 0;
    }
}
* mmcr set tx timeout(??) - set one-time timeout for MMCR hello/tc packets
 * /
void mmcr set tx timeout ( uint16 t ms )
    mmcr hello timeout = rtc get ticks() + ms;
    mmcr hello timeout enabled = 1;
* mmcr sendHELLO - broadcasts a HELLO packet
unsigned int mmcr sendHELLO ( hpkt mmcr t xdata*hello pkt )
    int len = PKT HEADER LENGTH; // size of the packet header (start + flag
+ dst(2) + src(2) + len
    hpkt mmcr t xdata *hpkt = ( ( hpkt mmcr t xdata * ) hello pkt);
```

```
hpkt->start = 0x55;
     hpkt->mac src = MY ADDR;
     // FILL END-TO-END ADDRESSING
     hpkt->dst id = 0xFF;
     hpkt->src id = MY ADDR;
     hpkt->length=HELLO LENGTH MMCR - PKT HEADER LENGTH;;
     // FILL THE MMCR field
    hpkt->energy=my energy ;
     // FILL the CHECKSUM
     // TODO: calculate checksum
     hpkt->crc = 65;
    return HELLO LENGTH MMCR;
* mmcr_recvHELLO - handles a received HELLO packet
 * should identify if this node is potential relay node
 * if yes, then send response (ACK?)
 */
unsigned int mmcr recvHELLO ( hpkt mmcr t xdata *hp )
    unsigned long int link factor, i=0;
     uint32 t energy = hp->energy;
     long int delay = TYPICAL DELAY FOR LINK;
     if (ENERGY MAX VALUE < energy)</pre>
          energy = ENERGY MAX VALUE;
     link factor = ( energy * MMCR BANDWIDTH FACTOR * DELAY SCALING ) /
(delay);
     for (i = 0; i < Neighborhood.OneHopNodes; i++)</pre>
          if (hp->mac src == Neighborhood.OneHop[i].ID)
               Neighborhood.OneHop[i].LinkCost= link factor;
     }
    mmcr_sendACK( (hpkt_mmcr_t xdata *) hp);
     return 1;
/**
 * mmcr sendACK - handles a sending of ACK packet
 * collects all info for packet
 * prepares timestamp
```

```
* /
unsigned int mmcr sendACK ( hpkt mmcr t xdata*hp)
     uint8 t i;
     uint8 t xdata pkt[ACK LENGTH MMCR];
     long int delay = TYPICAL DELAY FOR LINK;
     apkt t mmcr *ap = ( (apkt t mmcr *) pkt );
     uint16 t *pkt neighbor addr = (uint16 t *)&(ap->neighbor addr first);
     uint32 t *pkt neighbor cost = NULL;
     uint8 t *pkt crc;
     ap->start = START BYTE;
     ap->flags = FLAG ACK MMCR;
     ap->mac dst = hp->mac src;
                                 //MAC DST
     ap->mac src = MY ADDR;
                                 // MAC SRC
     ap->length = 15 + (Neighborhood.OneHopNodes*6);
     ap->dst id = hp->src id;
     ap->src id = MY ADDR;
     ap->energy = my energy ;
     ap->neighbor count=Neighborhood.OneHopNodes;
     for(i=0;i<Neighborhood.OneHopNodes;i++)</pre>
           *pkt neighbor addr = Neighborhood.OneHop[i].ID;
           pkt neighbor addr++;
           pkt_neighbor_cost = (uint32_t *)pkt_neighbor_addr;
           *pkt neighbor cost = Neighborhood.OneHop[i].LinkCost;
           pkt neighbor addr +=2;
     }
     pkt crc = (uint8 t *)pkt neighbor addr;
     *pkt crc = STOP BYTE;
     if ( AODVcounter update )
     {
           mmcr ACK MMCR counter++;
     my energy = my energy - ap->length;
#ifdef ENABLE XBEE API
     api send packet16 ( (uint8 t xdata*) ap, ap->length, ap->mac dst );
     return ap;
#else // _ENABLE_XBEE_API_
     return 0;
#endif // else ENABLE XBEE API
* mmcr revcACK - handles a received ACK packet
```

```
* should identify if this node is potential relay node
  * if yes, then send response (ACK?)
unsigned int mmcr recvACK (apkt_t_mmcr *apkt )
      apkt t mmcr xdata *ap = (( apkt t mmcr xdata * ) apkt);
      long int delay = TYPICAL DELAY FOR LINK;
     unsigned long int link factor;
     uint32 t energy = ap->energy;
     uint16_t *pkt_neighbor_addr = (uint16_t *)&(ap->neighbor_addr_first);
      uint32 t *pkt neighbor cost = NULL;
      int i,j,k;
      if (ENERGY MAX VALUE < energy)
            energy = ENERGY MAX VALUE;
      link factor = ( energy * MMCR BANDWIDTH FACTOR * DELAY SCALING ) /
(delay);
      for (i = 0; i < Neighborhood.OneHopNodes; i++)</pre>
            if (ap->mac src == Neighborhood.OneHop[i].ID)
                 Neighborhood.OneHop[i].LinkCost= link factor;
                  for (k = 0; k < ap->neighbor count; k++)
                        for (j = 0; j < Neighborhood.OneHop[i].Links; j++)</pre>
                              if (*pkt neighbor addr ==
Neighborhood.OneHop[i].Link[j])
                                    pkt neighbor cost = (uint32 t
*) (pkt neighbor addr+1);
                                   mmcr set route (*pkt neighbor addr, ap-
>mac src, link factor + *pkt neighbor cost);
                       pkt neighbor addr +=3;
                  }
      }
      return 1;
* mmcr sendTC - send channel switch message
  */
unsigned int mmcr_sendTC ( tpkt_t_mmcr xdata*tpkt )
      int len = PKT HEADER LENGTH; // size of the packet header (start + flag
+ dst(2) + src(2) + len
      //tpkt t mmcr xdata *tpkt = ( ( tpkt t mmcr xdata * ) tc pkt);
      tpkt->start = 0xAA;
      tpkt->flags = FLAG TC MMCR; // TC packet
```

```
tpkt->mac dst = MAC BROADCAST; // Broadcast
     tpkt->mac src = MY ADDR;
     tpkt->length = RTR MMCR TC PKT SIZE - PKT HEADER LENGTH; // of the
packet'd data
     // FILL END-TO-END ADDRESSING
     tpkt->dst id = 0xFF;
     tpkt->src id = MY ADDR;
     tpkt->energy=my energy ;
     tpkt->channel=MMCR SWITCH CHANNEL(DEFAULT RF CHANNEL);
     // FILL the CHECKSUM
     // TODO: calculate checksum
     tpkt->crc = 65;
     return RTR MMCR TC PKT SIZE;
}
* mmcr recvTC - switch channel
 * /
unsigned int mmcr recvTC ( tpkt t mmcr *tp )
     phy_set_RF_channel(tp->channel);
     return \overline{1};
* mmcr loop() - executes main loop block (BUT DOES NOT LOOP ITSELF!!!)
void mmcr loop( void )
     if (1 == mmcr hello timeout enabled)
          if ( mmcr hello timeout < rtc get ticks())</pre>
                mmcr hello timeout enabled = 0;
                mmcr_hello_timeout_ = RTC_OVERFLOW_TIMER_VALUE;
                mmcr hello timeout();
     }
* mmcr hello timeout(??) - runs the procedure of periodic sending of Hello
packets
 * /
void mmcr hello timeout()
     // Timeout reached -> is there a route found? or should I retransmit
HELLO?
     unsigned int mac addr = mmcr neighbor analyse (MY DEST);
```

```
if ( 0 < mmcr Hello countdown )
                      mmcr Hello countdown --; // count down HELLO
retransmissions
           // number of retransmission not reached -> resent HELLO
           YLED = \sim YLED;
           mmcr request send HELLO = 1;
           mmcr request send TC = 1;
           mmcr set tx timeout (ROUTE SEARCH HELLO INTERVAL);
           rtr MMCR Hello Phase();
           YLED = \sim YLED;
     }
     else
           // all Hello messages has been without response -> STOP
           mmcr Hello countdown =0;
           mmcr request send HELLO = 0; //1;
           mmcr request send TC = 0;
           mmcr state =MMCR STATE IDLE;
#ifdef NODE 1 //to handle any unprocessed acks
           mmcr set route(0x0B,0x0C,0xCCCC);
           mmcr set route(0x0B,0x0D,0xCCCD);
#endif
           enableDataTx = 1;
     }
* mmcr set route - adds or updates a route to "dst_id"
int mmcr set route ( unsigned int dst id, unsigned int next hop, uint32 t
metric )
{
     int i;
     mpr select();
     for (i = 0; i < rtable.length; i++)</pre>
           if (rtable.rentry[i].ID == dst id)
                 // rtable record spotted -> set next hop
                 if (rtable.rentry[i].NextHopID == 0xFFFF)
                 {
                      rtable.rentry[i].NextHopID = next hop;
                      rtable.rentry[i].NextHopCost = metric;
                      return 1; //successfully added route
                 else if (rtable.rentry[i].NextHopID == next_hop)
                      rtable.rentry[i].NextHopCost = metric;
                      return 1; //successfully updated route
                 else if (rtable.rentry[i].AltNextHopID == 0xFFFF)
                      rtable.rentry[i].AltNextHopID = next hop;
                      rtable.rentry[i].AltNextHopCost = metric;
```

```
return 1; //successfully added route
                else if (rtable.rentry[i].AltNextHopID == next hop)
                {
                      rtable.rentry[i].AltNextHopCost = metric;
                      return 1; //successfully updated route
                }
           }
     return 0; // two hop id not found!!!
* neighbor analyse - performs analysis of the neighbor table
           and selects the best node
 * /
unsigned int mmcr neighbor analyse (unsigned int dst id)
     unsigned int i, result = 0xFFFF; // by default FFFF indicating lack
of route
     /* check if dst id is a one hop neighbor */
     for (i = 0; i < Neighborhood.OneHopNodes; i++)</pre>
           if (Neighborhood.OneHop[i].ID == dst id)
                result = Neighborhood.OneHop[i].ID;
     /* check if dst id is a two hop neighbor */
     for (i = 0; i < rtable.length; i++)
           if (rtable.rentry[i].ID == dst id)
                if ((rtable.rentry[i].NextHopID == 0xFFFF) &&
(rtable.rentry[i].AltNextHopID == 0xFFFF))
                      result = 0xFFFF;
                else if ((rtable.rentry[i].NextHopCost <=</pre>
rtable.rentry[i].AltNextHopCost) && (rtable.rentry[i].NextHopID != 0xFFFF))
                     result = rtable.rentry[i].NextHopID;
                else if (rtable.rentry[i].AltNextHopID != 0xFFFF)
                      result = rtable.rentry[i].AltNextHopID;
     return result;
}
* sendDATA - handles a sending of DATA packet
 * 1) check if buffer ready then passes packet
 * 2) else temporarly stores
char MMCR send DATA base (unsigned int base)
{
     unsigned int mac d;
     pkt t * xdata pkt = ( pkt t* ) ( & ( buffer0[ base] ) );
//&(QBUFF ACCESS(base,0));
```

```
if (enableDataTx )
     //if(1)
          mac d = mmcr neighbor analyse ( pkt->dst id );
          if (0xFFFF==mac d)
                enableDataTx_ = 0;
                return 0;
          pkt->mac_dst = mac_d;
          pkt->mac src = MY ADDR;
          // send the packet
          if ( AODVcounter update )
               mmcr DAT MMCR counter++;
     }
     else
          APPEND LOG ( NODE ID STR, NODE ID STR LEN );
          APPEND LOG ( "DROP DATA\r", 10 );
          // drop the packet
          return 0;
     return 1;
* recvAcceptData - handles a received ACCEPT DATA packet from BS
 * 1) enables data transmission
 * 2)
 */
unsigned int mmcr recvAcceptData ( char *pkt )
{
     pkt t xdata *p = ( ( pkt t mmcr xdata * ) pkt);
     // analyse ACK packet
     if ( 0xFFFF == p->mac dst )
          // enable DATA transmission
          enableDataTx = 1;
          APPEND_LOG ( NODE_ID_STR, NODE_ID_STR_LEN );
          APPEND LOG ( "DATA - OK\r\r\r", 1\overline{2} );
          return 1;
     // wrong trasmitter
     return 0;
}
```