University of California
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Decentralized Information Processing in Wireless Peer-to-Peer Networks

A dissertation submitted in partial satisfaction
of the requirements for the degree
Doctor of Philosophy in Electrical Engineering

by

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2002

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University of California, Los Angeles

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To my parents.

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Vita

Publications

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Abstract of the Dissertation

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Professor Gregory J. Pottie, Chair

Decentralized information processing systems, such as wireless sensor networks, facilitate the acquisition, processing and dissemination of *information*. With the phenomenal growth in the development of digital hardware and wireless information technologies, the efficient handling of data in such systems has become of prime concern. This thesis deals with several inter-related problems relating to the processing of information in wireless peer-to-peer networks. It presents a collection of analysis, algorithmic techniques and results that are designed to optimize performance in such decentralized systems.

Sensors architectures are analyzed as the starting point of the study, and a unified, probabilistic, information processing approach to data fusion is presented for heterogeneous, multi-sensor networks. A likelihood metric aggregation architecture, based on the Bayesian approach, is highlighted as the central unifying technique for decentralized organization and interpretation of information. The need for fundamental performance limits for decentralized networks is then discussed, and some bounds are derived. In particular, the rate distortion region for a sensor fusion network with n-helper nodes in a Gaussian setting is described.

The asymptotic delay order for data packets in a random wireless network is also derived.

Next, some practical issues dealing with the administration of wireless resources in heterogeneous, multi-tiered ad hoc networks are discussed. A hybrid, gateway-based architecture and trajectory control algorithms for enabling range extended connectivity are presented. The dependability for such networks are then analyzed, and a distributed reliability algorithm is formulated that can be applied to optimize the dependability of decentralized, dynamic peer-to-peer networks. Specific fault tolerance algorithms for the gateway-based architecture are also devised.

Finally, bandwidth efficient techniques are analyzed that serve to extract the maximum spectral efficiency in point-to-point communications. A rate adaptive transceiver is designed that combines multi-input, multi-out (MIMO) antennas, orthogonal frequency domain multiplexing (OFDM) and low density parity check (LDPC) channel codes. Channel estimation and novel signal separation techniques are derived.

The overall goal is thus to develop a set of cross layer techniques, from the physical to the network layer, that can be applied to quantify some of the basic performance limits for distributed information processing systems.

CHAPTER 1

Introduction

1.1 Information Processing in Wireless Networks

The last few decades have witnessed a phenomenal growth in the research and development of solid-state, digital communications, computing and signal processing technologies. A beneficial aspect of all this concentrated activity has been the symbiotic convergence and amalgamation of various hardware, software and 'concept-driven' technologies. This approach, for example, has yielded most of the *information processing* devices that we take for granted today. Ostensibly, the purpose of all this activity has been to improve the human condition; but from a scientific and engineering point of view, the common unifying purpose that has shaped and guided the general research direction in these disciplines has been the desire to acquire, manipulate, and disseminate *information*. In the same vein, this thesis focuses on some inter-related problems of *information processing*.

In particular, the ability to electronically network together what were previously isolated islands of information sources and sinks has revolutionized many research disciplines. One such effort has been in the cooperative sensing and control of the environment (or more generally, *states of nature*). This can refer to measurements of physical parameters (e.g. temperature, humidity, etc.) or estimates of operational conditions (network loads, throughput, etc.), among other things. Previously, these activities were performed by isolated *sensors*, re-

quiring human supervision and control. However, with the advent of powerful hardware platforms and networking technologies, the possibility and advantages of *distributed sensing* has been recognized. Inspite of the advances in sensor technologies and the many computational methods and algorithms aimed at extracting information from a given sensor/actuator, the fact remains that no single sensor is capable of obtaining a required state information reliably, at all times, in different and sometimes dynamic environments. Furthermore, it has been established from the theory of distributed detection that higher reliability and lower probability of detection error can be achieved when observation data from multiple, distributed sources is intelligently fused in a decision making algorithm, rather than using a single observation data set. The main advantages of using networked, multi-system platforms can thus be summarized as follows.

- Reliability and greater accuracy through redundancy, by using multiple sensors to measure the same or overlapping quantities, and exploiting the fact that the signal relating to the observed quantity is correlated whereas the uncertainty associated with each sensor is uncorrelated.
- Diversity, where physical sensor diversity uses different sensor technologies together, and spatial diversity offers differing viewpoints of the sensor environment.
- Scalability, where decisions can be made over a larger state space of observations by having distributed, efficient local computations and hierarchical data fusion, thereby reducing the complexity of the command and control center's operations.

These issues are especially notable in the context of heterogeneous sensor/-actuator nodes. These devices may be networked or as part of larger mobile

platforms, forming an *ad hoc network* of wireless integrated information processing devices. Their function is to engage in cooperative, distributed sensing, computation, and communications for decision and action. However, there are significant research and engineering issues that need to be addressed before such heterogeneous systems can be successfully deployed, and these problems are currently at the scientific frontiers of Information Technology.

Focus of the thesis:

In this thesis, the above mentioned issues are studied in the context of the following general problem: given a multiplicity of wireless sensors, and a sensing/processing objective, what is the optimum set of tasks to—

(i) efficiently extract as much information as possible about a sensed environment;

(ii) process the data locally, and/or intelligently fuse the aggregate data at distributed hierarchical levels, according to the sensing objectives;

(iii) cooperate to maintain connectivity and interact with command and control centers for communicating decision variables and instructions.

The assumption is made of a system of heterogeneous sensor (or general ad hoc) networks, working cooperatively for a particular sensing objective. This distributed information processing approach can overcome the shortcomings of the alternative approach of using a highly sophisticated, but single sensor for the same objective. But, as noted above, the effective deployment of such distributed processing systems introduces some significant design issues, most notably: scalability, networking and communication protocols, transmission channel and power constraints, reliability, among others. The approach taken in this thesis is to

study these issues from several different viewpoints: information theoretic; data fusion and reliability based; and from practical cooperative, rate adaptive, efficient digital communications considerations. The objective is to gain insight into novel paradigms for the data fusion and performance analysis of networked sensors; for evaluating the asymptotic rate and delay properties of such random networks; for the architectures necessary for ad hoc structures involving a heterogeneous mix of such sensors, their reliability and fault tolerance issues, etc., and finally we also look at bandwidth efficient techniques that can be used for such distributed information processing systems.

One of the unifying features that is common to these approaches is the analysis of factors that affect performance when scaling the number of nodes in a sensor system from a few (when combinatorial methods for system performance may be tractable) to many (when statistical methods are the only options). The goal is to thus to determine these intelligent unifying techniques, approaching the problem from these varying analysis viewpoints: to quantify, analyze and understand the answers to some of the basic architectural and performance limits questions for distributed sensing systems. We have attempted to envision this philosophy as depicted in Figure 1.1.

1.2 Thesis Topics and Contributions

In this thesis, the following five topics relating to information processing in wireless peer-to-peer networks are studied, in the order listed. A summary of the contributions that have been made in each topic is provided below, and the detailed accounts are presented as Chapters 2 through 6 of this thesis.

1. **Data Fusion:** This is the problem of combining diverse and sometimes

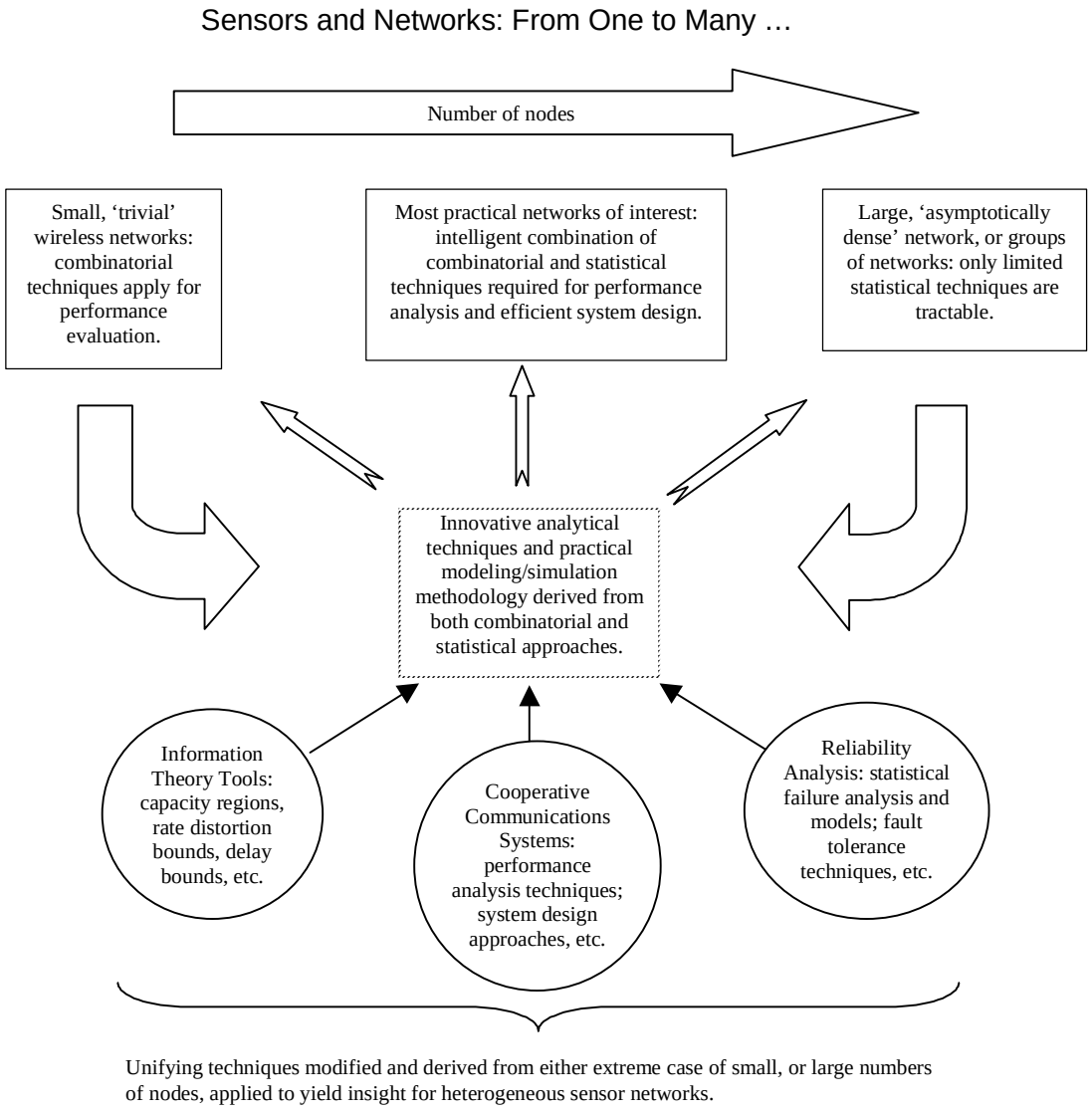

Figure 1.1: Unifying information processing techniques that yield insight for wireless sensing networks at different levels of granularity.

conflicting information provided by sensors in a multi-sensor system, in a consistent and coherent manner. The objective is to infer the relevant states of the system that is being observed or activity being performed. Using probabilistic and decision theoretic principles, this thesis presents a unified treatment of information processing, within the framework of the Bayesian Paradigm, as they relate to the issues of: data representation, fusion, and transmission in decentralized sensor networks. It is shown in Chapter 2 of this thesis that the core elements of decentralized, heterogeneous data fusion can be quantitatively formulated and analyzed using a generalized, modular information processing framework that is amenable to practical implementation.

2. **Information Theoretic Bounds:** This relates to results that delineate the boundaries of the *amount* of information processing that can be done with multi-terminal networks. It is evident that some fundamental limits are required to assess the optimality of any system design with regard to the "best design", given the available resources for a particular application. Unfortunately, comprehensive general information theories do not yet exist for decentralized, multi-terminal networks. So, in Chapter 3 of the thesis, some simplified asymptotic cases are studied. In particular, wireless sensor networks and the data fusion process are modeled from a classical information theoretic viewpoint, and the rate distortion region for correlated sources is derived for a special case. Most practical sensor and ad hoc networks are also too large for combinatorial or queueing theory analysis to determine fundamental properties such as end-to-end throughput and delay. In this regard, also in this chapter, a simplified scenario is considered, and based on recent results, the asymptotic delay order that is experienced

by an 'average' data packet in the network is derived.

3. **Resource Administration—the NGI framework:** This relates to the task of optimally configuring, coordinating and utilizing available sensor and ad hoc resources, often in a dynamic, adaptive environment. The objective is to ensure efficient[1] use of the wireless resources for the task at hand. In Chapter 4 of this thesis, this problem is approached in the context of a hybrid, multi-tiered network that is envisioned to be the architecture for the Next Generation Internet.[2] In particular, the role of mobile *gateways* is recognized as crucial in enabling cross-platform connectivity and managing fusion/relay processes (as in Chapter 2), and the resulting architectural implications are critically examined. Protocols are developed for the optimal trajectory control of gateways, and for load-balancing, etc.

4. **Dependability of Heterogeneous Networks:** Heterogeneous networks often rely on special network architectural arrangements for information processing (e.g. hierarchical or centralized nodes, defined backbone protocols, etc.). However, this also causes such networks to suffer a greater vulnerability due to faults and failures among its critical nodes. The focus of Chapter 5 of the thesis is to use graph theoretic techniques to analyze the dependability of heterogeneous wireless ad hoc networks. This approach is then used to yield insight into how to improve network protocols that can lead to more dependable systems. Detailed protocols are developed to optimize moderately sized networks against node and link failures, and several algorithms are developed to handle the specific case of gateways in the NGI

[1] *Efficiency*, in this context, is very general and can refer to power, bandwidth, overhead, throughput, or a variety of other performance metrics, depending upon the particular application.

[2] DARPA NGI project, http://www.darpa.mil/ipto/research/ngi.

context.

5. **Bandwidth efficient communications using MIMO-OFDM-LDPC transceivers:** With the exponential rise in the use of wireless systems, it has become increasingly important to extract the maximum diversity from the time, space and frequency dimensions in which radio frequency devices operate. The final chapter of the thesis (Chapter 6) deals with this physical layer issue for hybrid wireless networks, by proposing a novel approach that maximizes the raw spectral efficiency of transceivers. This is accomplished by using a combination of three recent developments in digital communication theory to form a space-time coded transceiver that can be adaptively optimized. These technologies are: multi-input, multi-output antenna technology (MIMO), orthogonal frequency domain multiplexing (OFDM), and the powerful low density parity check channel codes (LDPC). In this thesis, in particular, novel signal separation and channel estimation, as well as adaptive modulation schemes are suggested.

The research effort that is the subject matter of this thesis thus deals with several inter-related problems relating to the processing of information in wireless networks, and presents a collection of analysis, algorithmic techniques, and results that are designed to optimize performance in hybrid peer-to-peer networks. The detailed descriptions follow.

CHAPTER 2

Distributed Data Fusion in Sensor Networks: An Information Processing Approach

2.1 Introduction

Sensors, in the context of this thesis, refer to physical devices that exploit physical phenomena to measure quantities. A sensor can be defined to be any device that provides a quantifiable set of outputs in response to a specific set of inputs. Usually, the inputs are environmental or physical parameters of interest in natural or artificial systems, and the outputs are measurable attributes of those parameters. So, for example, a temperature sensor registers the temperature, a gas pressure sensor senses pressure values, and so on. Sensors can also be software algorithms, e.g. subroutines measuring the data load through routers, diagnostic subroutines monitoring the status of devices, etc.[1]

Sensors are expected to provide information about the state of nature. A particular sensor device is considered appropriate for a sensing task when a relationship or mapping exists between the measured quantity and the state of nature. The end goal of the sensing task is to acquire a description of the external world, predicated upon which can be a series of *actions*. For example, velocity sensors in an automobile report the speed of the vehicle, which the oper-

[1]Natural biological sensing systems, such as our eyes, ears, etc., and other types of sensing methodologies based on behaviorial, psychological metrics, etc. are not considered in this thesis.

ator of the vehicle then adjudges to perform some action (accelerate, decelerate, or maintain constant speed).

Typically, in devices to date, sensors have been integrated as part of larger, more complex systems are have designed for specific purposes. So, whether the objective has been to land a person on the moon, or to enable a vehicle to travel from a particular point to another point in space, the sensors on board such devices provide a variety of state information that the system/operator then uses in the course of its actions. In this context, sensors can be thought of as *information gathering, processing and dissemination* entities, as diagrammed in Figure 2.1. The data pathways in the figure illustrate an abstraction of the

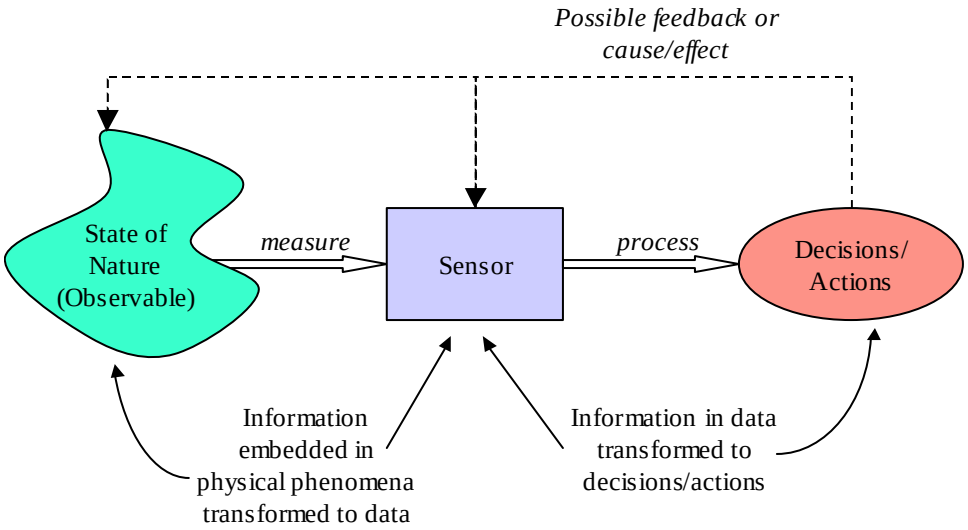

Figure 2.1: Information Processing in Sensors

flow of information in the system. Hence, sensor systems can fundamentally be studied from an Information Theory point of view [96]. From this perspective, all the issues of source coding, signal processing, rate distortion, fusion rules, etc. become relevant for sensor systems. In this chapter, we explore one of these issues in greater depth: the problem of data fusion in distributed sensor networks.

Rather than using single sensor platforms, modern sensor system hardware architectures often integrate multiple sensors that are physically disjoint or distributed in time or space, and that work cooperatively (an example of a *distributed-parameter* rather than a *lumped-parameter* system [10]). The extra dimensions and distributed nature of these systems add layers of complexity in the information gathering, processing and dissemination tasks, and is the central focus of this thesis.

The reasons for having a network of sensors, as opposed to a single sensor platform, essentially reduce to the advantages of *diversity*. Any practical sensing device has limitations on its sensing capabilities (e.g. resolution, bandwidth, efficiency, etc.). The primary limitation is that descriptions or physical models built on the data sensed by a device are, unavoidably, only approximations of the true state of nature. Such approximations are often made worse by incomplete knowledge and understanding of the environment that is being sensed and its interaction with the sensor. These uncertainties, coupled with the practical reality of occasional sensor failure greatly compromises reliability and reduces confidence in sensor measurements. Also, the spatial and physical limitations of sensor devices often means that only partial information can be provided by a single sensor.

As a result of these shortcomings, a single sensor has limited capability for resolving measurement ambiguities. And so, despite advances in sensor technologies and the many computational methods and algorithms aimed at extracting information from a given sensor, the fact remains that no single sensor is capable of obtaining a required state information reliably, at all times, in different and sometimes dynamic environments. This is especially true in the context of wireless sensor nodes on mobile platforms, forming a mobile *ad hoc* network of

sensors, which are expected to be the "eyes and ears" of a huge variety of future data processing devices.

It is thus plausible that efficient sensing systems must make use of a multiplicity of sensors, in a networked environment, in order to extract as much information as possible about a sensed environment. A network of sensors overcomes many of the shortcomings of a single sensor. The main advantages are:

- *redundancy* by using two or more sensors to measure the same or overlapping quantities and exploiting the fact that the signal relating to the observed quantity is correlated whereas the uncertainty associated with each sensor is uncorrelated;

- *diversity* and *complementarity*, where physical sensor diversity uses different sensor technologies together, and spatial diversity offers differing viewpoints of the sensor environment.

However, associated with these advantages, there are several problems that arise when using multiple sensors for any type of cooperative activity. These problems, also, are a result of the increased diversity and redundancy stemming from having multiple sensors, and in essence is a problem of efficient information management. The fundamental issues in using multiple sensors can be categorized into the following two broad areas [71]:

1. *Data Fusion:* This is the problem of combining diverse and sometimes conflicting information provided by sensors in a multi-sensor system, in a consistent and coherent manner. The objective is to infer the relevant states of the system that is being observed or activity being performed.

2. *Resource Administration:* This relates to the task of optimally configuring, coordinating and utilizing the available sensor resources, often in a dynamic,

adaptive environment. The objective is to ensure efficient[2] use of the sensor platform for the task at hand.

In comparison to lumped-parameter sensor systems (Figure 2.1), the issues mentioned above for multi-sensor systems can be diagrammed as shown in Figure 2.2.

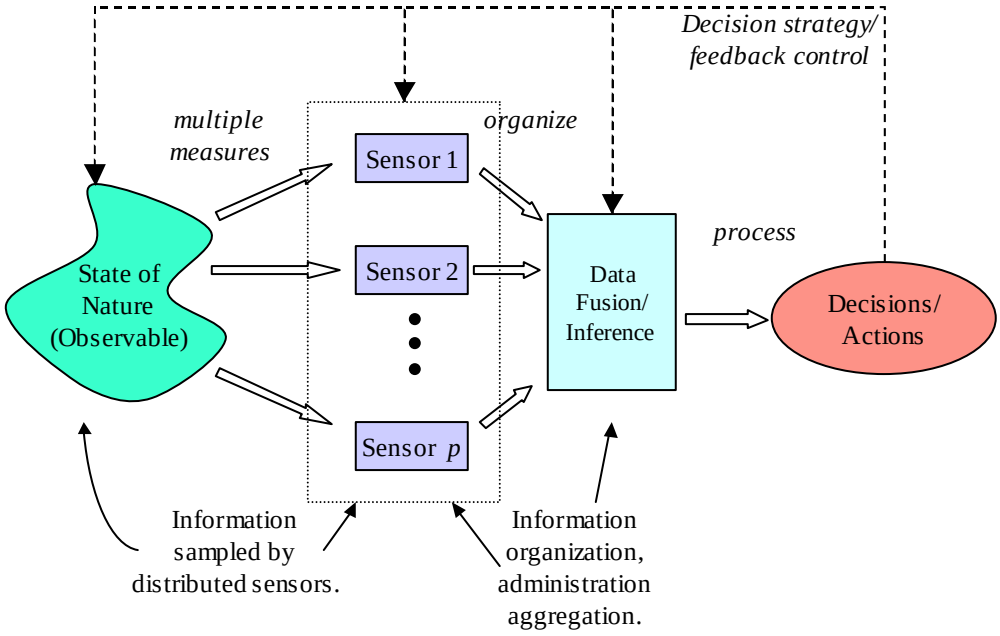

Figure 2.2: Information Processing in Distributed Sensors

2.1.1 Evolution Towards Multi-Sensor Systems

There has been a voluminous amount of research and development activity in the multi-disciplinary areas of data fusion and sensor systems. Most of the early research effort focused primarily on techniques motivated by specific applications,

[2] *Efficiency*, in this context, is very general and can refer to power, bandwidth, overhead, throughput, or a variety of other performance metrics, depending upon the particular application.

such as in vision systems, sonar, robotics platforms, etc. [47, 55, 62, 52]. Gradually, the inherent advantages of using multi-sensor systems were recognized [99, 6] and a need for a comprehensive theory of the associated problems of distributed, decentralized data fusion and multi-user information theory became apparent [93, 28, 17]. A fundamental paradigm shift occurred with the maturing of integrated circuit (IC) technologies in the last three decades. This allowed miniaturized, low-cost sensors (among a whole host of other electronic devices) to be mass produced and integrated with a wide variety of physical systems [24]. Parallel to this development was the equally phenomenal revolution in wireless communication technologies. A concentrated amount of effort was directed towards solving the general problems of wireless radio [48], and specific issues regarding wireless *ad hoc* or *peer-to-peer* networking [32, 79, 86]. Subsequently, it was only natural to combine these two disciplines—sensors and networking—to develop a new generation of distributed sensing devices that can work cooperatively to exploit diversity. This has led to the birth of the Wireless Integrated Networked Systems (WINS) of sensors, among other ad hoc platforms [72, 73], and has fuelled the associated research efforts over the last two decades in nano-technology and micro-electro-mechanical (MEMS) systems [76].

Various researchers have attempted to develop practical systems that partly address the aforementioned issues of efficient networking and data fusion for sensors in the context of specific applications. However, the general problem of efficient sensor administration for data fusion has not been comprehensively addressed for heterogeneous sensors, possibly configured as an ad hoc or peer-to-peer wireless network, in a mobile environment. In this thesis, in this and the following chapters, these issues are addressed in an integrated framework, based on information theoretic and system optimization principles. The objective is to determine platform-independent guidelines and design philosophies that can be

used irrespective of the underlying application or hardware/software architecture.

Sensor Fusion Research:

For sensor technology in general, the key research thrust to date has been in data fusion methodologies. As mentioned in Section 2.1, data fusion is the process by which data from a multitude of sensor is used to yield an optimal estimate of a specified state vector pertaining to the observed system [96], whereas sensor administration is the design of communication and control mechanisms for the efficient use of distributed sensors, with regards to power, performance, reliability, etc. The main issues in sensor data fusion and sensor administration have mostly been addressed separately, sometimes based on well-founded theories and sometimes in an ad hoc manner and in the context of specific systems and architectures. The research effort in sensor administration, in particular, has been addressed primarily in the context of wireless networking, and not necessarily in conjunction with the unique constraints imposed by data fusion methodologies.

To begin with, sensor models have been aimed at interpretation of measurements. Such an approach to sensor modeling is exemplified in the models presented by Kuc and Siegel [47], among others. Probability theory, and in particular, a Bayesian treatment of data fusion emerged as a simple yet powerful technique [96], and is arguably the most widely used method for describing uncertainty in a way that abstracts from a sensor's physical and operational details. Such quantitative methods have been used by researchers to evaluate and model uncertainty in vision sensing, for example. Qualitative methods have also been used to describe sensors, for example by Flynn [21] for sonar and infra-red applications. Much work has also been done in developing methods for intelligently combining information from different sensors. The basic approach has been to pool the information using what are essentially "weighted averaging"

techniques of varying degrees of complexity. For example Berger [6] discusses a majority voting technique based on a probabilistic representation of information. Non-probabilistic methods [29] used inferential techniques, for example for multi-sensor target identification. Inferring the state of nature given a probabilistic representation is, in general, a well understood problem in classical estimation. Representative methods are Bayesian estimation, Least Squares estimation, Kalman Filtering, and its various derivatives, etc. We anticipate that in systems of the future, the question of what techniques to use for data aggregation will be less pertinent than the question of how to use these techniques in a distributed fashion, which has not been addressed to date in a systematic fashion, except for some specific physical layer cases [97].

Sensor Administration Research:

In the area of sensor network administration, protocol development and management have mostly been addressed using application specific descriptive techniques for specialized systems [99]. Much of the work has been in the area of tracking radar systems, and robotics where the approach has been to develop models of the sensor behavior and performance, and then manage the sensor data transfer on that basis. This approach is facilitated by the centralized or hierarchical nature of these systems (please see Section 2.4 for further discussions on sensor network architectures). A large proportion of sensor allocation schemes are based on determining cost functions and performance trade-offs a priori [5], e.g. in using cost-benefit assignment matrices to allocate sensors to targets, or using Boolean matrices which defines sensor-target assignments based on sensor availability and capacity. Expert system approaches have also been used, as well as normative or decision-theoretic techniques. However, optimal sensor administration in this way has been shown by Tsitsiklis [93] to be very hard in the

general framework of distributed sensors, and practical schemes use a mixture of heuristic techniques (for example in data fusion systems involving wired sensors in combat aircrafts). Only recently have the general networking issues for wireless ad hoc networks been addressed (Sohrabi, Singh [87, 83]), where the main problems of self-organization, bootstrap, route discovery etc., have been identified. Application specific studies, e.g. in the context of antenna arrays (Yao, [103]) have also discussed these issued. However, few general fusion rules or data aggregation models for networked sensors have been proposed, with little analytical or quantitative emphasis. Most of these studies do not analyze in detail the issues regarding the network-global impact of administration decisions, such as choice of fusion nodes, path/tree selections, data fusion methodology, or physical layer signalling details.

Fusion Architectures:

With regards to implementable sensor fusion architectures, current systems are based on traditional centralized schemes, utilizing a central processor responsible for implementing data fusion, or at best a hierarchical system for relieving computation burdens at the central processor. But in the context of wireless systems, these schemes are not satisfactory because of the control and coordination signaling overhead required. Also, these hierarchical architectures are vulnerable to processor failure, computation bottlenecks and inflexibility. We believe that to overcome these shortcomings, the recent trend towards autonomous systems such as wireless sensor nodes (Pottie [73]) capable of creating an infrastructure in an automated fashion is a feasible approach. These systems offer a number of advantages: modularity by required the sensing and data fusion to take place at the local nodes, at the lowest possible hierarchical level (which satisfies the requirements of the various detection algorithms mentioned earlier); scalability

and flexibility, since the functionality is localized in the sensor and scaling the system is simply a matter of designing robust network protocols for the admission and removal of additional sensor nodes; and survivability and fault tolerance, due to the absence of a central processor, so the loss of nodes leads to a graceful degradation in performance.

However, as yet, there is scant analysis of general (application-independent) data fusion algorithms for such systems that operate in a wireless, distributed configuration, with local and global fusion operations in parallel. This thesis presents such an approach in the context of *Information Processing*, which can be considered as an information-theoretic approach to sensor administration data fusion.

2.1.2 An Information Processing Approach to Sensor Networks

It has been mentioned earlier that multi-sensor systems are basically information gathering, analyzing and transmitting systems. The information being handled almost always relates to a state of nature, and consequently, it is assumed to be unknown prior to observation or estimation. Thus, the model of the information flow shown in Figure 2.2 can be considered as a probabilistic model, and hence can be quantified using the principles of Information Theory [14, 27]. Furthermore, the process of data detection and processing that occurs within the sensors and fusion node(s) can be considered as elements of classical Statistical Decision Theory [70]. Using the mature techniques that these disciplines offer, a probabilistic information processing relation can then be quantified for sensor networks, and analyzed within the framework of the well-known Bayesian Paradigm [78]. Using this approach, this thesis presents a unified treatment of probabilistic information processing as they relate to the issues of data representation, fusion,

and transmission in decentralized sensor networks.

In particular, for the administration of multi-sensor systems, the autonomous nature of individual sensor nodes and the presence or absence of a central processor raises problems such as:

- consistency and consensus among decision-makers
- group vs. individual optimality for decisions
- data and network synchronization for coherent processing of information
- physical layer issues.

It is shown in this thesis that the core elements of these problems can be quantitatively formulated and analyzed using the information processing framework mentioned above. Performance issues can then be studied theoretically, and system level optimizations can be carried out efficiently. **References to the appropriate sections in the chapters here....**

Using this approach, this thesis sub-divides the issues into the following sub-problems.

1. Determination of appropriate information processing techniques, models and metrics for fusion and sensor administration.
2. Representation of the sensors process, data fusion, and administration methodologies using the appropriate probabilistic models.
3. Analysis of the measurable aspects of the information flow in the sensor architecture using the defined models and metrics.
4. Design of optimum data fusion algorithms and architectures for optimum inference in multi-sensor systems.

5. Design, implementation and test of associated networking and physical layer algorithms and architectures for the models determined in (4).

The subsequent sections in this thesis address these issues, beginning with a Bayesian scheme for generalizing the data fusion problem.

2.2 A Bayesian Scheme for Decentralized Data Fusion

Sensors provide an estimate of nature, and thus can be viewed as sources of information. In a multi-sensor system, several such information sources are available, so it is possible to implement different strategies for combining the information from multiple sources. Two issues are of immediate interest: **(i)** the nature of the information being generated the sensors, and **(ii)** the method of combining the information from disparate sources. We consider the first issue first.

2.2.1 Sensor Data Model for Single Sensors

It is a fact of nature that any observation or measurement by any sensor is always uncertain to a degree determined by the *precision* of the sensor. This uncertainty, or measurement *noise*, requires us to treat the data generated by a sensor probabilistically. We therefore adopt the notation and definitions of probability theory to determine an appropriate model for sensor data [25].

Definiton 2.1. A *state vector* at time instant t, is a representation of the *state of nature* of a process of interest, and can be expressed as a vector $\boldsymbol{x}(t)$ in a measurable, finite-dimensional vector space, Ω, over a discrete or continuous field, $\mathcal{F}$:

$$\boldsymbol{x}(t) = \begin{pmatrix} x_1 \\ x_2 \\ \vdots \\ x_n \end{pmatrix} \in \Omega \tag{2.1}$$

In Definition 2.1, the state vector is arbitrarily assumed to be n-dimensional. For example, a particular state of nature of interest can be the three dimensional position vectors of an airplane. The state space may continuous (such as for the position vectors of an airplane) or discrete (e.g. the on or off states of a switch). For generality, unless otherwise indicated, state vectors are hereafter assumed to be defined over the continuous, real number field.

$$\Omega \subseteq \mathbb{R}^n \tag{2.2}$$

Definiton 2.2. A *measurement vector* at time instant t is the information generated by a single sensor (in response to an observation of nature), and can be represented by an m-dimensional vector, $\boldsymbol{z}(t)$ from a measurement vector space Ψ.

$$\boldsymbol{z}(t) = \begin{pmatrix} z_1 \\ z_2 \\ \vdots \\ z_m \end{pmatrix} \in \Psi \subseteq \mathbb{R}^m \tag{2.3}$$

Intuitively, the measurement vector may be thought of as m pieces of data that a single sensor generates from a single observation at a single instant of time. Because of measurement error, the sensor output $\boldsymbol{z}(t)$ is an approximation of $\boldsymbol{x}(t)$—the true state of nature. It is important to note that $\boldsymbol{z}(t)$ may itself not be directly visible to the user of the sensor platform. A noise corrupted

version $\Gamma\{\boldsymbol{z}(t), \boldsymbol{v}(t)\}$, as defined below, may be all that is available for processing. Furthermore, the dimensionality of the sensor data may not be the same as the dimension of the observed parameter that is being measured. For example, continuing with the airplane example, a sensor may display the longitude and latitude of the airplane at a particular instant of time via GPS (a 2-dimensional observation vector), but may not be able to measure the altitude of the airplane (which completes the 3-dimensional specification of the actual location of the airplane in space).

The *measurement error* itself can be considered as another vector, $\boldsymbol{v}(t)$, or a *noise process* vector, of the same dimensionality as the observation vector $\boldsymbol{z}(t)$. As the name suggests, noise vectors are inherently stochastic in nature, and serve to render all sensor measurements uncertain, to a specific degree.

Definiton 2.3. An *observation model*, Γ, for a sensor is a mapping from state space Ω to observation space Ψ, and is parameterized by the statistics of the noise process:

$$\Gamma_{\boldsymbol{v}} : \Omega \mapsto \Psi. \tag{2.4}$$

Functionally, the relationship between the state, observation and noise vectors can be expressed as:

$$\boldsymbol{z}(t) = \Gamma\left\{\boldsymbol{x}(t), \boldsymbol{v}(t)\right\}. \tag{2.5}$$

Objective: The objective in sensing applications is to infer the unknown state vector $\boldsymbol{x}(t)$ from the error corrupted and (possibly lower dimensional) observation vector $\boldsymbol{z}(t), \boldsymbol{v}(t)$. If the functional specification of the mapping in Equation (2.4), and the noise vector $\boldsymbol{v}(t)$, were known for all times t, then finding the inverse mapping for one-to-one cases would be trivial, and the objective would be easily achieved. It is precisely because either or both parameters may be random that gives rise to various estimation architectures for inferring the state vector from

the imperfect observations. A geometric interpretation of the objective can be presented as shown in Figure 2.3(i). The simplest mapping relationship Γ that

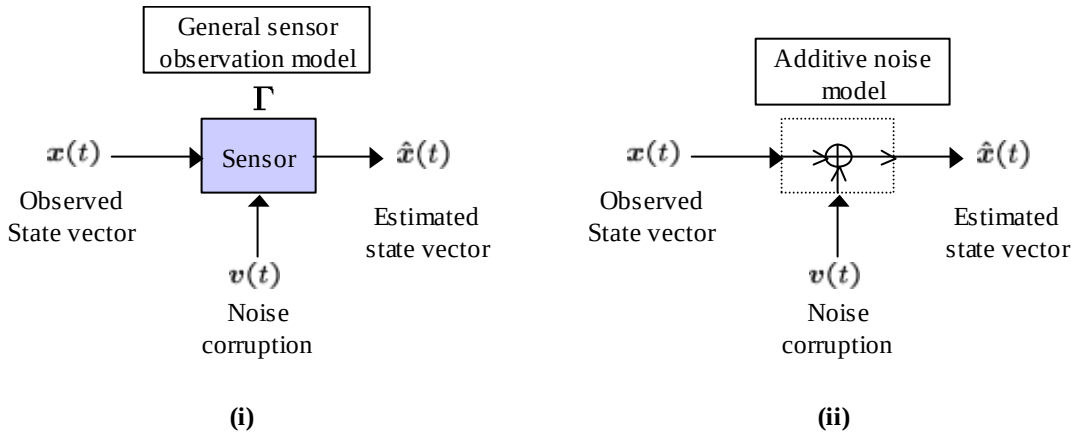

Figure 2.3: Sensor data models: (i) General case (ii) Noise additive case.

can be used as a sensor data model is the *additive* model of noise corruption, as shown in Figure 2.3(ii), which can be expressed as:

$$\boldsymbol{x} = \Gamma\left(\boldsymbol{z} + \boldsymbol{v}\right). \tag{2.6}$$

Typically, for well designed and matched sensor platforms, the noise vector is small compared to the measurement vector, in which case a Taylor approximation can be made:

$$\boldsymbol{x} = \Gamma(\boldsymbol{z}) + \left(\nabla_{\boldsymbol{z}}\Gamma\right)\boldsymbol{z} + (\text{higher order terms}) \tag{2.7}$$

where $\nabla_{\boldsymbol{z}}$ is the Jacobian matrix of the mapping Γ with respect to the state measurement vector $\boldsymbol{z}$. Since the measurement error is random, the state vector observed is also random, and we are in essence dealing with random variables. Thus, we can use well established statistical methods to quantify the uncertainty in the random variables [78]. For example, the statistics of the noise process $\boldsymbol{v}(t)$ can be often be known *a priori*. Moments are the most commonly used measures for this purpose, and in particular, if the covariance of the noise process is known,

$\mathbf{E}\left\{\boldsymbol{v}\boldsymbol{v}^T\right\}$, then the covariance of the state vector can be expressed as:

$$\mathbf{E}\left\{\boldsymbol{x}\boldsymbol{x}^T\right\} = (\nabla_{\boldsymbol{z}}\Gamma)\,\mathbf{E}\left\{\boldsymbol{v}\boldsymbol{v}^T\right\}(\nabla_{\boldsymbol{z}}\Gamma)^T. \tag{2.8}$$

For uncorrelated noise $\boldsymbol{v}$, the matrix $(\nabla_{\boldsymbol{z}}\Gamma)\,\mathbf{E}\left\{\boldsymbol{v}\boldsymbol{v}^T\right\}(\nabla_{\boldsymbol{z}}\Gamma)^T$ is symmetric and can be decomposed using singular value decomposition [80]:

$$(\nabla_{\boldsymbol{z}}\Gamma)\,\mathrm{E}\left\{\boldsymbol{v}\boldsymbol{v}^T\right\}(\nabla_{\boldsymbol{z}}\Gamma)^T = \left(\mathsf{S}\mathsf{D}\mathsf{S}^T\right) \tag{2.9}$$

where S is an $(n \times n)$ matrix of orthogonal vectors $\boldsymbol{e}_j$ and D are the eigenvalues of the decomposition:

$$\mathsf{S} = (\boldsymbol{e}_1, \boldsymbol{e}_2, \cdots, \boldsymbol{e}_n), \quad \boldsymbol{e}_i\boldsymbol{e}_j = \begin{cases} 1 \text{ for } i = j \\ 0 \text{ for } i \neq j \end{cases} \tag{2.10}$$

$$\mathsf{D} = diag\,(d_1, d_2, \ldots, d_n) \tag{2.11}$$

The scalar variance in each direction corresponding to each of the components of $\boldsymbol{x}$ is given by the corresponding component of D. When all the directions are considered for a given state $\boldsymbol{x}$, the geometrical result is an ellipsoid in n-dimensional space, with the principal axes in the directions of the vectors $\boldsymbol{e}_k$ and $2\sqrt{d_j}$ as the corresponding magnitudes. The volume of the ellipsoid is the uncertainty in $\boldsymbol{x}$. The 2-dimensional case is shown in Figure 2.4. Therefore, in a geometric sense, the aim is to reduce the volume of the uncertainty ellipsoid. All the techniques for data estimation, fusion, and inference are designed towards this goal [63].

The most celebrated method among them is the probabilistic method derived from Bayes' Law [25].

2.2.2 Bayesian Estimation and Inference

Given the inherent uncertainty in measurements of states of nature, the end goal in using sensors, as mentioned in the previous section, is to obtain the best

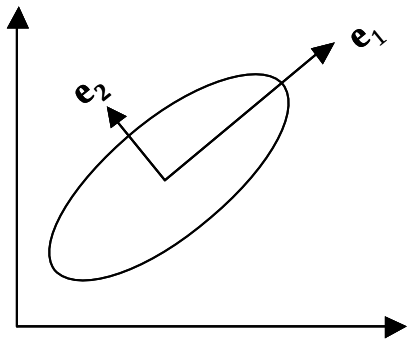

Figure 2.4: Ellipsoid of state vector uncertainty

possible estimates of the states of interest for a particular application. The Bayesian approach to solving this problem is concerned with quantifying *likelihoods* of events, given various types of partial knowledge or observations, and subsequently determining the state of nature that is most probably responsible for the observations as the 'best' estimate.

The issue of whether the Bayesian approach is intrinsically the 'best' approach for a particular problem[3] is a philosophical debate that is not discussed here further. It may be mentioned, however, that arguably, the Bayesian paradigm is most *objective* because it is based only on observations and 'impartial' models for sensors and systems.

In the Bayesian approach, the information contained in the (noise corrupted) measured state vector $\boldsymbol{z}$ is first described by means of probability distribution functions (PDF). Since all observations of states of nature are causal manifestations of the underlying processes governing the state of nature[4], the PDF of $\boldsymbol{z}$ is conditioned by the state of nature at which time the observation/measurement was made. Thus, the PDF of $\boldsymbol{z}$ conditioned by $\boldsymbol{x}$ is what is usually measurable

[3]In contrast with various other types of inferential and subjective approaches [78]

[4]Ignoring the observer-state interaction difficulties posed by Heisenberg Uncertainty considerations.

and is represented by:

$$F_{\boldsymbol{Z}}(\boldsymbol{z} \mid \boldsymbol{x}) \tag{2.12}$$

This is known as the *Likelihood Function* for the observation vector. Next, if information about the possible states under observation is available (e.g. *a priori* knowledge of the range of possible states), or more precisely the probability distribution of the possible states $F_{\boldsymbol{X}}(\boldsymbol{x})$, then the prior information and the likelihood function (2.12) can be combined to provide the *a posteriori* conditional distribution of $\boldsymbol{x}$, given $\boldsymbol{z}$, by the famous Bayes' Theorem:

Theorem 2.1.

$$F_{\boldsymbol{X}}(\boldsymbol{x} \mid \boldsymbol{z}) = \frac{F_{\boldsymbol{Z}}(\boldsymbol{z} \mid \boldsymbol{x}) F_{\boldsymbol{X}}(\boldsymbol{x})}{\int_{\Omega} F_{\boldsymbol{Z}}(\boldsymbol{z} \mid \boldsymbol{x}) F_{\boldsymbol{X}}(\boldsymbol{x}) \, \mathrm{d}F(\boldsymbol{x})} = \frac{F_{\boldsymbol{Z}}(\boldsymbol{z} \mid \boldsymbol{x}) F_{\boldsymbol{X}}(\boldsymbol{x})}{F_{\boldsymbol{Z}}(\boldsymbol{z})} \tag{2.13}$$

Usually, some function of the actual likelihood function, $g(T(\boldsymbol{z}) \mid \boldsymbol{x})$, is commonly available as the processable information from sensors. $T(\boldsymbol{z})$ is known as the *sufficient statistic* for $\boldsymbol{x}$ and Equation (2.13) can be reformulated as:

$$F_{\boldsymbol{X}}(\boldsymbol{x} \mid \boldsymbol{z}) = F_{\boldsymbol{X}}(\boldsymbol{x} \mid T(\boldsymbol{z})) = \frac{g(T(\boldsymbol{z}) \mid \boldsymbol{x}) F_{\boldsymbol{X}}(\boldsymbol{x})}{\int_{\Omega} g(T(\boldsymbol{z}) \mid \boldsymbol{x}) F_{\boldsymbol{X}}(\boldsymbol{x}) \, \mathrm{d}F(\boldsymbol{x})} \tag{2.14}$$

If the observations are assumed to be carried out in discrete time steps, according to a desired resolution, then a vector version of the above-mentioned formulation can be derived. Defining all observations upto time index r as:

$$\boldsymbol{Z}^r \triangleq \{\boldsymbol{z}(1), \boldsymbol{z}(2), \ldots, \boldsymbol{z}(r)\} \tag{2.15}$$

then the posterior distribution of $\boldsymbol{x}$ given the set of observations $vecZ^r$ is:

$$F_{\boldsymbol{X}}(\boldsymbol{x} \mid \boldsymbol{Z}^r) = \frac{F_{\boldsymbol{Z}^r}(\boldsymbol{Z}^r \mid \boldsymbol{x}) F_{\boldsymbol{X}}(\boldsymbol{x})}{F_{\boldsymbol{Z}^r}(\boldsymbol{Z}^r)} \tag{2.16}$$

Using the same idea, a recursive version of Equation (2.16) can also be formulated as follows:

$$F_{\boldsymbol{X}}(\boldsymbol{x} \mid \boldsymbol{Z}^r) = \frac{F_{\boldsymbol{Z}}(\boldsymbol{z}(r) \mid \boldsymbol{x}) F_{\boldsymbol{X}}(\boldsymbol{x} \mid \boldsymbol{Z}^{r-1})}{F_{\boldsymbol{Z}}(\boldsymbol{z}(r) \mid \boldsymbol{Z}^{r-1})} \tag{2.17}$$

in which case all the r observations do not need to be stored, and instead only the current observation $\boldsymbol{z}(r)$ can be considered at the r^{th} step. This version of the Bayes' Law is most prevalent in practice since it offers a directly implementable technique for fusing observed information with *prior beliefs*.

2.2.3 Classical Estimation Techniques

In Section 2.2.2, a likelihood function framework was developed for the measured state vectors from sensors. Given this framework, a variety of inference techniques can now be applied to estimate the state vector $\boldsymbol{x}$ (from the time series observations from a single sensor). Note that the estimate, denoted by $\hat{\boldsymbol{x}}$, is derived from the posterior distribution $F_{vecx}(\boldsymbol{x} \mid \boldsymbol{Z}^r)$ and is a point in the uncertainty ellipsoid of Figure 2.4. The objective of all the estimation techniques outlined in this section is to reduce the volume of the ellipsoid, which is equivalent to minimizing the probability of error based on some criterion. Three classical techniques are now briefly reviewed: *Maximum Likelihood, Maximum A Posteriori* and *Minimum Mean Square Error* estimation.

Maximum Likelihood (ML) estimation involves maximizing the likelihood function (Equation 2.12) by some form of search over the state space Ω:

$$\hat{\boldsymbol{x}}_{ML} = \arg\max_{\boldsymbol{x} \in \Omega} F_{\boldsymbol{Z}^r}\left(\boldsymbol{Z}^r \mid \boldsymbol{x}\right) \tag{2.18}$$

This is intuitive since the PDF is greatest when the correct state has been guessed for the conditioning variable. However, a major drawback of this technique is that for state vectors from large state spaces, the search may be computationally

expensive, or infeasible. Despite this shortcoming, this method is widely used in many disciplines, and is prominent for wireless digital reception techniques [74].

Maximum a posteriori (MAP) estimation technique involves maximizing the posterior distribution from observed data as well as from prior knowledge of the state space:

$$\hat{\boldsymbol{x}}_{MAP} = \arg\max_{\boldsymbol{x}\in\Omega} F_{\boldsymbol{x}}\left(\boldsymbol{x} \mid \boldsymbol{Z}^r\right) \tag{2.19}$$

Since prior information may be subjective, objectivity for an estimate (or the inferred state) is maintained by considering only the likelihoood function (i.e. only the observed information). In the instance of no prior knowledge, and the state space vectors are all considered to equally likely, the MAP and ML criterion can be shown to be identical.

Minimum Mean Square Error (MMSE) estimation is an estimation technique that attempts to minimize the estimation error by searching over the state space, albeit in an organized fashion. This is the most popular technique in a wide variety of information processing applications, since the variable can often be found analytically, or the search space can be reduced considerably or investigated systematically. The key notion is to reduce the covariance of the estimate. Defining the mean and variance of the posterior observation variable as:

$$\bar{\boldsymbol{x}} \triangleq \mathrm{E}_{F(\boldsymbol{x}|\boldsymbol{Z}^r)}\{x\} \tag{2.20}$$

$$\mathrm{Var}(\boldsymbol{x}) \triangleq \mathrm{E}_{F(\boldsymbol{x}|\boldsymbol{Z}^r)}\{(\boldsymbol{x}-\bar{\boldsymbol{x}})(\boldsymbol{x}-\bar{\boldsymbol{x}})^T\} \tag{2.21}$$

it can be shown that the least squares estimator is one that minimizes the Euclidean distance between the true state $\boldsymbol{x}$ and the estimate $\hat{\boldsymbol{x}}$, given the set of observations $\boldsymbol{Z}^r$. In the context of random variables, this estimator is referred to as the MMSE estimate and can be expressed as:

$$\hat{\boldsymbol{x}}_{MMSE} = \arg\min_{\boldsymbol{x}\in\Omega} \mathrm{E}_{F(\boldsymbol{x}|\boldsymbol{Z}^r)}\{(\boldsymbol{x}-\bar{\boldsymbol{x}})(\boldsymbol{x}-\bar{\boldsymbol{x}})^T\} \tag{2.22}$$

To obtain the minimizing estimate, Equation (2.22) can be differentiated with respect to $\hat{\boldsymbol{x}}$ and set equal to zero, which yields:

$$\nabla_{\hat{\boldsymbol{x}}} \int_{\boldsymbol{x}\in\Omega} \{(\boldsymbol{x}-\bar{\boldsymbol{x}})^T(\boldsymbol{x}-\bar{\boldsymbol{x}})\} F\left(\boldsymbol{x} \mid \boldsymbol{Z}^r\right) \mathrm{d}F(\boldsymbol{x})$$
$$= -2 \int_{\boldsymbol{x}\in\Omega} (\boldsymbol{x}-\bar{\boldsymbol{x}}) F\left(\boldsymbol{x} \mid \boldsymbol{Z}^r\right) \mathrm{d}F(\boldsymbol{x}) = 0$$
$$\text{from where} \quad \hat{\boldsymbol{x}} = \mathrm{E}\left\{\boldsymbol{x} \mid \boldsymbol{Z}^r\right\} \tag{2.23}$$

Thus the MMSE estimate is the conditional mean. It also can be shown that the MMSE estimate is the minimum variance estimate, and when the conditional density coincides with the mode, the MAP and MMSE estimators are equivalent.

These estimation techniques and their derivatives such as the Wiener and Kalman filters [42] all serve to reduce the uncertainty ellipsoid associated with state $\boldsymbol{x}$ [63], which was the stated objective of this section.

As mentioned at the outset in Section 2.2.1, all the techniques presented thus far are applicable only to the case of a single sensor, where multiple time-step observations are used to reduce uncertainty. When multiple, distributed sensors are involved, in a variety of configurations and topologies, some additional machinery is required to be able to combine the information from these disparate sources. This is developed in the next section.

2.2.4 Sensor Data Model for Multi-Sensor Systems

When a number of spatially and functionally different sensor systems are used to observe the same (or similar) state of nature, then the data fusion problem is no longer simply a state space uncertainty minimization issue. The distributed and multi-dimensional nature of the problem requires a technique for checking the

usefulness and validity of the data from each of the not necessarily independent sensors. The data fusion problem is more complex, and general solutions are not readily evident. This section explores some of the commonly studied techniques and proposes a novel, simplified methodology that achieves some measure of generality.

The first issue is, once again, the proper modeling of the data sources. The nomenclature and technique introduced in Section 2.2.1 can be extended to multiple sensors. If there are p sensors observing the same state vector, but from different vantage points, and each one generates its own observations, then we have a collection of observation vectors $\boldsymbol{z}_1(t)$, $\boldsymbol{z}_2(t), \ldots, \boldsymbol{z}_p(t)$, which can be represented as a combined matrix of all the observations from all sensors (at any particular time t):

$$\boldsymbol{Z}(t) = \begin{pmatrix} \boldsymbol{z}_1(t) & \boldsymbol{z}_2(t) & \cdots & \boldsymbol{z}_p(t) \end{pmatrix} = \begin{bmatrix} z_{11} & z_{21} & \cdots & z_{p1} \\ z_{12} & z_{22} & \cdots & z_{p2} \\ & & \ddots & \\ z_{1m} & z_{2m} & \cdots & z_{pm} \end{bmatrix}. \tag{2.24}$$

Furthermore, if each sensor makes observations upto time step r for a discretized (sampled) observation scheme, then the matrix of observations $\boldsymbol{Z})(\boldsymbol{r})$ can be used to represent the observations of all the p sensors at time-step r (a discrete variable, rather than the continuous $\boldsymbol{Z}(t)$). If memory is allowed for the signal processing of these data, then we can consider the super-matrix $\{\boldsymbol{Z}^r\}$ of all the observations of all the p sensors from time step 0 to time step r:

$$\{\boldsymbol{Z}^r\} = \bigcup_{i=1}^{p} \boldsymbol{Z}_i^r \tag{2.25}$$

$$\text{where} \quad \boldsymbol{Z}_i^r = \{\boldsymbol{z}_i(1), \boldsymbol{z}_i(2), \ldots \boldsymbol{z}_i(r)\} \tag{2.26}$$

This suggests that to use all the available information for effectively fusing the

data from multiple sensors, what is required is the global posterior distribution $F_{\boldsymbol{x}}(\boldsymbol{x} \mid \{\boldsymbol{Z}^r\})$, given the time-series information from each source. This can be accomplished in a variety of ways, the most common of which are summarized below.

The **Linear Opinion Pool** [89] aggregates probability distributions by linear combinations of the local posterior PDF information $F_{\boldsymbol{x}}(\boldsymbol{x} \mid \boldsymbol{Z}_i^r)$ (or appropriate likelihood functions, as per Equation (2.12)):

$$F(\boldsymbol{x} \mid \{\boldsymbol{Z}^r\}) = \sum_j w_j F\left(\boldsymbol{x} \mid \boldsymbol{Z}_j^r\right) \tag{2.27}$$

where the weights w_j sum to unity and each weight w_j represents a subjective measure of the reliability of the information from sensor j (the reliability of sources and links are discussed in further detail in Chapter **Insert Reliability Chapter reference here!!!!!!!**. The process can be illustrated as shown in Figure 2.5. Bayes' theorem can now be applied to Equation (2.27) to obtain a

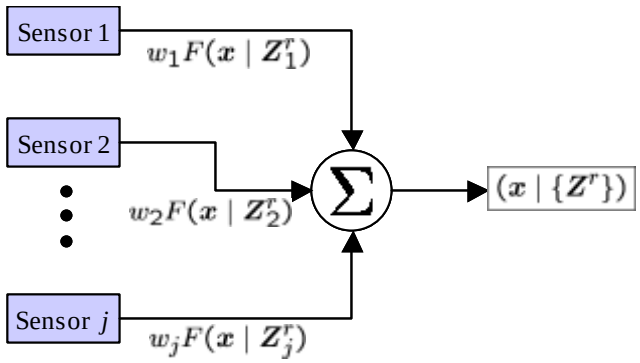

Figure 2.5: Multi-Sensor Data Fusion by Linear Opinion Pool

recursive form, which is omitted here for brevity. It is well known that one of the shortcomings of the linear opinion pool method is its inability to reinforce opinion because the weights are usually unknown except in very specific applications.

The **Independent Opinion Pool** is a product form modification of the linear

opinion pool and is defined by the product:

$$F\left(\boldsymbol{x} \mid \{\boldsymbol{Z}^r\}\right) = \alpha \prod_j F\left(\boldsymbol{x} \mid \boldsymbol{Z}_j^r\right) \tag{2.28}$$

where α is a normalizing constant. The fusion process in this instance can be illustrated as shown in Figure 2.6

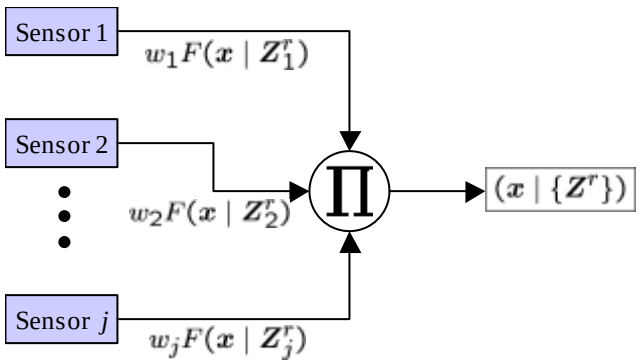

Figure 2.6: Multi-Sensor Data Fusion by Independent Opinion Pool

This model is widely used since it represents the case when the observations from the individual sensors are essentially independent. However, this is also its weakness, since if the data is correlated at a group of nodes, their opinion is multiplicatively reinforced, which can lead to error propagation in faulty sensor networks. Nevertheless, this technique is appropriate when the prior state space distributions are truly independent and equally likely (as is common in digital communication applications).

To counter the weaknesses of the two common approaches summarized above, we propose a third fusion rule, which we refer to as the **Likelihood Opinion Pool**.

Theorem 2.2 (Likelihood Opinion Pool Principle). The Likelihood Opinion

Pool Data Fusion rule can be formulated as the following recursive rule:

$$F\left(\boldsymbol{x} \mid \{\boldsymbol{Z}^r\}\right) = \alpha F\left(\boldsymbol{x} \mid \left\{\boldsymbol{Z}^{r-1}\right\}\right) \left[\prod_j \underbrace{F\left(\boldsymbol{z}_j(r) \mid \boldsymbol{x}\right)}_{\text{likelihood}}\right] \tag{2.29}$$

Proof. The distribution of $\boldsymbol{x}$ conditioned on all the observations upto time r is given by Bayes Theorem:

$$\begin{aligned} F\left(\boldsymbol{x} \mid \{\boldsymbol{Z}^r\}\right) &= \frac{F\left(\{\boldsymbol{Z}^r\}\right) \mid \boldsymbol{x}) F(\boldsymbol{x})}{F\left(\{\boldsymbol{Z}^r\}\right)} \\ &= \frac{F\left(\boldsymbol{Z}_1^r, \boldsymbol{Z}_2^r, \ldots \boldsymbol{Z}_p^r \mid \boldsymbol{x}\right) F(\boldsymbol{x})}{F\left(\boldsymbol{Z}_1^r, \boldsymbol{Z}_2^r, \ldots \boldsymbol{Z}_p^r\right)} \end{aligned} \tag{2.30}$$

For sensor systems, it is reasonable to assume that likelihoods from each sensor i, that is $F\left(\boldsymbol{Z}_i^r \mid \boldsymbol{x}\right)$ are independent because the only parameter that the observations have in common is the state. Therefore:

$$F\left(\boldsymbol{Z}_1^r, \boldsymbol{Z}_2^r, \ldots \boldsymbol{Z}_p^r \mid \boldsymbol{x}\right) = F\left(\boldsymbol{Z}_1^r \mid \boldsymbol{x}\right) F\left(\boldsymbol{Z}_2^r \mid \boldsymbol{x}\right) \cdots F\left(\boldsymbol{Z}_p^r \mid \boldsymbol{x}\right) \tag{2.31}$$

$$\implies F\left(\boldsymbol{x} \mid \{\boldsymbol{Z}^r\}\right) = \frac{F\left(\boldsymbol{Z}_1^r \mid \boldsymbol{x}\right) F\left(\boldsymbol{Z}_2^r \mid \boldsymbol{x}\right) \cdots F\left(\boldsymbol{Z}_p^r \mid \boldsymbol{x}\right) F(\boldsymbol{x})}{F\left(\boldsymbol{Z}_1^r, \boldsymbol{Z}_2^r, \ldots \boldsymbol{Z}_p^r \mid \boldsymbol{x}\right)} \tag{2.32}$$

which can then recursively be written as shown in the statement of the theorem.

□

The Likelihood Opinion Pool method of data fusion can be illustrated as shown in 2.7. The likelihood opinion pool technique is essentially a Bayesian update process and is consistent with the recursive process derived in general in Equation (2.17). It is interesting to note that a simplified, specific form of this type of information processing occurs in the so called *belief propagation* [67] type of algorithms that is widespread in artificial intelligence and the decoding theory for channel codes. In our exposition, however, the assumptions and derivations have been identified and explicitly derived, and stated in a general form suitable for application to multi-sensor systems. This offers us valuable insight as to

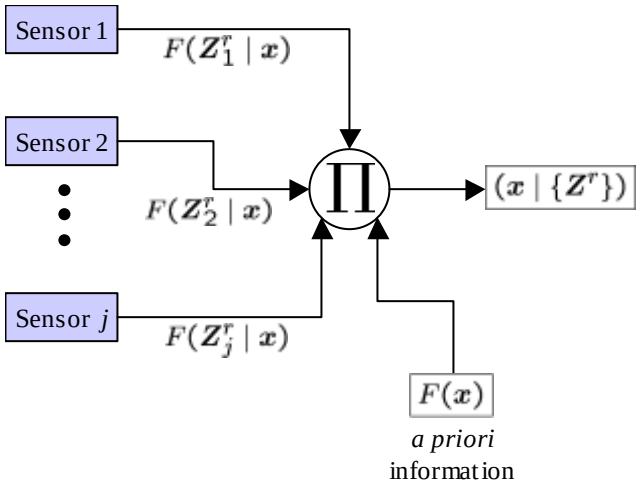

Figure 2.7: Multi-Sensor Data Fusion by Likelihood Opinion Pool

how the probabilistic updates help to reinforce the 'opinions' when performing a distributed state space search. This viewpoint will also be used when we make the connection with channel decoding techniques in Chapter 6, where we discuss Low Density Parity Check codes for the design of bandwidth efficient sensor nodes.

2.3 Information Theoretic Justification of the Bayesian Method

In Section 2.2, probability distributions were seen to be the key element that allowed a quantitative description of the observables, the observer, and associated errors. As such, the likelihood functions and distributions contain information about the underlying states that they describe. This approach can be extended further to actually incorporate measures for the information contained in these random variables. In this manner, an information theoretic justification can be obtained for the proposed Likelihood Opinion Pool for multi-sensor data fusion, which is the objective of this section. Some key concepts from Information Theory

are required first.

2.3.1 Information Measures

Information theory was developed to determine the fundamental limits on the performance of communication systems [81]. Detection theory on the other hand, as we have seen, involves the application of statistical decision theory to estimate states of nature. Both these disciplines can be used to treat problems in the transmission and reception of information. This synergy was first explored by researchers in the 1950s and 1960s [60], and the well established source and channel coding theories have spawned as a result.

A similar approach can be taken to leverage the fundamental concepts of these disciplines for multi-sensor data fusion and distributed detection problems. This has been studied since the 1970s and 1980s [96], and a brief survey was provided in Section 2.1.1. In Section 2.2, we also presented the Bayesian approach which we believe is the most straightforward application of the key results from detection theory for accomplishing distributed data-fusion. In this section, an information theoretic justification is provided as to the utility of the Bayesian update method. It also serves to provide insight towards the practical design of algorithms based on the proposed likelihood opinion pool fusion rules, which are discussed in Section 2.4.

To build an information theoretic foundation for data fusion, the most useful fundamental metric is the Shannon definition of *Entropy*.

Definiton 2.4. Entropy is the uncertainty associated with a probability distribution, and is a measure of the descriptive complexity of a PDF [11]. Mathematically:

$$h\{F(\boldsymbol{x})\} \triangleq \mathrm{E}\{-\ln F(\boldsymbol{x})\} \tag{2.33}$$

Note that alternative definitions of the concept of information which predate Shannon's formulation, e.g. the *Fisher Information Matrix* [20], are also relevant and useful, but not discussed here further.

Using this definition, an expression for the entropy of the posterior distribution of $\boldsymbol{x}$ given $\boldsymbol{Z}^r$ at time r (which is the case of multiple observations from a single sensor) can be expressed as:

$$h(r) \triangleq h\left\{F(\boldsymbol{x} \mid \boldsymbol{Z}^r)\right\} = -\sum F(\boldsymbol{x} \mid \boldsymbol{Z}^r) lnF(\boldsymbol{x} \mid \boldsymbol{Z}^r) \tag{2.34}$$

From this definition, the entropy relationship for Bayes Theorem can be developed as follows:

$$\begin{aligned} \mathrm{E}\left\{-\ln[F(\boldsymbol{x} \mid \boldsymbol{Z}^r)]\right\} &= \mathrm{E}\left\{-\ln[F(\boldsymbol{x} \mid \boldsymbol{Z}^{r-1})]\right\} \\ &\quad - \mathrm{E}\left\{\ln\left[\frac{F(\boldsymbol{z}(r) \mid \boldsymbol{x})}{F(\boldsymbol{z}(r) \mid \boldsymbol{Z}^{r-1})}\right]\right\} \qquad (2.35) \\ \implies h(r) &= h(r-1) - \mathrm{E}\left\{\ln\left[\frac{F(\boldsymbol{z}(r) \mid \boldsymbol{x})}{F(\boldsymbol{z}(r) \mid \boldsymbol{Z}^{r-1})}\right]\right\} \qquad (2.36) \end{aligned}$$

This is an alternative form of the result that conditioning with respect to observations reduces entropy (cf. [14]). Using the definition of mutual information, Equation (2.35) can be written in an alternative form as shown below.

Definiton 2.5. For an observation process, *mutual information* at time r is the information about $\boldsymbol{x}$ contained in the observation $\boldsymbol{z}(r)$:

$$\mathrm{I}(\boldsymbol{x}, \boldsymbol{z}(r)) \triangleq \mathrm{E}\left\{\ln\left[\frac{F(\boldsymbol{z}(r) \mid \boldsymbol{x})}{F(\boldsymbol{z}(r))}\right]\right\} \tag{2.37}$$

from where

$$h(r) = h(r-1) - \mathrm{I}(r) \tag{2.38}$$

which means that the entropy following an observation is reduced by an amount equal to the information inherent in the observation.

The insight to be gained here is that by using the above mentioned definitions of entropy and mutual information, the recursive Bayes update procedure derived in Section 2.2.2 Equation (2.17) can now be seen as an information update procedure:

$$\mathrm{E}\left\{\ln[F(\boldsymbol{x} \mid \boldsymbol{Z}^r)]\right\} = \mathrm{E}\left\{\ln[F(\boldsymbol{x} \mid \boldsymbol{Z}^{r-1})]\right\} + \mathrm{E}\left\{\ln\left[\frac{F(\boldsymbol{z}(r) \mid \boldsymbol{x})}{F(\boldsymbol{z}(r) \mid \boldsymbol{Z}^{r-1})}\right]\right\} \tag{2.39}$$

which can be interpreted as:

posterior information = prior information + observation information.

The information update equations can then be written for the proposed Likelihood Opinion Pool fusion rule, which becomes:

$$\begin{aligned}\mathrm{E}\left\{\ln[F(\boldsymbol{x} \mid \boldsymbol{Z}^r)]\right\} &= \mathrm{E}\left\{\ln[F(\boldsymbol{x} \mid \boldsymbol{Z}^{r-1})]\right\} \\ &\quad + \sum_j \mathrm{E}\left\{\ln\left[\frac{F(\boldsymbol{z}_j(r) \mid \boldsymbol{x})}{F(\boldsymbol{z}_j(r) \mid \boldsymbol{Z}^{r-1})}\right]\right\}\end{aligned} \tag{2.40}$$

The utility of the log-likelihood definition is in the fact that the information update steps then reduce to simple additions, and are thus amenable to hardware implementation without problems of overflow, dynamic range scaling, etc.

This section has thus shown that the Bayesian probabilistic approach is theoretically self-sufficient for providing a unified framework for data fusion in multi-sensor platforms. The information theoretic connection to the Bayesian update makes the approach intuitive, and shows rigorously how the proposed Likelihood Opinion Pool method serves to reduce the ellipsoid uncertainty.

In the following sections, these theoretical developments are used to design and determine the actual architectures that can be used for data fusion in practical multi-sensor systems.

2.4 Multi-Sensor Data Fusion Architectures

The information processing approach of 2.2 allows us to reduce the problem of data fusion, regardless of the application, network or physical layer design, to one of information fusion. This, as we have seen, is basically a Bayesian update of likelihoods. Thus, the problem of designing an appropriate architecture for data fusion can be decoupled from system-specific concerns. This is a major advantage. The general results derived in the previous sections can now be used to determine appropriate platform independent data fusion architectures for distributed sensor networks.

Some versions of this concept were recognized early in the robotics and vision systems community [63, 21]. In the following sections, the architectures are classified and a novel scheme based on the Likelihood Opinion Pool formulation is provided.

2.4.1 Classification of Sensor Network Architectures

The information processing approach can be used to design three distinct types of sensor data fusion network architectures.

1. Centralized Architecture
2. Hierarchical Architecture
3. Distributed Architecture (fully and non-fully connected)

We provide the general definitions of each of these types of architectures in the following paragraphs, borrowing terminology from graph theory [100].

Definiton 2.6. A *Centralized Architecture* for multi-sensor data fusion is a star network topological arrangement of sensor nodes, with a central processor at the

root that is responsible for collecting, processing and interpreting the measurements from all sensor devices.

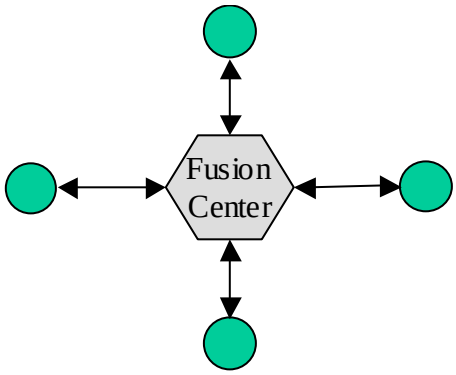

Figure 2.8: Centralized Data Fusion Architecture

The architecture can be visualized as shown in Figure 2.8. This is a classical design and is an extension of the layout used for single sensor systems. It has a number of advantages, the primary one being that the data fusion algorithm can be a relatively simple modification of that used for single sensor systems—thus legacy systems can be well-supported. Also, synchronization issues are not a problem since the central processor has access to all the data nodes directly, and can pool all the information in the correct sequence as well as identify faulty nodes. This system is highly tolerant of sensor failures (but highly intolerant of processor failures), and thus has found widespread use in robotics, automobiles, aircraft, process control and other applications. The fault tolerance issues in similar architectures for mobile wireless networks are further discussed in Chapter 5. The engine combustion control system in automobiles is a classic example, pooling information from a multitude of sensors to a central processor for processing, interpretation and deciding upon the engine control parameters.

The main disadvantage of the centralized architecture is in its inability to scale and its inflexibility to change. Since the processor requires direct access

to each of the sensors, the system is unmanageable for large numbers of sensors. The computational burden on the processor grows exponentially. In a wired application, the problems of physically connecting the nodes to the processor also rules out its use for distributed applications. In mobile wireless scenarios, direct single-hop links from the fusion center to all distributed nodes are hardly always feasible, and the overhead and QoS issues involved in establishing and maintaining the required star-network with the processor as the root can be overwhelming. Finally, processor fault (in)tolerance is a catastrophic shortcoming. Any failures of the central processor renders the architecture useless. To counter this defect, mission critical systems often have double and *triple redundancy modes* [50], e.g. the guidance system in the Space Shuttle. However, redundancy may not be cost effective or manageable in applications requiring mass scenarios.

To counter some of the disadvantages of centralized architectures, hierarchical fusion arrangements have been devised.

Definiton 2.7. A *Hierarchical Architecture* for multi-sensor data fusion is a tree network topological arrangement of sensors and processors. The tree is rooted at the global fusion center, and the sensors occupy the leaves of the tree, and intermediate nodes serve as the local fusion processors.

A hierarchical arrangement can be illustrated as shown in Figure 2.9. In such schemes, local fusion processors handle the data from a subset of the sensors, and a global fusion center then pools the information from the local fusion sources for processing and inference. The global fusion center serves to relieve the computational burden on each of the local processors by a divide and conquer strategy.

The primary advantage of the hierarchical system is in its ability to incorporate different classes of sensors as part of different local groups. Hence, heterogeneous types of networked sensor systems can be supported in this scheme.

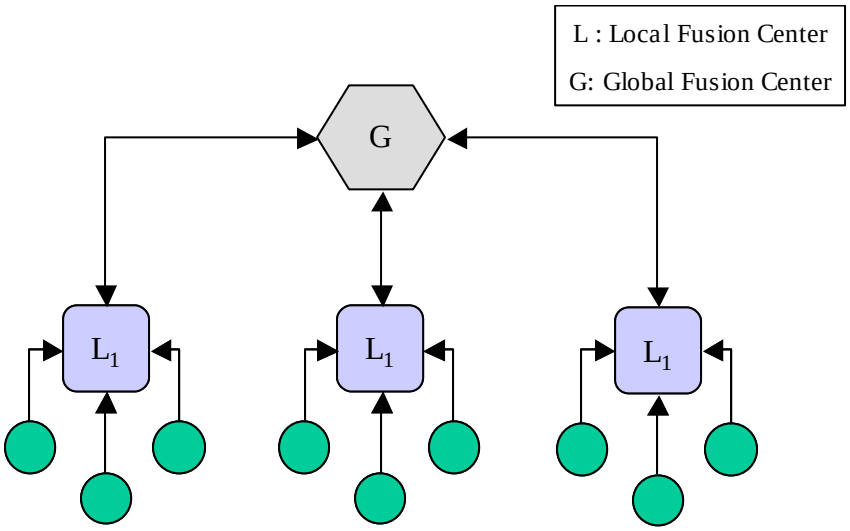

Figure 2.9: Hierarchical Data Fusion Architecture

To draw upon the example of the position sensors of an airplane mentioned in Section 2.2.1, the hierarchical fusion center for this application can pool the latitude and longitude data from the GPS receiver, with the altitude data from the altimeter (even though they are different sensors with different physical models) to determine the unique position of the aircraft in flight.[5] The global fusion processor also relieves the computational demands on the local sensor processors by introducing multiple levels of abstraction. A hierarchical architecture is precisely the scheme that we have used to enable connectivity in a hybrid mobile ad hoc networks, as discussed in detail in Chapter 4.

The main disadvantage of hierarchical systems is increased system complexity owing to the fact that two separate fusion algorithms are required—at the local and global fusion levels. Furthermore, the communication requirements among the various units, in particular the global and local processors, may be a performance bottleneck in wireless or distributed systems. The global node in the

[5]It is of interest to note that the human nervous system can also be considered as a hierarchical sensor architecture, with the brain serving as the global fusion center.

hierarchy also inherits the vulnerability to failures as in the centralized architecture case. However, it is of interest to note that the global fusion process may itself reside on one (or more) of the local fusion centers. This enables a form of processor redundancy fault tolerance. In case the global fusion center fails, and one of the other surviving local fusion centers can function as a global center, then fusion control can pass to this new global fusion center. There are issues regarding handshaking and leader election under such circumstances. However, these are manageable, and the scheme has been implemented in the context of gateways in hybrid mobile ad hoc networks of processors (Chapter 4).

For maximum flexibility, a decentralized system for multi-sensor data fusion offers the best means of overcoming the problems associated with both centralized and hierarchical architectures.

Definiton 2.8. A *Decentralized Architecture* for multi-sensor data fusion is an ad hoc network topology of sensors nodes and data fusion processors, with at least one connected component consisting of sensors and processors.

A decentralized architecture is an example of an arbitrary ad hoc network of nodes (Section 4.2). However, for the case of sensor networks, it is of interest to have at least one connected component of sensors and processors, because, without communication between these elements, there can be no fusion process possible. Thus, one extreme case of a decentralized architecture is a fully connected graph of sensor and processor nodes, and the other extreme is an arbitrarily connected network of sensor and processor nodes (Figure 2.10).

Decentralized architectures do not rely on any specific sensor, or fusion processor (which may be the same physical device), for its operations. Therefore, any fault or failure of any subset of the nodes do not cause a catastrophic failure of the system. This has very desirable fault tolerance properties. Furthermore,

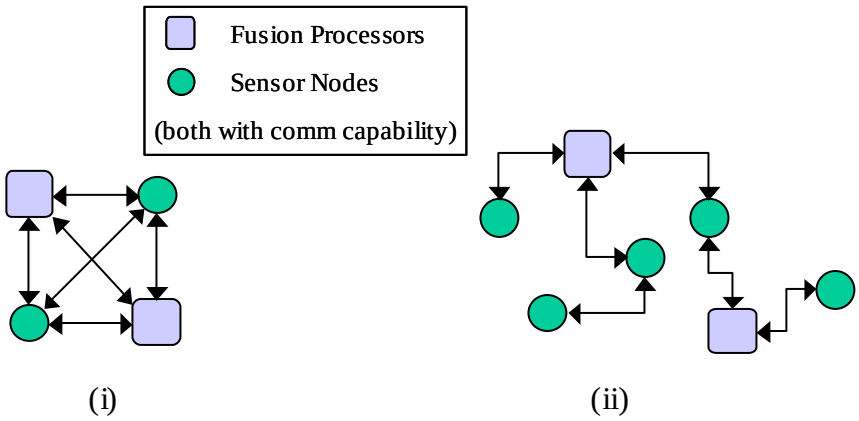

Figure 2.10: Decentralized Data Fusion Architectures: (i) Fully Connected (ii) Arbitrary

changes in sensor technology are easy to implement, since they affect only subsets of nodes at a time, and there is no 'down-time'. This also allows for adaptive algorithms to be implemented on such platforms, where selective groups of nodes and processors are dynamically re-programmed for specific applications.

These advantages come at the expense of significant organizational and engineering complexity. Unlike centralized schemes, decentralized architectures require a common network and communication protocol to be able to exchange data. Synchronization of the events, observations, inferences, and interpretations require coordination among disparate and distributed participants, and may not be possible under all circumstances. Furthermore, inter-operability has to be engineered into systems before they can function in a truly heterogeneous, ad hoc environment. Unfortunately, some of these problems have been shown to have no efficient algorithmic solutions [93], so a plethora of heuristic techniques have been developed with differing objectives. Some of these issues are further discussed in the context of networks in general ad hoc networks in Chapters 4.

2.4.2 An Architecture for Likelihood Opinion Pool Data Fusion

In this section, the architectural concepts outlined in Section 2.4.1 are extended to introduce a generalized form of a distributed multi-sensor data fusion architecture, based on the Likelihood Opinion Pool formulation.

To begin with, a choice needs to be made at the outset regarding the basic type of information that will be transmitted in a multi-sensor system. For example, upon observation and measurement of a state of nature, a sensor may transmit either the raw data $\boldsymbol{z}(r)$ at each time step r, or the likelihood information $[\alpha_i F(\boldsymbol{z}_i(r) \mid \boldsymbol{x})]$, or the sufficient statistics formulations $F(T\{\boldsymbol{z}_i(r)\} \mid \boldsymbol{x})$ (Section 2.2.4). However, the communication of raw sensor data requires the central processor to have knowledge of the models for each sensor so that the likelihoods can be computed from the observations from all the sensors. This is an inflexible option. Instead, as we have seen from Section 2.3.1, the likelihood contains essentially the same information, but in a much more convenient form, since then the sensor model can be kept locally at each sensor and only 'soft' metrics can be sent to a general fusion processor. From a systems implementation and computation point of view, this is an attractive choice. Sufficient statistics metrics are essentially similar, the only difference being a data space transformation; the fusion processor still does not need to maintain a sensor profile for each sensor in the network.

With likelihoods being the currency, as it were, being traded in the sensor network, we note that in the Bayesian information processing formulation (Section 2.3), the update at time step r for a single sensor can be written in terms of the log-likelihood as:

$$\ln F(\boldsymbol{x} \mid \boldsymbol{Z}^r) = \ln F(\boldsymbol{x} \mid \boldsymbol{Z}^{r-1}) + \ln[\alpha F(\boldsymbol{z}(k) \mid \boldsymbol{x})] \tag{2.41}$$

This recursive law expresses the information processing that should takes place in a sensor and can be used to guide the design of a information processor for a single sensor. The block diagram is illustrated in Figure 2.11.

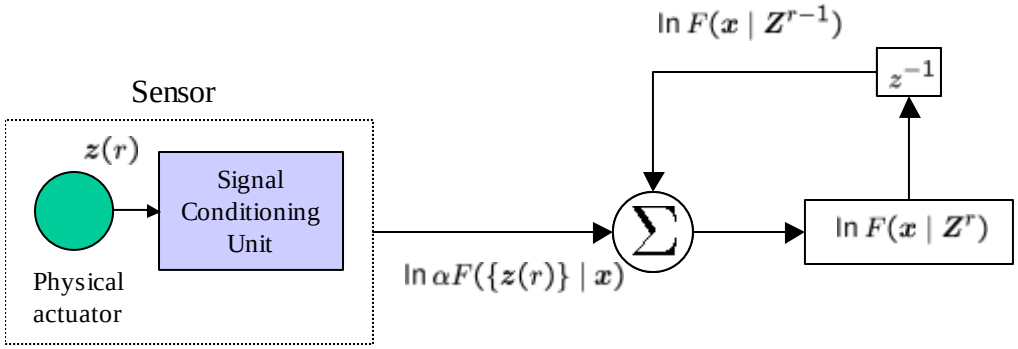

Figure 2.11: Likelihood Information Processing for Single Sensors

In the figure, the sensor is shown as a physical device that generates raw data. The signal conditioning unit pre-processes the data (filtering, sampling, smoothing, etc.) and generates the likelihood information according to its specific sensor model. In comparison with channel coding theory, this is a 'soft' decision metric (continuous valued), rather than 'hard' decisions (binary 0-1 type information). The feedback block then implements the recursive Bayesian update and combines the likelihood information with the log of the distribution of the state, given all the time step observations thus far, to generate the posterior distribution of the state.

The single sensor likelihood processing unit can be used as the building block for implementing all the multi-sensor architectures that have described in Section 2.4.1. We omit the discussion for centralized and hierarchical architectures, since our interest is in obtaining a design for the decentralized, distributed case. In a distributed system, each node makes its own observations, if so designed, but is also required to obtain the global posterior by communicating information

with the other observers in the system. If the system is fully connected, then each node has access to every other node's information, so the global posterior computation should be identical at each node. However, this situation is neither always feasible (the network may not be fully connected), nor always desired (only a subset of a large number of sensor nodes may be involved in a particular observation scenario).

In the discussion that follows, we assume that the state of nature being observed at sensor i is $\boldsymbol{x}_i$. Then, for the case of multi-sensor systems designed to operate under the Likelihood Opinion Pool data fusion rule, we have seen in Section 2.2.4, Equation (2.29) that the fusion rule is simply a product of the likelihoods (or a sum of the log-likelihoods). Therefore, at each sensor i, the data fusion can be performed by a simple log update relationship as follows:

$$\boxed{\ln F(\boldsymbol{x}_i \mid \boldsymbol{Z}^r) = \ln F(\boldsymbol{x}_i \mid \boldsymbol{Z}^{r-1}) + \sum_j \underbrace{\ln\left[\alpha_j F(\boldsymbol{z}_j(r) \mid \boldsymbol{x}_j\right]}_{\text{communicated}}} \tag{2.42}$$

Furthermore, each sensor node i also computes a local partial posterior distribution based only on the local observation information and a global prior:

$$\ln F(\boldsymbol{x}_i \mid \{\boldsymbol{Z}^{r-1}\} \cup \boldsymbol{z}_i(r)) = \ln F(\boldsymbol{x}_i \mid \{\boldsymbol{Z}^{r-1}\}) + \ln[\alpha_i F(\boldsymbol{z}_i(r) \mid \boldsymbol{x}_i)] \tag{2.43}$$

This partial posterior is a summation of information known globally before time step r and the information that the sensor i contributes at time r. This can be exploited by the other data processors in the network to gauge the *quality* of the information that the sensor provides—a sort of fault detection in a distributed environment.

Thus, the main result of this section—the general architecture for multi-sensor data fusion using the likelihood opinion pool—can be summarized and pictured as shown in Figure 2.12.

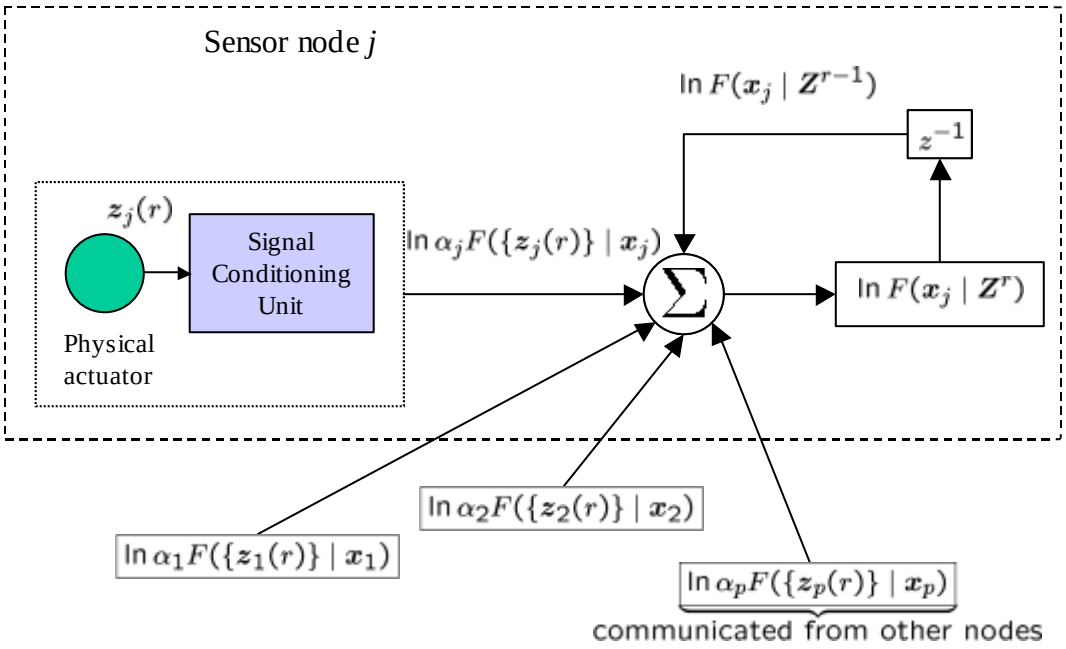

Figure 2.12: Architecture for Multi-sensor Data Fusion using Likelihood Opinion Pool

2.5 Concluding Remarks

In this chapter, we have presented a probabilistic, information processing approach to data fusion in multi-sensor networks. The Bayesian approach was seen to be the central unifying tool in formulating the key concepts and techniques for decentralized organization of information. Thus, it offers an attractive paradigm for implementation in a wide variety of systems and applications. An information theoretic justification of the method was evident, and actual architectures for the proposed fusion scheme was also seen to be a natural consequence of the formulation.

While this technique is certainly not the only approach possible for sensor networks, it is our contention that this offers a simple, yet comprehensive guide for most practical systems of interest. The remaining chapters in this thesis

explore some of these practical issues in greater detail, concentrating on the physical and network layer aspects of the problem of decentralized information processing in wireless networks.

CHAPTER 3

Information Processing Bounds in Data Networks: Simplified Cases

In the previous chapter, we considered information processing as the end goal of all network and communication management tasks, which we collectively to referred to as *network resource administration* (Section 2.1). Needless to say, inefficient adminstration of networking resources and tasks, such as routing, medium access control (MAC), physical layer issues (e.g. coding, modulation, power levels, etc.) can make the whole information processing objective moot. Thus, general bounds are desirable on the what is, and what is not, possible with a given set of resources and constraints, for a given set of objectives. The proper match between the two is the entire field of communication and network engineering.

Many results have been derived over the last half century with regards to the bounds on performance of protocols at all levels of the OSI stack ([81, 74, 91]). In this chapter we consider two special cases; the rate distortion region for the n-Helper Gaussian network, and the asymptotic delay in random wireless networks.

Specifically, we consider the wireless sensor and information processing networks of Chapter 2 from an information theory point of view and bound the rate distortion region for the special case of correlated Gaussian sources where n sources provides partial side information to one main source. We show that an explicit expression for the lower bound of the rate is possible. This is discussed

in Section 3.1. The main utility of this result is to enable comparison of the performance of the practical data fusion schemes that were suggested in Section 2.4.2 with the theoretically predicted limits.

In the context of quality of service (QoS) for wireless networks, we also consider the problem of delay that information packets encounter as a result of the resource administration tasks undertaken by the various protocol layers in practical networks. Despite voluminous analysis of the many different types of routing, medium access, and physical layer techniques and protocols, there remains a need to analytically model the entire network stack cohesively to evaluate end-to-end performance bounds. Hence, it is still not clear what the performance limits of such networks are likely to be, especially when scaled to large numbers of nodes. We tackle a simplified version of this scenario by considering parameters and conditions that can be extended to the limit, to simplify the analytical difficulties, and present a limiting study of the 'average' delay characteristics for information packets in mobile networks. We show that, in the limit, the delay observed per information bit that is transmitted by a 'typical' node in the network averages essentially to the order of $O\left(\sqrt{n}\right)$ as a first approximation. This is discussed in Section 3.2.1.

3.1 Rate Distortion of n-Helper Gaussian Sources

The main problem that this section addresses is: what data rates can be supported for data fusion in a sensor network, given a specified data distortion level that can be tolerated? Unfortunately, an exact answer to this question is not available in the general case, and thus we have considered the specific case of correlated Gaussian sources in this paper.

The motivation for this problem comes from performing distributed detection of phenomena. As discussed in detail in Chapter 2, it is well known from the theory of distributed detection that higher reliability and lower probability of detection error can be achieved when observation data from multiple, distributed sources is intelligently fused in a decision making algorithm, rather than using a single observation data set [96]. This, coupled with the fact that fabrication technological advances have made low-cost sensors incorporating wireless transceivers, signal processing and sensing in one integrated package a desirable low-cost option, it is inevitable that such devices will be widely used in detection applications such as security, monitoring, diagnostic, remote exploration etc. This has given rise to the development of wireless integrated networked sensors (WINS) [73], as shown in general in Figure 3.1.

However, the effective deployment of such distributed processing systems introduces some significant design issues, most notably: networking and communication protocols, transmission channel and power constraints, and scalability, among others [72, 73],. However, it is also evident that some fundamental limits are required to assess the optimality of any system design with regard to the "best design". Thus, an information theoretic analysis of the system is required. We have assumed that the primary constraint for our applications of interest is power.

A WINS system invokes a multi-terminal analysis, as diagrammed in 3.1.

For this type of a system, all the traditional types of multi-terminal channels considered in information theory appear: the multiple access channel (communication pathways shown in 3.1 numbered as 1), the broadcast channel (2), the relay and interference channel (3), etc. [28]. Additionally, the channel may be fading or more complex. Unfortunately, in the absence of a general information

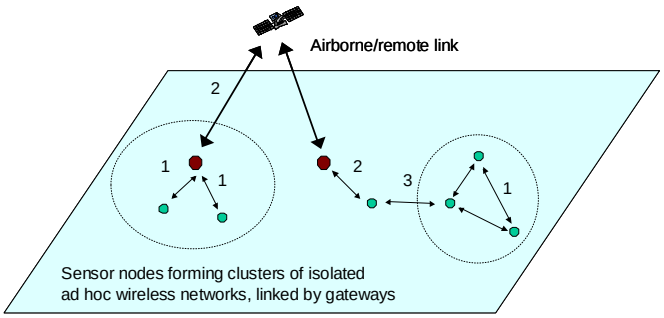

Figure 3.1: Wireless Integrated Network Sensor System

theory of multi-terminal networks, there is, as yet, no analytical way of evaluating performance bounds for whole systems, such as one illustrated above, for a specific type of task for which the network might be employed, e.g. distributed detection. It is the overall goal of our project to apply the results known so far to obtain, if not global optimum information limits, but optimality criteria for each of the individual sub-blocks. In this regard, advances have been made with certain simplifying considerations, most notably the rate-distortion bounds for multiple, correlated nodes. The focus so far has been on pathways 1, inside the local loop.

For the individual local network loops, the problem is one of efficient communication and data fusion for detection. Associated problems such as network boot-strap, algorithms determining the minimum number of nodes necessary for reliable detection of a phenomenon, etc. have been studied and are not further discussed here [86]. Instead the coding problem is considered. The multi-terminal coding theory problem for two correlated memory-less sources with separate encoders has been solved by Slepian and Wolf [84]. The correlated sources assumption is valid in the WINS case, since for nodes observing the same target, the data generated for each sensor is expected to be correlated. Also, in the WINS case, it is apparent that power efficiency can be incorporated by allowing a distortion

criterion, since there may be several data fusion centers, and since local processing provides a far higher power gain than RF transmissions. Thus, what becomes of interest then is how much distortion can be tolerated if the sensor network is to achieve some measure of efficiency in distributed detection – in other words, the rate-distortion bound.

Previous work in this area by Wyner and Ziv [102], Han and Kobayashi [34, 33], and Csizar [17] have all focused on special extensions of Slepian and Wolf, but the general rate-distortion regions characterization problem has remained unsolved. We have extended the special case for correlated memoryless Gaussian sources to the $n-$ sources case (with partial side information). The next section presents the analytic formulation of the problem and the main result.

3.1.1 Analytic Formulation

Consider the multi-sensor system as shown below (Figure 3.2).

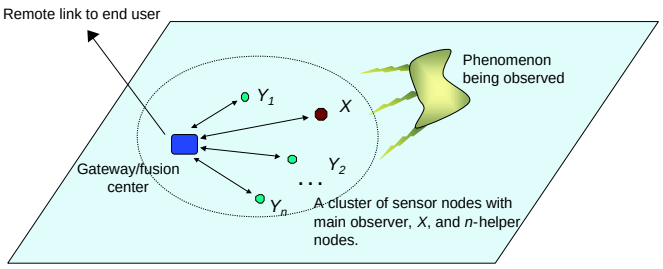

Figure 3.2: Data Fusion for a Wireless Networked Sensor System

A portion of a distributed cluster of sensor nodes (perhaps mobile) is observing a phenomenon and generating source data. Algorithms exist which can determine which nodes in the proximity of the phenomenon need to be activated and which can remain dormant [28]. Once this boot-up process is completed, the node observation data is assumed to be Gaussian (for analytical simplicity), with one

data node acting as the main data source (e.g. that which is closest to the phenomenon), and the remaining nodes generating correlated data. The coding challenge is then to determine appropriate codes and data rates such that the gateway/data-fusion center can reproduce the data from the main node using the remaining nodes as sources of partial side information, subject to some distortion criteria.

Thus for a main source, X, and n correlated sources, Y_i, such that

$$\{X_t, Y_{1t}, \cdots Y_{nt}\}_{t=1}^{\infty}$$

are stationary Gaussian memoryless sources, for each observation time, t=1, 2, 3, ..., we let the random $(n+1)$-tuplet $(X_t, Y_{1t}, \cdots Y_{nt})$ take values in $X \times Y_1 \times \cdots Y_n$. The joint probability density function is given by the usual expression for the multi-dimensional Gaussian probability density function, where the covariance matrix can be denoted as:

$$\begin{bmatrix} \sigma_X^2 & \rho_{XY_1}\sigma_X\sigma_{Y_1} & \cdots & \rho_{XY_n}\sigma_X\sigma_{Y_n} \\ \rho_{XY_1}\sigma_X\sigma_{Y_1} & \sigma_{Y_1}^2 & \cdots & \rho_{Y_1Y_n}\sigma_{Y_1}\sigma_{Y_n} \\ \vdots & \vdots & \ddots & \vdots \\ \rho_{XY_n}\sigma_X\sigma_{Y_n} & \rho_{Y_1Y_n}\sigma_X\sigma_{Y_1} & \cdots & \sigma_{Y_n}^2 \end{bmatrix} \tag{3.1}$$

We can write independent copies of $\{X_t\}_{t=1}^{\infty}$ as $X^n = X_1, X_2, \cdots, X_n$ and similarly for $\{Y_t\}_{t=1}^{\infty}$, *k=1,2,...,n*. Next, we consider a coding system where data sequences $X^n,\ Y^n,\ Z^n$ are separately encoded to $\varphi_1(X^n),\ \varphi_1(Y^n),\ \varphi_1(Z^n)$ and sent to the information processing / data fusion center. The decoder function, $\psi = (\psi_1, \psi_2, \psi_3)$, observes the $(n+1)-$tuplet $\{\varphi_1(X^n),\ \varphi_1(Y^n),\ \varphi_1(Z^n)\}$ and estimates $\left(\hat{X}^n, \hat{Y}^n, \hat{Z}^n\right)$. We let $\Im_{n,\delta}(R_1, R_2, R_3)$ denote the set of all such coding and decoding schemes, $(\varphi_1, \varphi_2, \varphi_3, \psi)$, which can exist with the properties mentioned above. We take :

$$d_1 : X^2 \to [0, \infty), \quad d_2 : Y^2 \to [0, \infty), \quad d_3 : Z^2 \to [0, \infty) \tag{3.2}$$

as the distortion measures, which, in our case, is the squared distortion measure, and we let the average distortions be $\Delta_1 = E\left\{\frac{1}{n}\sum_{t=1}^{n} d_1(X_t, \hat{X}_t\right\}$ (similar expressions for the other sources). Then for given positive numbers $D_X, D_1, \ldots, D_n$, a rate $(n+1)$-tuplet $R_X, R_1, \ldots, R_n,$, is admissible if for any $\delta > 0, \quad n \geq n_0(\delta)$, there exists a *(n+2)-tuplet* $(\varphi_1, \varphi_2, \varphi_3, \psi) \in \Im_{n,\delta}(R_1, R_2, R_3)$, such that $\Delta_i \leq D_i + \delta, \quad i = 1,\ 2,\ 3.$

In our specific case, we do not care about the reproduction of the Y_i's, so the D_i's can be large. Rather, the Y_i's act as helpers to reproduce X by providing side information at the data fusion node. This is the so called n-helper case. Then for an encoding system using the Y_i,'s as n-helpers, the rate-distortion region given by:

$$\Re(D_X, D_1, \cdots D_n) = \{(R_X, R_1, \cdots R_n) \ : \ (R_X, R_1, \cdots R_n) \text{ is admissible}\} \tag{3.3}$$

for a given set of rates and distortion measures, is desired. For the special case of the correlated Gaussian sources, extending previous results [28, 17], we now state our main result for an admissible rate.

Theorem 3.1. We consider the following encoding functions:

$$\varphi_X : X^m \to \aleph_1 = \{1,\ \cdots,\ C_1\} \cdots \varphi_i : Y_i^m \to \aleph_i = \{1,\ \cdots,\ C_i\}$$

to be such that the rate constraints being satisfied are

$$\frac{1}{m} \log C_i \leq R_i + \delta, \quad i = X, 1, 2, \ldots n.$$

then for an admissible rate $(R_X, R_1, R_2, \cdots R_n)$, and for some $D_i's > 0$, the n-helper system data rates for correlated Gaussian sources can be fused to yield

an effective date rate (with respect to source X) satisfying the following lower bound

$$R_X \geq \frac{1}{2} \log \left\{ \frac{\sigma_X^2}{D_X} \cdot \left[\prod_{k=1}^{n} \left(1 - \rho_{XY_k}^2 + \rho_{XY_k}^2 \cdot 2^{-2R_k} \right) \right]^{\frac{1}{n}} \right\}. \tag{3.4}$$

This is the desired rate distortion region.

Proof:

The method of employing joint weakly δ-typical tuplet, based on typical sequences, is used in the proofs of the characterization, rather than a measure-theoretic approach [17]. We assume that an admissible set of rates exists, and we let

$$W_x = \varphi_X \left(X^n \right), \; W_i = \varphi_i \left(Y^n \right).$$

Then:

$$n(R_x + \delta) \geq \log(C_1) \geq H(W_1) \quad \geq \quad I(X^n; \hat{X}^n) - \frac{1}{n} \sum_{k=1}^{n} I(X^n; W_k) \tag{3.5}$$

$$n(R_i + \delta) \geq \log(C_i) \geq H(W_i) \quad \geq \quad I(Y_i^n; W_i) \tag{3.6}$$

Note that $W_i \to Y_i^n \to X^n$ forms a Markov Chain, thus defining:

$$F_n(D) \quad = \quad \inf_{\left\{ \hat{X}^n : \Delta_1 \leq D \right\}} \frac{1}{n} I(X^n; \hat{X}^n) \tag{3.7}$$

$$G_i(R) \quad = \quad \sup_{\left\{ \begin{array}{c} W_i : \frac{1}{n} I(Y_i^n; W_i) \leq R \\ W_i \to Y_i^n \to X^n \end{array} \right\}} \frac{1}{n} I(X^n; W_i) \tag{3.8}$$

we get:

$$R_X + \delta \geq F_n(D_X + \delta) - \frac{1}{n} \sum_{k=1}^{n} G_k(R_k + \delta) \tag{3.9}$$

By the Gaussian property of the sources, and concavity of the logarithm function, we get:

$$\tfrac{1}{n} I \left(X^n; \hat{X}^n \right) \geq \tfrac{1}{2} \log \left(\tfrac{\sigma_X^2}{D} \right) \quad \Rightarrow \quad F_n \left(D \right) \geq \tfrac{1}{2} \log \left(\tfrac{\sigma_X^2}{D} \right) \tag{3.10}$$

Finally, an upper bound of $G_i(R)$ is obtained by an extension of the technique for the two node case (entropy of the power inequality, monotonicity, Jensen's inequality):

$$\frac{1}{n} I(X^n; W_i) \leq \frac{1}{2} \log \left(\frac{1}{1 - \rho_{XY_i}^2 + \rho_{XY_i}^2 \cdot 2^{-2R}} \right) \tag{3.11}$$

Substituting the bounds in the expression for $F_n(D)$, we obtain:

$$\begin{aligned} R_X + \delta &\geq F_n(D_X + \delta) - \frac{1}{n} \sum_{k=1}^{n} G_k(R_k + \delta) \\ &\geq \frac{1}{2} \log \left(\frac{\sigma_X^2}{D_1 + \delta} \right) + \frac{1}{2} \sum_{k=1}^{n} \log \left(1 - \rho_{XY_k}^2 + \rho_{XY_k}^2 \cdot 2^{-2(R_k + \delta)} \right)^{\frac{1}{n}} \end{aligned} \tag{3.12}$$

Letting $\delta \to 0$, we obtain the final result:

$$R_X \geq \frac{1}{2} \log \left\{ \frac{\sigma_X^2}{D_X} \cdot \left[\prod_{k=1}^{n} \left(1 - \rho_{XY_k}^2 + \rho_{XY_k}^2 \cdot 2^{-2R_k} \right) \right]^{\frac{1}{n}} \right\}. \tag{3.13}$$

3.1.2 Conclusion

In this section, we have considered the rate distortion problem for a sensor network employing data fusion at a node. The main assumption has been that the n-helper nodes are all producing correlated Gaussian data, which has then enabled us to obtain an analytic form for the rate distortion bound. The primary utility of this result is to compare practical data fusion schemes with the predicted bounds—in particular, to determine which are the most critical and sensitive parameters affecting rate and performance of a data fusion scheme in a network sensor system.

The main limitation of the work is that currently, extensions to the more 'real-world' scenario of non-Gaussian sources and channels is not obvious. However, efforts are underway to formulate numerically solvable versions for the more realistic scenarios. Our ultimate goal is, in the absence of any tractable analytical expression, to obtain at least an iterative algorithm, or a convex optimization

form. Ideally, this would allow us to accurately predict maximum rate/ minimum distortion pairs for a wide variety of channels and sources for random arrangements of sensors. We would also like to be able to investigate various 'what-if" scenarios in simulation set-ups for particular types of configurations and coding implementations.

Thus, we hope eventually to be able to use these bounds to definitively compare various data fusion and network communication schemes for wireless sensor networks, with regards to their performance and efficiency.

3.2 Asymptotic Delay in Random Wireless Networks

3.2.1 Introduction

As mentioned earlier, ad hoc networks are self-governing, self-organizing, distributed networks consisting of portable computers, or hosts, that communicate via a common wireless channel [32]. Their main advantage is that they can be rapidly deployed in situations where no wireless communication infrastructure exits, or where such an infrastructure is difficult or expensive to implement.

Such networks are of great practical and theoretical interest. They are of practical interest precisely because the future of digital communications will undoubtedly be wireless, at least the "last mile" links, and most likely peer-to-peer, in an ad hoc fashion. They are of theoretical interest, because the rigorous analysis of the performance issues for wireless networks is still not complete. There has been a large amount of research conducted regarding the information theory of multi-user channels, However, many open problems still remain in the field of network information theory, and the issues regarding performance bounds for delay, system capacity, achievable rates, suitable codes etc. are chief among them.

As pointed out in [19], a satisfactory union among the two disciplines is not yet complete.

One recent study that has gone a long way towards that goal is [31], in which the authors determined the uniform achievable rates for nodes in an asymptotically large ad hoc network, as well as several bounds for the capacity of the network under different protocols. However, the problem of determining the"average delay" in such arbitrary networks still has not been tackled in a general manner. This is of interest since in practical networks, bounds on delay have a direct relationship to the quality of service that large networks can expect to deliver. To the designer of such networks, it can provide insight as to the limits of performance and what we can expect from such networks, and how the myriad of currently available protocols scale to the theoretical limit.

With regards to prior art, in [15, 16], the author studies the delay characteristics of specific network elements which can be interconnected to form complex networks. However, this yields tractable analysis only when the input traffic characteristics obey certain burstiness properties, and cannot be readily applied to the wireless case, or used to compute an overall 'average' delay. In the studies done in [3] and [91], the authors look specifically at queuing and its impact on information flow in a network, but once again, the analysis does not lend itself easily to a general analysis of the delay in a mobile ad hoc network. The techniques are elegant, but further research is required to determine if they can be modified to apply in a straightforward manner to the mobile wireless case.

In this section, we instead undertake a simplified study of this problem and attempt to formulate a very simple bound for a restricted scenario. We propose to use the results of [31] and a probability result derived in [61] and attempt to extract a first order bound. Specifically, we consider the average delay that

bits generated by arbitrary nodes in the network can expect to experience while being routed from source to destination, as the nodes in the network increase asymptotically. The following section contains a brief description of the analysis.

3.2.2 Analysis

We consider the same scenario as the author in [61], some details of which are reproduced below. n nodes are randomly distributed over an unbounded area. The x and y coordinates of the node locations are assumed to have zero-mean Gaussian distributions, for the purposes of analytical derivation. Then assuming that the transmission range of each mobile is R, and the Gaussian distribution has variance σ^2, then the probability density function (pdf) of the link distance between two arbitrary nodes is given by:

$$p_r(r) = \frac{r}{2\sigma^2} e^{-r^2/4\sigma^2}. \tag{3.14}$$

The probability of a 2-hop connection between an arbitrary source and destination pair, separated by a distance r, is then given by:

$$\begin{aligned} P_2 &= \mathrm{P}\{1 \to 2\} \quad \text{in 2 hops} \\ &= \mathrm{P}\{R < r < 2R\} \quad \textit{and} \text{ at least 1 other node in area of intersection} \\ &= \int dx_1 \int dy_1 \int dx_2 \int dy_2 \, \mathrm{p}_{x,y}(x_1, y_1, x_2, y_2) \\ &\times \left[1 - \left[1 - \int dx_3 \int dy_3 \mathrm{p}_{x,y}(x_3, y_3) \right]^{n-2} \right] \end{aligned} \tag{3.15}$$

As n approaches infinity, the expression given above can be approximated with the following upper bound (derivation in [61]):

$$P_2 < \int_{R^2/4\sigma^2}^{R^2/\sigma^2} e^{-\nu}(1-0)\, d\nu \tag{3.16}$$

which simplifies to $(e^{-R^2/4\sigma^2} - e^{-R^2/\sigma^2}) \triangleq P_{2\infty}$. It can then be shown by a similar argument that the asymptotic probability of an m-hop connection is given by:

$$P_m < e^{-(m-1)^2R^2/4\sigma^2} - e^{-m^2R^2/4\sigma^2} \triangleq P_{m\infty} \tag{3.17}$$

and therefore an upper bound on the average number of hops between the node pairs can be calculated from:

$$\mathrm{E}\{h\} = \sum_{m=1}^{n-1} mP_m < \sum_{m=1}^{\infty} mP_{m\infty}. \tag{3.18}$$

In [61], it is seen by numerical analysis that this expression converges to a nearly linear function of the ratio of node dispersion parameter (standard deviation of the Gaussian) to transmission range. Thus, using non-linear regression techniques, and subsequent linearization, we are left with the following remarkably simple approximation for the asymptotic average number of hops that an 'average' data bit takes from and 'average' source to its intended destination node, for an arbitrary wireless network with Gaussian (spatially) distributed nodes:

$$\mathrm{E}\{h\} < 0.5 + 1.772(\sigma/R). \tag{3.19}$$

In terms of actual distance, we can simply multiply by the transmission range of each node R, to obtain the average hop distance per transmitted bit. Note that this result is not limited by any specific routing, access, modulation protocol. We can now utilize this result and the asymptotic capacity results for the Physical Model (arbitrary) network as in [31], with some appropriate modifications regarding domain geometry. It has been shown in [31] that the *transport capacity* of such an arbitrary network of n nodes scattered in a disk of unit area is of the order $O\left(W\sqrt{n}\,\right)$ bit-meters per second (which is equal to on the order

of $O\left(W/\sqrt{n}\right)$ bit-meters per second *per* node). Since the average hop distance given in 3.19 is for an unbounded disk, we make the following observation. The nodes in the scenario described earlier are zero-mean Gaussian distributed, therefore, 99.7% of all the nodes are expected to lie within a 3σ radius of the center. The resulting area of the disk of radius 3σ disk is $9\pi\sigma^2$. We can therefore scale the results in [31] by this factor.

Another issue is the fact that the capacity results are cited for uniform distribution over the unit area disk, whereas the hop distance calculations were done for a Gaussian distribution. Once again, in the limit of n approaching very large values, we can appeal to the central limit theorem regarding node separation distribution for the two cases and equate them as a first order approximation for asymptotic analysis.

Thus, we finally have a first order approximation for the expected delay a bit experiences in an arbitrary wireless network.

$$\begin{aligned} D &\sim O\left\{R\left(0.5+\frac{1.772\sigma}{R}\right)\Big/\frac{9\pi\sigma^2 W}{\sqrt{n}}\right\} \\ &= O\left\{\frac{\sqrt{n}}{W\sigma^2}\cdot(0.056\pi R+0.197\pi)\right\} \end{aligned} \tag{3.20}$$

where R is the transmission radius of a representative node, σ^2 is the variance of each of the x and y coordinate of the Gaussian dispersion, and W is the feasible transmit rate of each node. Since we are interested in the asymptotic relationship of the delay to the network parameters, we can further ignore all the constant factors in the expression above, and are thus left with the following order expression for the 'average' delay experienced by a random bit in the network:

$$D \sim O\left\{\frac{R}{W\sigma^2}\cdot\sqrt{n}\right\} \tag{3.21}$$

As a function of the network size alone, we obtain:

$$D \sim O\left(\sqrt{n}\right) \tag{3.22}$$

3.2.3 Concluding Remarks

The scenario as discussed above is somewhat limited in scope since the dispersion pattern was fixed to be Gaussian and node capabilities were limited since the impact of mobility was not considered. Additionally, the results we relied upon to derive the order expression for the delay assumes a globally optimal scheme for the routing, medium access, etc., which may not realistic. However, these assumptions do yield insight and a simple analysis into the typical delay characteristic of bits in the network. A future research direction is to relax some of the assumptions and observing where/if the analysis become intractable, and what alternative strategies can be applied, e.g. by incorporating some queuing results from [3, 91].

CHAPTER 4

Gateway Optimization for Connectivity in Heterogeneous Multi-Tiered Wireless Networks

In this chapter, the hypothetical data fusion architectures discussed briefly in Chapter 2 are considered in the context of some real-life, large-scale networks. Practical issues, which were referred to as *sensor organization* earlier (Section 2.1.1), we now investigate in terms of performance and associated costs. In particular, we note the emergence of mobile ad hoc network (MANET) technology, which allows practical deployment of many of the decentralized information processing concepts that have been researched over the last few decades. In this chapter, we focus on a narrow subset of those concerns: specifically, the issue of facilitating scalability and range extension in partitioned mobile ad hoc networks.

We propose that to facilitate scalability in and to provide connectivity between partitions that might occur in wireless networks as a consequence of mobility, we can envision a 'range extension' network that consists of airborne communication platforms, and satellites. These airborne or satellite nodes will maintain communication links with specific 'gateway' nodes among the mobile ground nodes. To communicate with a node that is geographically distant or belongs to a different network partition, a node can relay its data packets through an appropriate mobile gateway and via the range extension network. If we envision that the MANET is divided into different groups and a mobile gateway is deployed for

each such group, an objective then, is to determine the trajectory of the mobile gateway to best serve the ad hoc group to which it belongs, in terms of network performance metrics such as throughput and latency. In this chapter, this problem of computing the optimal position for a gateway is reduced to a linear optimization problem by means of some simplifying but realistic assumptions. We suggest methods that may be deployed to enable the gateway to follow this optimal trajectory as closely as possible (within the practical constraints imposed by its velocity and maneuverability). Simulation results for various scenarios show a 10-15% improvement in the throughput and in latency (per group containing a gateway) if a gateway has a dynamic trajectory whose locus follows the computed optimal position, as compared to a gateway that is statically placed at a regular position, or to a gateway that has a random trajectory.

4.1 Introduction

Mobile ad hoc networking technology [32] may be appropriate for linking mobile computers in an office or home environment, deploying wireless sensors in remote or inhospitable terrain, coordinating disaster relief efforts after natural catastrophes, or in tactical deployments for situation awareness applications [51]. A major challenge in the wide deployment of MANETs has been in achieving scalability. Furthermore, due to the range limitations of the nodes in the ad hoc network, the network might often be divided into isolated partitions. In order to achieve scalability in terms of efficient communications between geographically distant nodes or between nodes that belong to different isolated partitions (each of which is an ad hoc group by itself), it is desirable to provide a supporting infrastructure in the form of a range extension network. This infrastructure is also essential to interface the MANET with the Internet.

This range extension network could typically consist of airborne relay nodes or low earth orbit/geostationary satellites. In order to interface the ad hoc network with the range extension network, one can envision the deployment of special gateway nodes in the ad hoc network. These are 'on-ground' nodes that might be more power/processing capable than the other ad hoc nodes on the ground and are equipped with the appropriate hardware for communicating with the satellite/airborne nodes. This architecture can therefore be visualized to consist of two layers. The first layer includes the ad hoc network, and the second includes the range extension network consisting of satellites or airborne nodes. The mobile gateway provides the interface for the communications between the two layers and hence we shall call a gateway node a "Cross Layer Communication Agent" or CCA from this point onwards. Similar architectures have been previously considered for enabling hierarchical routing or multicasting [44, 2]. In [44], and [2], topological information is used to build spanning trees rooted at convenient nodes in the MANET. In [43], the authors have investigated the feasibility of partitioning MANETs into groups with cluster-heads or repositories of information to enable efficient information dissemination [43]. In contrast, our objective is the determination of the CCA trajectory that a mobile gateway or CCA has to follow in order to optimize inter-domain network performance.

A CCA may be assigned to a group of ad hoc nodes or it might be affiliated with a geographical domain and placed statically at the center of the domain. In the former case, it links the range extension network with the specific group of ad hoc nodes. In the latter case, it provides the ad hoc nodes within the specific geographical domain with a link to the range extension network. If the CCA is mobile and is affiliated with an ad hoc team of mobile units, the question arises as to where the CCA ought to be located relative to the mobile units. In other words, the objective is to specify the locus of the position that a mobile CCA

is to follow and the rules that govern this trajectory with respect to the relative motion of the other mobile units. We design a methodology for defining the CCA trajectory based upon the location, loading, etc. of the other mobile nodes in the ad hoc group that the CCA serves. We show that network performance improves (for communication involving nodes in different clusters of nodes), in terms of throughput and latency, if the CCA trajectory is computed based on our methodology.

We derive a relatively simple analytic formulation for the optimal CCA position, which is equivalent to a linear optimization problem. This is discussed in Section 4.3. We also provide an algorithmic implementation of the formulation, and discuss the effect of some of the parameters in this section. Various scenarios are considered for deliberation. In Section 4.4, we estimate the overhead for implementing this architecture with the aid of typically used media access control (MAC) and routing protocols. We also investigate the computational complexity of our CCA trajectory definition algorithm. In Section 4.5, we discuss our simulation framework and discuss the results. We conclude in the final section.

4.2 System Model

We motivate the discussion of the system architecture that we laid out in the previous section in the context of the following scenario. Consider separate groups of mobile ad hoc nodes operating in a terrain with blockages and deployment area restrictions (e.g. troop divisions deployed in a mountainous area). Each group would have one or more CCAs capable of communicating with an airborne or satellite node with which it has a direct line of sight connection. As an example, in Figure 4.1, we have considered two isolated groups of mobile nodes, each forming a MANET by themselves, and each having its own CCA. The ad hoc

group of nodes that use a particular CCA to communicate via the range extension network are said to belong to that CCA's *domain*. The CCA in each group is then the conduit via which the ad hoc nodes in the separate groups can send data packets to each other, with the routing assistance of the airborne node. The airborne node can also serve to connect the clusters to a wired infrastructure (e.g. command and control centers outside the theater of operations).

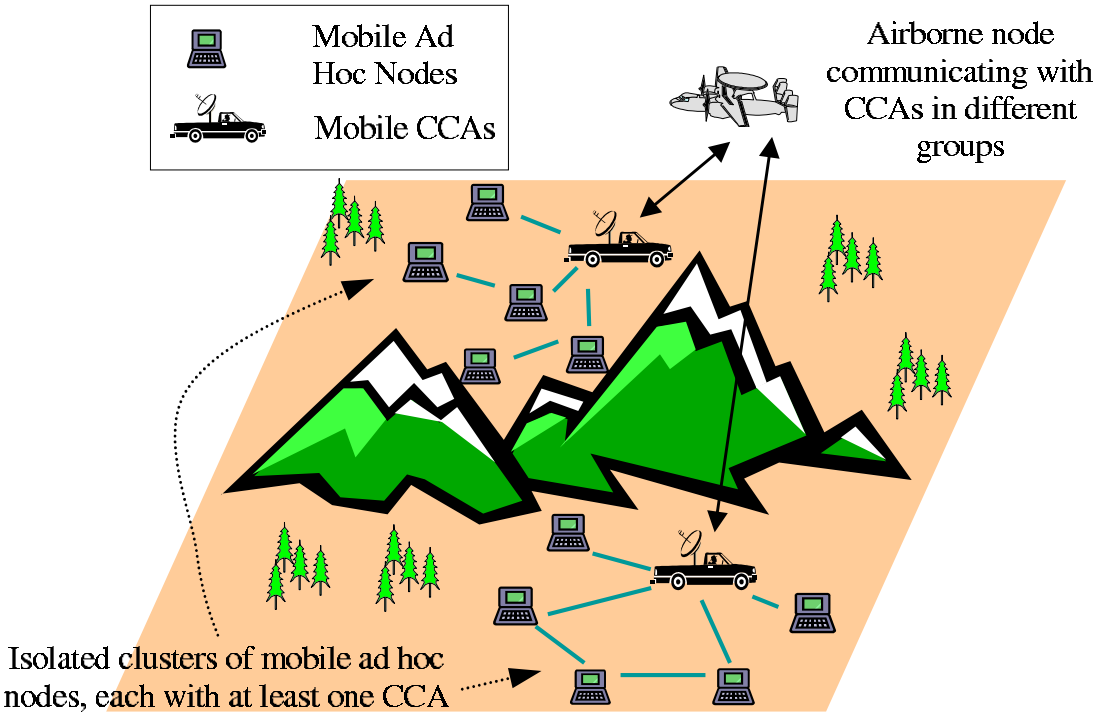

Figure 4.1: Ad hoc network of two groups of mobile nodes and CCAs.

As the nodes in a particular group move, the objective then is to specify the 'optimal' trajectory for the mobile ground CCA associated with that group. For communication intended for nodes within a given group, the nodes would not be compelled to use the CCA, but would instead rely on the underlying MANET architecture using traditional routing, MAC protocols, etc. By intelligently positioning the CCA, we might expect to achieve better network performance for

inter-domain node communications, (i.e. data communication between nodes that are in the domains of different, possibly geographically isolated CCAs) than if the CCA were allowed to move randomly with respect to the nodes it its domain.

We can assume any suitable mobility model for depicting the motion of the ad hoc nodes [37]. Thus, some of the nodes may be statically deployed (sensor nodes) or, the nodes may move according to a random waypoint model [41]; or, they may move according to a model in which they approximately follow one particular node that is considered the 'leader', etc.

The terrain in which the MANETs are deployed may contain regions where either the node, or the CCA, or both, cannot reside. These can be regions where, e.g., channel impediments create radio nulls, or hazard zones which expose the CCA or specific nodes to harm (in tactical deployments), or where the terrain is inaccessible, etc. We refer to these regions as *blockages*.

The performance metrics that could potentially be improved by optimizing the CCA trajectory include, but may not be limited to, the following:

- *inter-domain* network data throughput
- *inter-domain* network packet transport delay
- total power expended (or maximum power consumed per packet/bit per node in the CCA)
- data transmission reliability (packet drop/error rate)
- volume of the network control messages and resulting signaling overhead.

The procedure by which the weighted centroid is computed requires the following:

1. Each node is equipped with a GPS device that enables the node to determine its position.

2. Each node can estimate its offered load in real time

3. Terrain information (such as specific coordinates or boundaries of radio null regions where radio signals cannot propagate, inaccessible terrain, etc.) is available at each CCA. This can easily be made available at network inception[1].

The details of the actual communication mechanisms that enable the MANET to function are not directly relevant in the development of our analytical formulation for computing the optimum trajectory that our CCA ought to follow (Section 4.3). For ***intra***-domain node communications (i.e. communication between nodes that are in the domain of the same CCA), the MANET could rely on well-established protocols such as the IEEE 802.11 MAC protocol for media access control [54], DSR, DSDV, or AODV for routing [41, 69], etc. to establish and maintain connectivity. For ***inter***-domain node communications, data will have to be routed through the CCA and via the range extension network.

4.3 CCA Trajectory Update Algorithm: Formulation and Analysis

In this section we describe the algorithm for determining the trajectory of mobile CCAs such that it is optimal in terms of 'relative position' with respect to the group of ad hoc nodes that it serves. By having the CCA follow the trajectory

[1]Position based schemes have previously been suggested and studied for ad hoc networks, [44], [2].

determined by this algorithm, we expect (and later show by means of simulations) that we will achieve a significant improvement in network performance, in comparison with a scheme that has static CCAs or has the CCAs follow a trajectory defined by a random way-point model. We describe our algorithm assuming that there is single CCA per domain in Section 4.3.1. However, it is possible for several nodes among a cluster of nodes to be capable of communications with the range extension network and hence any of these nodes could assume the role of a CCA. Fortunately, in all these cases, the base algorithm that we propose remains unchanged. Increased layers of complexity can be added to the base algorithm to enable the CCAs to participate in node 'hand-off' as in cellular networks, or to intelligently share the load generated by the nodes in the overlapping regions of intersecting domains. These features may be exploited to achieve further performance improvements. A brief discussion of the possibilities is presented in Section 4.3.2.1.

4.3.1 Node Domains Containing a Single CCA

As mentioned earlier, one might expect that the performance of the network would be best if we could optimally position the CCA within the group of ad hoc nodes that it belongs to. We refer to this optimal position as the *weighted geographic centroid*. This section describes the procedure that computes this weighted centroid. We formulate an optimization problem that the CCA solves periodically with the help of information that it has gathered from the other ad hoc nodes, to determine its trajectory. The parameters that the CCA can take into account in formulating the optimization problem could include node positions, each node's offered load, data traffic patterns, priority of the generated traffic, the channel signal to interference noise ratio (SIR), among others. The

choice of which parameters to include is determined by the specific network metrics (listed in Section 4.2) that are of importance. We can easily have explicit formulations for every desired optimization objective. As we shall see, the optimization formulations could essentially be represented as either linear or convex programs [9]. For the purposes of this discussion, we consider positions and their offered load as our primary parameters and the network throughput and the average delay experienced by inter-domain data packets as our basic performance metrics.

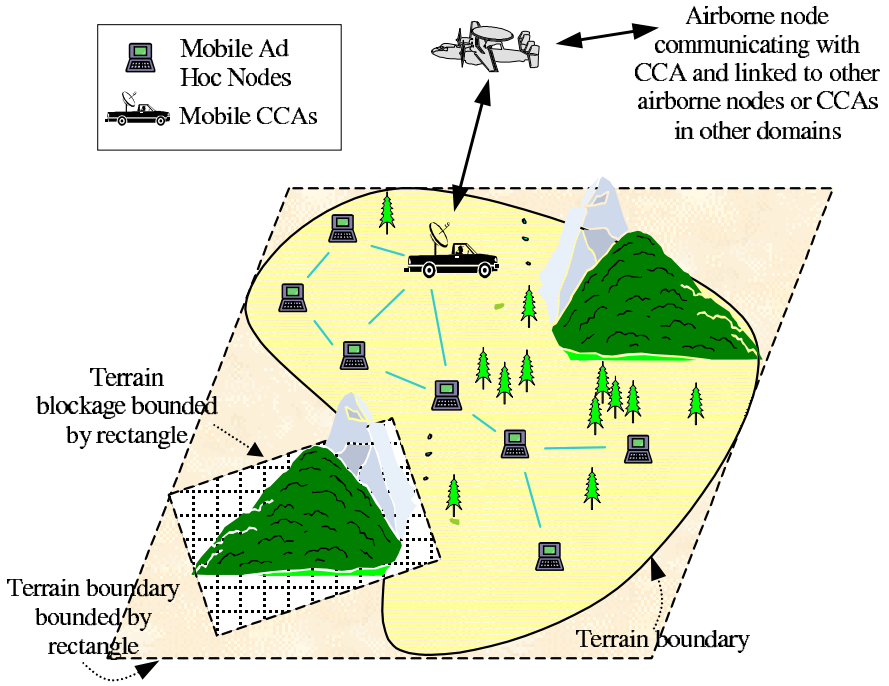

Figure 4.2: CCA domain with location bounds.

Figure 4.2 represents a typical scenario showing ad hoc groups with a single CCA per group. As shown in the figure, terrain contains blockages (described in Section 4.2) that can always be represented by simple rectangular regions. In particular, each terrain blockage can be bounded by the smallest rectangular region that contains that blockage. If we further assume that the ad hoc group

is deployed in a finite area which represents the 'domain' of a CCA, then the area of deployment may also be similarly bounded by a rectangular region. This representation is useful since the constraints that govern the position of a CCA can then be described by simple linear equations (based on the coordinates of the rectangle boundaries). Note that it is extremely hard if not impossible to characterize the arbitrary geometries that these regions might have by exact mathematical expressions. Thus, our model for the region over which the CCA can move consists of a rectangular area with smaller rectangular regions marked as 'no-go' areas.

A fundamental characteristic of MANETs, (also shown in Figure 4.2) is that, not all the nodes in the domain have a direct link to the CCA. Some nodes will be outside the single-hop radio range of the CCA and will have to route their data packets to the CCA via multiple hops through other ad hoc nodes[2]. The cost function that we use in computing the weighted geographic centroid takes the offered load of the individual nodes and the priorities of the packets generated at each node into account. This ensures that the CCA is closest to the most heavily loaded nodes, or to the nodes that generate packets of the highest priority, as the requirement might be. Thus, when all packets are of the same priority, the CCA is positioned such that most of the data packets reach the CCA in a single hop, thereby ensuring a better utilization of the available resources. In the case wherein nodes generate packets of different priorities, the CCA's position would ensure that most of the higher priority packets would reach the CCA in a single hop.

[2]These data packets are destined for nodes in other domains and thus *must* be routed through the CCA to the range extension network.

4.3.1.1 Optimization Formulation

We assume that the CCA acquires the coordinates of each node in its domain, that needs to send inter-domain data packets[3]. The weighted centroid is simply a factored distance norm, and the constraints are linear inequalities. For a domain with n nodes, the optimization problem can be formulated as:

$$\text{minimize} \quad \sum_{i=1}^{n} f(\tau_i, \rho_i) \cdot |\mathbf{x}_0 - \mathbf{x}_i| \tag{4.1}$$

$$\text{subject to} \quad \mathbf{w}_1 \leq \mathbf{x}_0 \leq \mathbf{w}_2 \tag{4.2}$$

$$\mathbf{x}_0 \geq \mathbf{b}_{2k} \tag{4.3}$$

$$\mathbf{x}_0 \leq \mathbf{b}_{1k} \tag{4.4}$$

The optimization variable in the norm minimization expression, Equation. 4.1, is the CCA position, represented by the 2-D (ground) position vector, $\mathbf{x}_0$, with reference to any suitable origin in the terrain of interest. The other $\mathbf{x}_i$'s represent the coordinate vectors of the mobile nodes with respect to the same origin. The values of $\mathbf{x}_i$'s are obtained at each sampling instant and a new value of $\mathbf{x}_0$ is computed. The weighting factor, $f(\rho_i, \tau_i)$ is a user defined function that depends on the i^{th} node's load, ρ_i, and priority, τ_i. Note that when we use the term load, we in fact refer to the data traffic that the i^{th} node wants to send to nodes in other domains. We do not consider the data load due to intra-domain communication among the nodes, although this may affect the available bandwidth for inter-domain communications[4]. Depending on the type of traffic being generated by the nodes, the function $f(\rho_i, \tau_i)$ can be defined appropriately to reflect CBR, or

[3]We will discuss the technique by which this acquisition is realized and the resulting overhead incurred involved in Section 4.4.

[4]Alternatively, we can assume that separate channels exist for intra-domain and inter-domain communications.

variable bit rate (VBR) traffic and with or without defined priorities. The terms $\mathbf{w}_i$ are the vector coordinates representing the outer rectangle circumscribing the domain (bottom left and the top right points), and $\mathbf{b}_{ik}$'s are similar vectors that represent the boundary of the k^{th} blockage. Therefore, we are minimizing the sum of the weighted geometric distance from the CCA to each of the nodes, subject to the boundary and blockage constraints.

As stated earlier, this problem is a non-linear optimization problem [8]. However, if we ensure that the cost function is the $\mathcal{L}^2$ norm, since the constraints are linear, the problem is a convex program (all norm functions are convex). This problem, therefore, has a global optimum and any of the standard convex optimization algorithms can be used to find the optimal position, $\mathbf{x}_0$ for the CCA (e.g. steepest descent, Newton's methods, etc. [8], [9])[5]. Furthermore, the convex optimization problem itself can be easily transformed into a simple linear program (LP) and the cost function can be replaced by an equivalent linear cost function. The resulting LP can then be solved far more efficiently via modern interior point methods [59]. An implementation would require the CCA to perform the **CCA Trajectory Update Algorithm** which is as follows.

Algorithm 4.1. CCA Trajectory Update Algorithm

- *Input constants (set 'a priori')*: terrain and blockage boundaries, sampling times, optimization metric of interest.
- *Output*: optimum CCA location computed at specific sampling times.
- {*While nodes in the domain have inter-domain data packets to send*}, **DO**:

[5]Since the problem is formulated as a convex program, the solution is found by numerical methods.

1. Collect or estimate the position of each node, $\mathbf{x}_i$, at each sampling instant.
2. Collect from each node, an estimate of its current load and the priority that it desires[6].
3. Perform a local computation to solve the LP equivalent to the optimization problem in Equation 4.1 and obtain optimum CCA location for that sampling instant.
4. Move towards the optimal location in the most suitable manner, as allowed by the physical constraints[7].
5. REPEAT (at next sampling instant)

The motion of the CCA can be further governed by certain rules to prevent race conditions and such. As an example, one can have a hysteresis rule that helps to prevent excessive CCA sensitivity, wherein a computed 'new' CCA location has be greater than a minimum of some pre-specified δ units from the present location before we decide to move the CCA. Another hysteresis rule might require the CCA to remain at a newly computed location that it has moved to, for a specific time period (usually several frame update intervals).

4.3.2 Overlapping CCA Domains

In the methodology discussed so far, we have only considered domains with a single CCA serving a group of ad hoc nodes. In this section, we consider a case in which ad hoc nodes have the ability to chose one CCA among a set of CCAs. Without loss of generality, we can think of this scenario to be equivalent

[6]The priority is determined by policy and is not discussed further.

[7]The problem of navigating from one location to another, on a 2-dimensional surface with obstacles with no pre-established paths, is a vast area of research in robotics and is not discussed in this thesis [64].

to the case of two or more domains with single CCAs that intersect due to their proximity. A node that lies in the overlapping region of the intersecting domains can choose any of the CCAs to relay its inter-domain traffic to the range extension network.

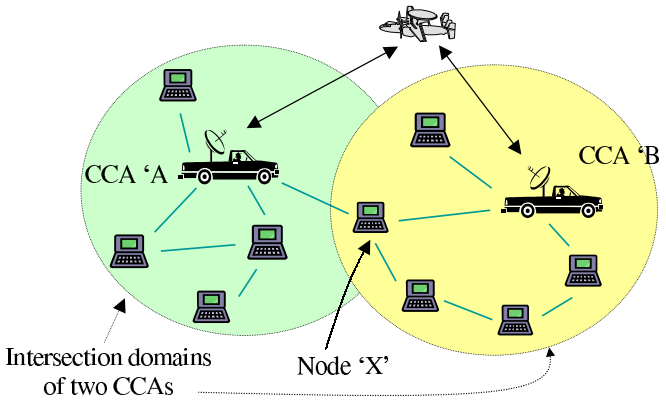

Figure 4.3: Overlapping CCA domains

In Figure 4.3, each CCA has a specific domain space defined by a geographical area (the circles enclosing the CCAs A and B) and serves a set of ad hoc nodes that are confined to this region. Although the specific motion of the individual nodes may themselves be essentially random, they remain affiliated with, and within the boundary of the domain of a particular CCA. When two or more such independent domains intersect, then one or more of the ad hoc nodes may lie within the domains of multiple CCAs. In Figure 4.3 node X was originally affiliated with CCA A, but is now also in the domain of CCA B. It is now possible for node X to communicate with CCA B (appropriate signalling methods will be required but are not discussed further). In fact node X might want to migrate from the domain of CCA A to that of CCA B, i.e. it might wish to switch its affiliation. In such a case, node X will indicate its intent to CCA B. Then one of the following is possible:

- CCA B may completely ignore this message from node X. In this case, its trajectory remains unaffected by node X's message.

- CCA B may agree to relay the data message from/to node X to/from the range extension network. However, it may not use the control information from node X (that reflect the position of node X, its offered load, etc.) in determining its trajectory.

- CCA B may consider node X to temporarily belong to its domain. In this case, not only does CCA B act as node X's interface to the range extension network, but it also takes the position, the offered load, and other parameters associated with node X into account while determining its trajectory.

Node X might choose to affiliate with both CCAs A and B. In that case it transmits an intent message to CCA B as in the previous case and CCA B could react in accordance to the policy defined in one of the earlier cases, listed above. If node X is granted partial affiliation by CCA B, it can distribute its inter-domain load between CCA A and CCA B. Alternatively, the CCAs may elect to 'handoff' selective nodes to each other, as in cellular networks[8]. Thus, one might in this case, be able to uniformly balance the inter-domain load in the network between the CCAs, to the extent possible. This feature and its effect on the optimal CCA trajectory are described in Section 4.3.2.1.

For each of the cases above, a different optimization rule may be formulated, and the choice of the rule to use is application specific. However the crucial point to note is that in spite of this added complexity, the CCAs can still use the same convex programming formulation that was presented earlier.

[8]We are assuming that this type of CCA to CCA cooperation can be efficiently done since they will both be one hop away from a common aerial/satellite node.

4.3.2.1 Balancing Load Between CCAs

We consider again, the scenario shown in Figure 4.3 as an example. Our objective is to have the nodes that are in the overlapping regions of the intersection of multiple CCA domains choose an appropriate CCA so as to evenly distribute the offered load between the individual domains. We reiterate that by 'load' we refer to the offered data load that is generated due to the packets that are to be transported via the range extension network.

We assume that the aggregate offered loads at CCA A and B are ρ_A and ρ_B packets per unit time, respectively, and that node X offers a data load of ρ. Node X can now assist in load balancing by directing the data packets that it either generates or relays to the appropriate CCA, such that the loads in the two domains are as close to each other as possible. Thus, if $\rho_A > \beta\rho_B$ where β is some preset threshold, then node X can be instructed to route a fraction α, $0 \leq \alpha \leq 1$, of its load to CCA B, while the remaining load $(1-\alpha)\rho$ is routed to CCA A. When the two CCA domains have their loads balanced, we essentially have:

$$\rho_A - \alpha\rho \approx \rho_B + (1-\alpha)\rho \tag{4.5}$$

If node X does not generate enough data packets to satisfy the equation, then it can route all its packets to CCA B ($\alpha = 1$) for the duration that it remains in the overlapping region of the domains of two CCAs. Note that in this discussion, we assume that the nodes that belong to an ad hoc group will have to stay with that group. Thus only a temporary switch in CCA affiliation is permitted. However, it is easy to extend the method to consider permanent migration of nodes from one CCA's domain to that of another.

The relatively simple load-balancing technique described in the previous paragraphs can be extended to include cases in which multiple nodes are in the over-

lapping regions of multiple intersecting domains. In that case, in order to prevent more than one node from simultaneously initiating load transfer, the CCAs can coordinate with each other to orchestrate the procedure and instruct the node that is most heavily loaded to attempt to split its load between the two domains. This procedure may then be iteratively repeated until the loads in the two domains are balanced.[9]

4.4 Computational Complexity and Overhead

In implementing our algorithm, it is essential that the incurred overhead in terms of the number of control message that are exchanged is small. In addition, the algorithm should be not be computationally expensive since the time taken by the CCA to compute the optimal location must be small as compared to the time taken by the CCA to move to that location. In this section, we provide estimates of the incurred overhead and discuss the computational complexity of our algorithm.

4.4.1 MAC Protocol, Routing Support and Overhead

For implementing step 3, two tasks are required. First, the identity of the CCA (e.g. an IP address or MAC address) has to be made known to all the nodes. This information can be software programmed into all the nodes before network deployment, or can be broadcast to all the nodes in the domain. This broadcast is, in fact, required if there are multiple CCAs in the domain, and they are operating in some cyclic order for specified periods of time. Alternatively, each node could obtain the address of the CCA on a reactive basis, that is, if and

[9]This procedure has been simulated and the results are discussed in Section 4.5.

when they have inter-domain packets to send. The overhead in this case is about the same as in discovering a specific node within a MANET.

Second, the CCA has to obtain state information from all the nodes. If we consider the Open Systems Interface (OSI) or the TCP/IP layered architecture, the CCA Trajectory Algorithm can be considered as a program running at the network layer. It requires the CCA's network layer to obtain the coordinates, the offered load, and other parameters from all the other nodes in its domain. To do so, the CCA would need to broadcast a *global query message* to all the nodes in its domain, once every sampling period.

If there are n mobile nodes in the network, then a 'proactive' scheme requires that these nodes transmit an update every sampling period. Then number of routing messages in the worst case is on the order of $O(n^2)$ per sampling period (assuming flooding is employed to transport these messages) [43]. However, if the nodes that have inter-domain data to send constitute only a small fraction of the total number of nodes in the network, say α, then the number of transmissions will be reduced by this factor to $\alpha \cdot n^2$, which is again $O(n^2)$. By intelligently using the routing tables in order to relay control information, this overhead can be further reduced.

The sampling period is dependent on the rate at which the topology of the network changes. This rate depends on the density of the network and the velocity of the nodes in the network. If a table driven routing protocol is used, routing updates are required to be disseminated in order to cope with the changes in topology. The control information that is required for trajectory control may simply be 'piggy-backed' onto the routing update messages[10]. In fact this control information may be embedded in the MAC layer 'hello' or even piggy-backed onto

[10]Most of these parameters are usually single 8-bit or 16-bit numbers, per node, so payload data length is not an issue.

the data payload that is routed to the CCA. The CCA would then, appropriately, extract the appropriate control information. This makes the overhead required for gathering state information for the algorithm to work to be essentially the same as, or marginally incremental to the overhead required for enabling the underlying routing and MAC protocols. For the overhead incurred in deploying various MANET protocols, the reader is referred to [79]. We note here that in network simulations, we modified the standard Destination-Sequenced Distance-Vector Routing (DSDV) protocol [68] by piggy-backing appropriate control information onto routing update messages.

4.4.2 Optimization Complexity

It is important that the optimization algorithms that we describe in Section 4.3.1.1 *not* be computationally intensive, since our objective is to administer trajectory control of the CCA in real-time. In other words, the time that it takes for the CCA to compute its new position based on the data that it gathers during a sampling period should be less than the sampling period itself. The computation of the trajectory involves solving a linear program numerically. It is well known that modern interior point LP solvers have a worst-case performance of $O(n^3)$ [9], where n is the number of variables in the LP. In our formulation, n would correspond to the number of nodes in the network. Thus, for a network of 100 nodes, each of which is assumed to generate inter-domain data packets, we would expect calculations on the order of 100^3 or 1 million iterations per update period. A typical update period is of the order of 0.5 seconds in our simulations. Currently available 'off-the-shelf', inexpensive microprocessors can process on the order of tens of millions, or hundreds of millions of instructions per second [39]. For a network with 1000 nodes, the complexity increases significantly, requiring 1

billion iterations per update period. In such cases, however, it is far more efficient to simply deploy more CCAs, and thus sub-divide the larger domain into smaller domains of ad hoc networks.

To prevent a possible computational bottleneck at the CCA (which is burdened with both the inter-domain communications of the ad hoc network and the computation of the optimal position), a dedicated processor may be employed on each CCA. This would enable the optimization computations and the communication operations to proceed in parallel.

4.5 Simulation Framework and Results

In order to evaluate the performance of our trajectory control algorithm, we used the *ns-2* network simulator, release 2.1b6 [95] as our primary simulation platform. We also used Carnegie Mellon University's code extensions for supporting wireless ad hoc networks. We employed the following implementations that the code-extensions provide: (a) the link layer and the 802.11 MAC protocol modules; (b) CBR, VBR data sources; (c) the physical channel model, which is the two-ray radio propagation model where signals experience attenuation in accordance to a fourth power path loss model, and fading/shadowing effects are not included; (d) a random waypoint motion model to govern the motion of the mobile nodes. The supporting routing protocol that was used was DSDV [68]. We also conducted simulations with appropriate modifications to the Dynamic Source Routing (DSR) algorithm [77]. Our results were similar and we omit a discussion of this due to space limitations.

We created data structures that dynamically maintained the coordinates of each node and other control information, in the form required for a LP solver.

Next, the LP algorithm itself was integrated into the *ns-2* simulation framework by means of a C function call from the main program code. To implement the optimization algorithm, we used a modified version of PCx [58]. PCx is a linear program solver developed at the Optimization Technology Center at Argonne National Laboratory. To successfully integrate this module into our simulation, we incorporated a hook in the main *ns-2* code and passed the optimization parameters to the PCx LP solver. These parameters are the data sets that have been alluded to earlier (node positions, blockage locations, the load offered by each node, priority requested, etc.). With this information, at each sampling instance, PCx invokes a variant of Mehrotra's predictor-corrector algorithm [59] with the higher order correction strategy of Gondzio [30]. This approach is among the most effective methods currently known for solving linear programs.

We were interested only in the inter-domain networking performance and consequently, all the data packets that the nodes generated in each domain were always deterministically addressed to the CCA[11]. We assume that upon receiving these packets, the CCA delivers them to the range extension network by invoking a separate set of communication protocols and mechanisms. Furthermore, we have assumed that the CCA's trajectory is governed only by 'uplink' traffic when we factor in the offered load in our optimization formulation. It is fairly straightforward to extend the same methods to take the 'downlink traffic' into account while determining the CCA trajectory.

The first case that we consider is that of equally loaded nodes generating packets of the same priority. In this case, $f(\rho_i, \tau_i) = 1$, for all i, and the problem is now equivalent to minimizing the sum of the distances from the CCA to the nodes, subject to the boundary and blockage constraints.

[11]Intra-domain traffic in the MANET does not affect the CCA trajectory.

The results shown in Figure 4.4 through Figure 4.8 are for the following system parameters. We recall that a domain refers to the geographical area that bounds the realms of operation of a particular ad hoc group. We chose this domain to be a rectangular region of size 10,000 units by 10,000 units. The number of mobile ad hoc nodes per domain varies from 10 to 100; the nodes are assigned velocities chosen in accordance to a uniform distribution between a minimum of 0 units/s (stationary) to a maximum of 25 units/s, and move in line with the random waypoint model [41]. The CCA velocity is chosen to be at most one and a half times the maximum speed of the ad hoc nodes. All the nodes are equally loaded and generate traffic 50% of the time. The mobile nodes transmit their coordinates to the CCA once every 0.5 seconds, and the optimization calculations are also repeated with this frequency. The simulations are run for a total of 5000 seconds.

In Figure 4.4, simulation data was collected for three different scenarios: a CCA that is placed statically at the center of its domain; a CCA that is moving according to a random waypoint model; and finally, a CCA, the locus of whose trajectory is being updated using the optimization calculations discussed earlier. As expected, the results comparing the throughput in the three different cases show that the network throughput is the best when the CCA moves in accordance with the computed optimal trajectory.

Note that when our algorithm is implemented, the improvement in throughput is as high as 10% per domain (ignoring inter-domain interference effects). This is the improvement seen per single CCA domain. Since a region of interest may contain several domains, the overall throughput improvement can be very significant for the network as whole.

The advantage of the dynamic CCA placement technique is much more evi-

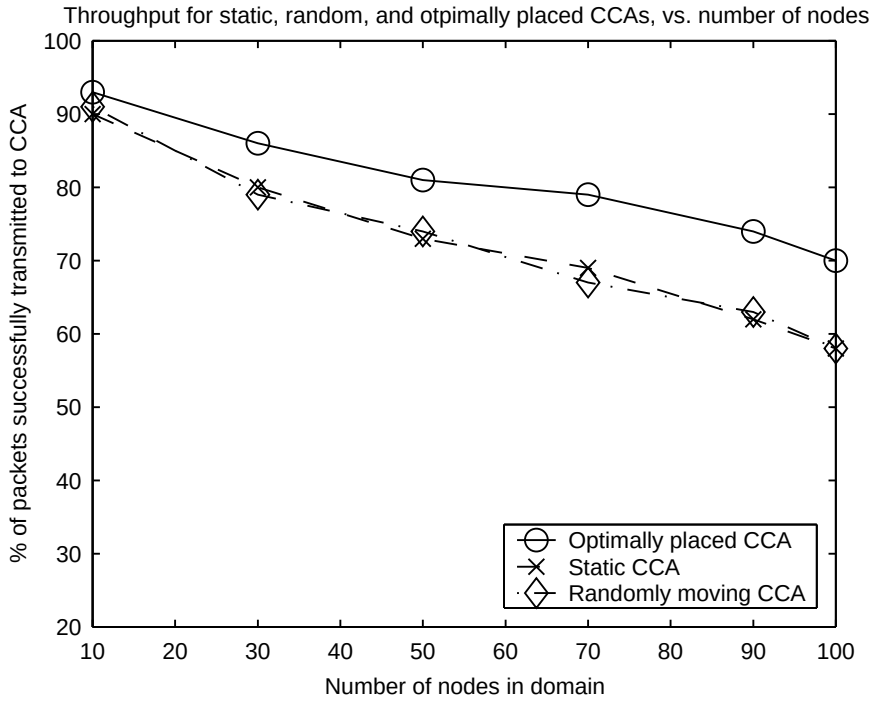

Figure 4.4: Comparison of network throughput with optimally placed CCA versus statically placed or randomly moving CCA.

dent when we increase the area covered by the objects blocking radio signals in each domain (Figure 4.5).

In Figure 4.5, initially, with very few blockages in the domain, the number of packets that are successfully transmitted to the CCA is roughly the same for the two cases, i.e., the case in which the CCA is statically placed, and the case in which the CCA is optimally positioned. However, as the number of blockages is increased causing the area covered by the blockages to increase, the open space in the domain decreases as a result, and the throughput drops dramatically if the CCA is statically placed. By comparison, when the CCA is optimally placed, we noticed the improvement in throughput to be as high as 60% per domain. In these simulations, we have assumed that the blockages permit no signal transmission

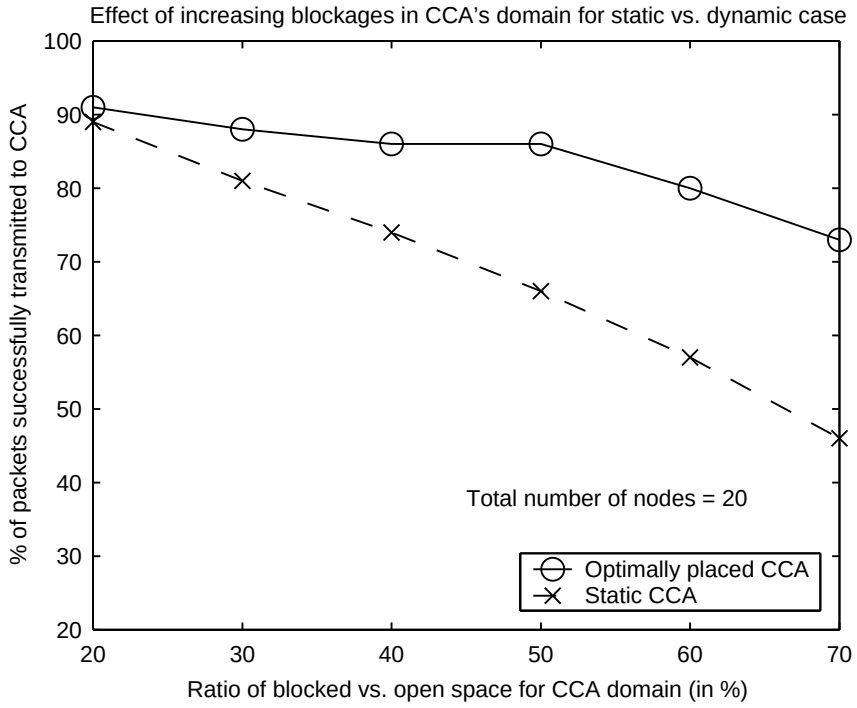

Figure 4.5: Effect of blockages on the performance of the static vs. optimally placed CCAs.

through them whatsoever, so some of the nodes may be completely cut off and isolated from the CCA. For very large numbers of blockages, the performance for the dynamically placed CCA also suffers because, then, the CCA may not be able to move quickly enough to the optimal positions, due to the constraints imposed by the blockages, and by its velocity.

Next, we performed simulations in which the ad hoc nodes offered unequal loads to the network. We first compared the network throughput when the CCA position was optimized with, and then without taking the offered load into account. The cost functions in the two cases were $f(\rho_i, \tau_i) = \rho_i$, and $f(\rho_i, \tau_i) = 1$, respectively. Each ad hoc node generated data packets so as to offer a load that uniformly varied from 10% to 90% . The results are shown in Figure 4.6. As

expected, the network performs better, in terms of throughput if the effect of the load is incorporated into the cost function, with improvements of up to 30% per domain. This may be expected, since for the case in which the loading parameter is included in the cost function, the CCA is closer to the nodes generating the bulk of the data traffic, and hence most of the packets can be delivered directly to the CCA without relays. The network performance in terms of throughput thus improves.

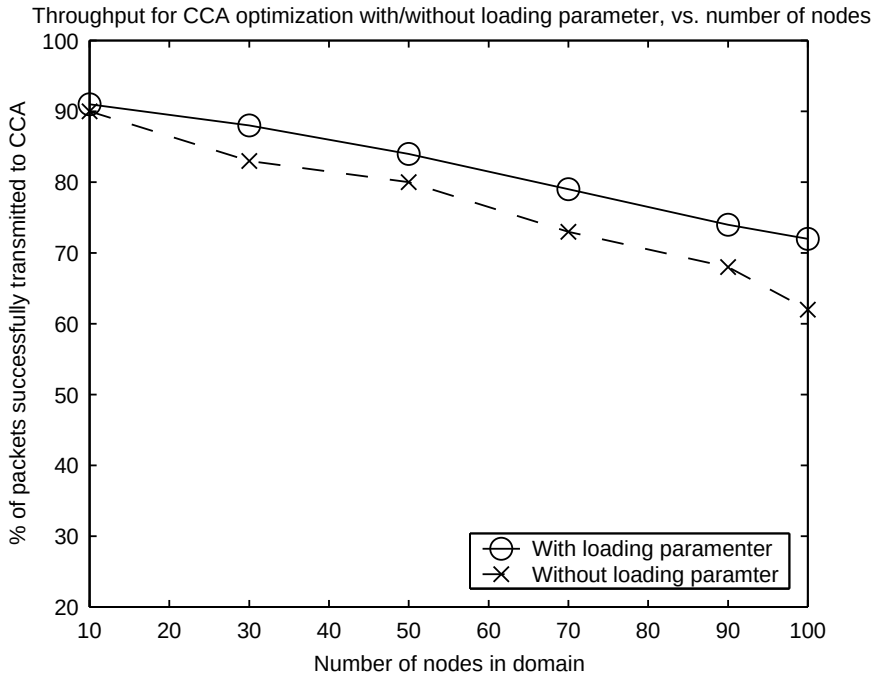

Figure 4.6: Comparison of CCA optimization calculations with/without the effect of loading parameters.

We next considered the effects of choosing the optimal CCA trajectory on data packet latency. The number of nodes in a domain was kept fixed at 20, and the offered load of an arbitrarily chosen member of the ad hoc group was varied from 10% to 90% while the other nodes generated a constant load of 10%. The results are shown in Figure 4.7. If the CCA is optimally positioned, we expect that the

data packets will experience a lower latency, on the average, than if the CCA were to be statically placed. This is because in the case in which the CCA is optimally positioned, it is closer to the heavily loaded node, and thus a large number of the data packets would no longer have to hop multiple times in order to reach the CCA, which might be the case if the CCA were to be statically placed. Thus, by positioning the CCA in accordance with the optimally computed trajectory (as opposed to positioning it statically at the center of the domain), packets are seen to experience an overall reduction in latency (by as much as 30%).

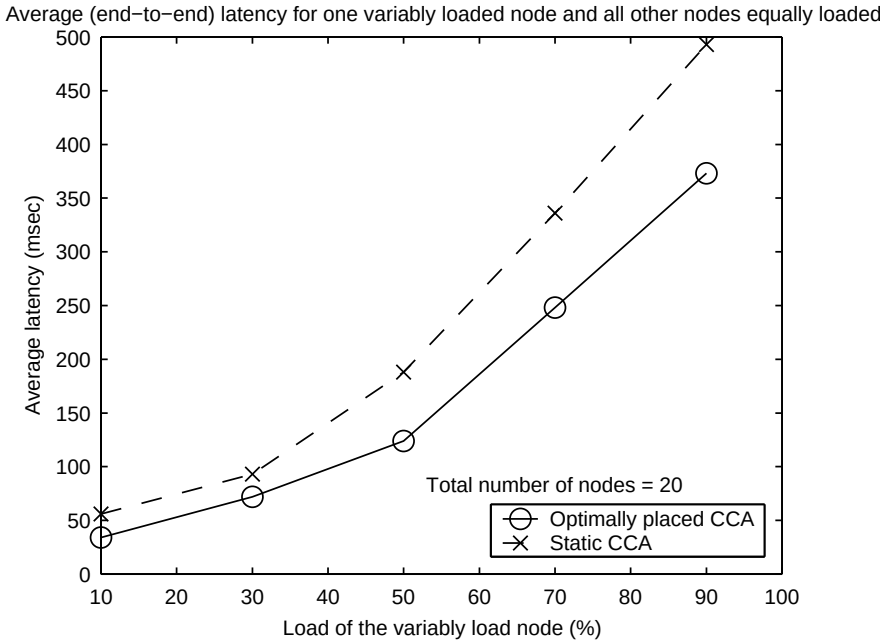

Figure 4.7: The effect of loading on latency of transmitted data packets for the static and dynamically placed CCAs

To test the performance of the network when the cost function is altered to incorporate different priorities for different nodes[12] (Section 4.3.2.1), the following simulation set-up was used: 100 nodes were deployed and the load generated by

[12]Note that a node's priority depends on the priority of the packets that it generates at the given time. This is dynamic, (akin to the offered load), and will change with time.

each node was progressively increased from 10% to 90%. For a specific value of the offered load, half the nodes (chosen randomly in accordance to a uniform distribution) generated high priority traffic ($f(\rho_i, \tau_i) = \tau_i = 10$), whereas the remaining nodes generated low priority data traffic ($f(\rho_i, \tau_i) = \tau_i = 1$). For each of these cases, the average message latency was measured while using the CCA trajectory update algorithm with, and without, the priority class as a parameter in the cost function. The results are depicted in Figure 4.8.

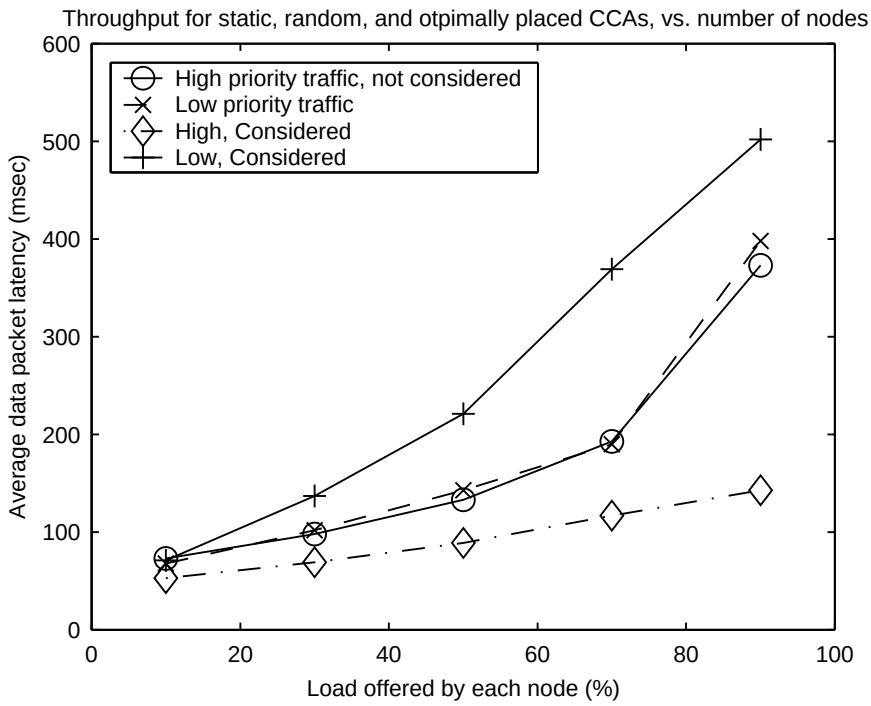

Figure 4.8: Effect of including the effect of priority in the optimization cost function.

As discussed in Section 4.3.1.1, the CCA now favors the nodes generating the higher priority traffic. These higher priority packets are delivered more efficiently–directly instead of via multiple hops–and with improved latency. Since the system capacity is fixed, the price that is paid, however, is that the lower priority traffic

suffers increased latency and reduced throughput as a consequence. This is evident in the plots shown in Figure 4.8. We also note that, as expected, when the cost function in our optimization formulation does not take packet priorities into account, but considers only the coordinates and the offered load of each node, there is no significant difference in the latency incurred by the different classes of traffic.

Finally, several simulations were performed for the cases in which multiple domains intersect, and the nodes that are in the overlapping regions of the intersecting domains now have the ability to choose their affiliations among the available CCAs. For manageable simulation run-times, the set-up consisted of four domains, each with 50 ad hoc nodes forming a group, deployed over a geographical area of 50000 units x 50000 units. We assume that 20% of the geographical area is covered by harsh terrain that act as blockages that do not allow radio signals to pass through. Each node generates an offered load of 50% and the control messages that report updates are generated every 0.5s. Initially the domains are placed such that they not intersect. Each ad hoc group can move randomly over the entire region. The group is confined to a geographical area defined by a circular domain with radius 10,000 units and moves together as a single entity. Each ad hoc node within the group can move independently in accordance with a random waypoint model, within the area of its domain. When domains intersect, nodes can change affiliations as defined by the policy in place. As described in Section 4.3.2.1, we can choose an affiliation strategy that would force nodes that are in the overlapping region to affiliate with the least loaded domain. This would result in balancing the load among the CCAs in the intersecting domains. One might expect a gain in the throughput when we implement this strategy as compared to an affiliation strategy in which nodes are statically affiliated with domains. In our simulations, we compared the above two affiliation strategies to

quantify performance enhancements that could be seen in typical scenarios. The initial load offered in each of the four domains was chosen to be different (20%, 40%, 60% 90%). As the simulation time progressed, the domains intersected, and the nodes in the overlapping regions attempted to compensate for the difference in the offered loads in the four domains by switching affiliations as described. The

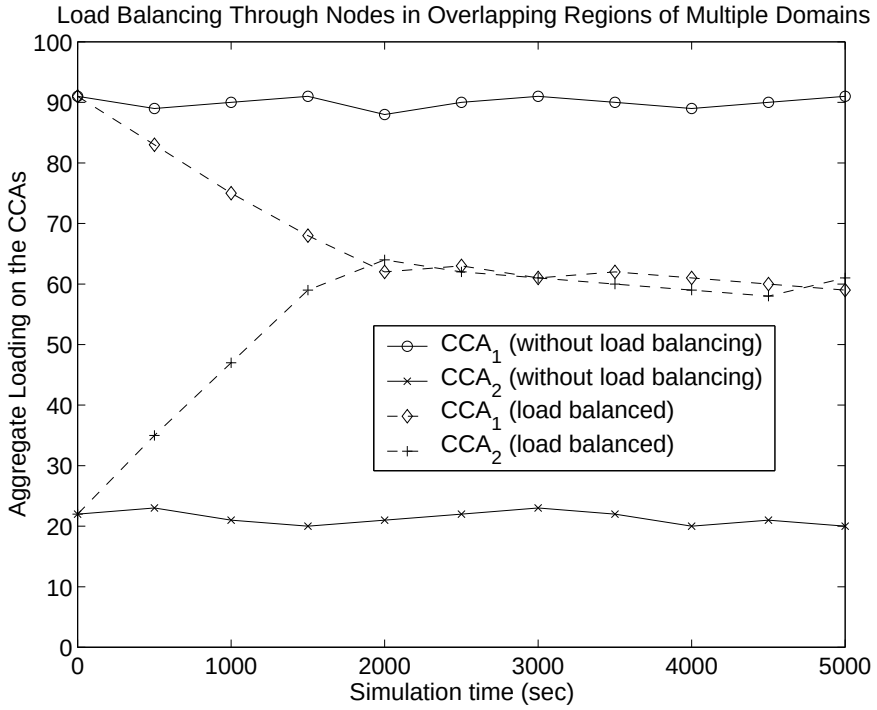

Figure 4.9: Comparison of CCA loads with and without having a method to balance loads when domains intersect.

offered load in each domain converged to the same value within a short period of time, after the domains intersected. This time depended on the velocity of the nodes in the domains. In Figure 4.9, we show the offered loads on two of the four CCA's. The loads differ to begin with, but as the domains intersect load from the heavily loaded CCA starts being switched to the lightly loaded CCA until the loads on the two CCAs are balanced. Note that without the dynamic affiliation

strategy, the loads in each domain remain static, and equal to the offered load at the initiation of the simulation.

4.6 Conclusions

In order to support scalability in ad hoc networks, one can envision the deployment of a range extension network that consists of airborne nodes, satellites, power transmission towers and such. In order to interface the ad hoc network with the range extension network, one approach could be to deploy gateways that we refer to as CCAs to relay data traffic from/to an ad hoc group to/from the range extension network. This is akin to providing a centralized infrastructure within the infrastructure-less ad hoc network. The objective of this work is to determine where the CCA is to be placed relative to the ad hoc group of nodes such that certain network performance metrics are optimized. This objective can be formulated as a set of convex optimization problems. By means of suitable modifications, we simplify these convex formulations such that they can be very efficiently solved by numerical methods.

We evaluate the improvements in network performance that we achieve by means of extensive simulations, where the CCA is enforced to follow the computed optimal trajectory. Simulation results indicate that the network throughput improves by about 10-15% per CCA domain[13] (an ad hoc domain consisting of about 50 nodes), if the CCA moves in accordance with the optimally computed trajectory as opposed to being static or moving in accordance to a random waypoint model. A similar improvement is seen in terms of a reduction in latency that the data packets experience. The cost function in our optimization formu-

[13]The overall improvement would be several multiples of this value, since a region of interest will likely contain several CCA domains.

lation can also be appropriately modified to support better performance for high priority traffic, as compared to the performance for lower priority traffic. We also consider cases wherein the nodes have the ability to choose different CCAs for relaying their inter-domain traffic and show that this ability can be exploited to balance the loads in different domains, thereby improving performance further.

We also show that the operations that are required in order to thus define an optimal trajectory for the mobile CCAs, can be performed efficiently and quickly by most commercial, off-the-shelf micro-processors, and with little additional overhead.

One particular extension of interest, is to incorporate fault tolerance in our composite ad hoc network, since the CCA is a catastrophic single point of failure. In this scenario, we can envision multiple nodes in a domain, each of which is capable of acting as the CCA. In that case, should the primary CCA fail, then, according to some pre-determined rule, one of these dormant CCAs can take over as the new CCA. This issue is investigated in detail in Section 5.4 in Chapter 5, under the more broad context of dependability of hybrid networks.

CHAPTER 5

Dependability of Wireless Heterogeneous Networks

As we have seen in Chapters 3 and 4, a sensor network—and by extension, ad hoc networks—can be very general, involving heterogeneous collections of several networks including: wireless ground ad hoc nodes, airborne nodes, low earth orbiting and geostationary nodes, access portals to the wired infrastructure, or gateways to other hybrid networks. The challenge is to enable and maintain connectivity between all the disparate components, and to serve the communication needs of the client nodes in the best manner possible. We outlined a detailed architecture in Chapter ****** as one possible technique for accomplishing this objective—by using gateway nodes as inter-domain communication portals. However, it was seen that such gateway-based architectures then resemble a hierarchical structure for information flow, as discussed earlier in Section 2.4.1 (Figure 2.9). This causes the network to have single points of failures, and thus decreases the overall reliability of the entire network[1].

In this chapter, these issues are explored for general heterogeneous networks. As discussed in Chapter 2, peer-to-peer, decentralized architectures offer flexibility and diversity for of data fusion and information flow, as opposed to centralized or hierarchical architectures. As we will see in this chapter, an additional impor-

[1]Some mitigating techniques involving redundant gateways and associated protocols are discussed in detail later in this chapter, in Section 5.4

tant benefit is increased reliability and survivability—terms that will be precisely defined in the sequel. However, these benefits all come at the price of increased network administration cost and complexity. One of the goals of this chapter will be to analyze and develop novel techniques that will allow us to optimize the reliability of networks for efficient information processing, subject to the complexity and overhead costs.

5.1 Introduction

Although significant work has been done in the context of wireline networks for improving system reliability, availability and survivability (e.g. for the public switched telephone network), very little work has been done for infrastructure based wireless networks (e.g. the cellular and PCS networks [92]), and for ad hoc networks (e.g. IEEE802.11 or Bluetooth based systems [85, 105]). With the enormous interest in the development of such infrastructure-less communication devices in recent times, both in the commercial and military world, it is imperative to investigate and understand well the statistical reliability and dependability of such systems. The focus of this chapter is the development of analysis and design tools to enhance the *dependability* of peer-to-peer wireless networks. The goal is to design a secure and adaptive networking and communication infrastructure for a system of hybrid wireless sensor nodes or general ad hoc networks.

We first introduce some terminology from the network reliability discipline. For both wireline and wireless networks, the current consensus for measuring the ability of a network to avoid or cope with a failure is by means of three inter-related parameters, as defined below.

Definiton 5.1. The *reliability* of a network is its ability to perform a designated set of functions for a specified operational lifetime, under a given set of conditions.

Definiton 5.2. The *availability* of a network is its ability to perform a designated set of functions at any given instant, under a given set of conditions. Average availability is thus a function of how often something fails and how long it takes to recover from a failure.

Definiton 5.3. The *survivability* of a network is its ability to perform a designated set of functions, given a set of network infrastructure failures resulting in service outage. The outage can, in general, be described by the number of services affected, the number of subscribers affected, and the duration of the outage, for a specified set of conditions.

The reliability, availability and survivability of a network is collective referred to as the *dependability* of the network. A heterogeneous peer-to-peer wireless network can be considered as a 'complex system', where the various parts of the system give rise to the collective behaviors of the system [46]. In such cases, the performance and reliability of the system as a whole are influenced by, and intertwined with, subtle and often indirect effects of the individual components. Our goal is to build a platform for the analysis of these problems in a systematic fashion. As mentioned earlier, a great deal of research has already been done in the related fields of hardware/software fault tolerance, statistical reliability analysis of systems, probabilistic analysis of complex systems, graph theory, etc. [50, 82]. Our aim is to build on these foundations by identifying and appropriately modifying the key techniques from this vast body of knowledge that will enable analysis or yield insight for the case of heterogeneous ad hoc networks.

Several issues make this problem considerably complex. These are: general ad hoc networks can be mobile, with dynamic performance parameters having no

long run equilibrium conditions; they may have distributed feedback effects which may be both intended or unintended; they may be primarily 'agent-based' where the agents, e.g. fusion nodes (Chapter 2) or gateway nodes (Chapter ******) act locally within some sort of group or hierarchy, but often in contention with other groups of agents engaged in similar actions (a sort of fractal organization at a coarse level).

The basic question of reliability of such systems is notoriously hard to quantify or analyze, for the simple reason that the reliability of the individual components *do not* always indicate the reliability for the system as a whole. However, it is unequivocal that the network infrastructure has to be reliable and fault tolerant. Its performance has to degrade gracefully in the face of different type of failures (e.g. progressive, cascading, catastrophic, malicious attacks, or probabilistic), and failure-recovery mechanisms have to be robust and efficient.

The standard approach is to first formalize the relevant concepts from complex systems by: 1) understanding the ways of modeling and describing the complex system, 2) mapping the interactions of the heterogeneous components that in turn give rise to patterns of behavior, and 3) analyzing the process of formation of complex systems through pattern formation and evolution.

For reliability modeling of multi-tiered networks, we show in the subsequent sections that this is essentially equivalent to characterizing the connectivity properties of the network in an abstract fashion. We rely on the mature mathematical discipline of graph theory to provide a comprehensive foundation. Furthermore, statistics can be obtained for the individual node and link failure probabilities (or estimated from the protocols themselves by inferential or Monte Carlo techniques), and standard statistical and estimation techniques can then be used to derive connectivity estimates, within confidence-interval limits. This is an ex-

perimental task that this chapter does not address explicitly; rather we assume that this data is available for the final step, which is the design of algorithms to maximize the reliability of the network, given system constraints, as discussed in subsequent sections.

With regards to mapping the interactions of the heterogeneous components (e.g. gateways and nodes that it serves), we have modified standard reliability results from the cellular and wire-line networks to apply to the ad hoc case. The mobility and dynamic nature of the ad hoc network itself can then be averaged as a sort of outer integration akin to $E(X)$ expectation calculations in probability science, subject to estimating motion and terrain statistics, and we can thus obtain figures for reliability of the network as a whole, (as experienced by an 'average' node in the network).

Finally, as these complex systems evolve with the addition/removal of network elements, the effects on overall system reliability depend on being able to modularize the reliability computations as much as possible to the level of the different sub-networks, and to derive their inter-relationships. Unfortunately, most graph theory problems of this type have been shown to be NP-hard problems [40]. Therefore, the usual recourse is to employ approximations and asymptotic analysis techniques for bounding the results, as illustrated for the case of delay in random wireless networks in Section 3.2.1.

5.2 Graph Theory Fundamentals for Modeling Ad Hoc Networks

In this preliminary section, we focus on the task of modeling ad hoc networks for the purpose of dependability analysis. This is essentially equivalent to char-

acterizing the connectivity properties of the network in an abstract fashion. Fortunately, we can rely on the mature mathematical discipline of graph theory to accomplish this task [100]. The basic idea is to formulate some appropriate reliability measures to which we can then apply current graph-theoretic techniques.

5.2.1 Deterministic Graphs

A static network of wireless or wired communication nodes (e.g. static sensor networks) can be modeled as a deterministic *directed graph* [100]. The nodes of the graph can correspond to the transmitting or receiving units, and the edges of the graph would correspond to the connections that link the nodes in the network and describe its network topology/architecture. With this simple model, we can obtain estimates of the average connectivity properties of the network, and thus bound the reliability metrics of interest. However, since general ad hoc networks can be mobile, this basic deterministic graph model is insufficient to capture the effects of mobility on the topology of the network. Instead, this is accomplished by considering the mobile wireless ad hoc network as a *random graph*, where the links state (as well as the node states) at particular instants of time have probability distributions (Section 5.2.2). Some terminology is introduced first.

Definiton 5.4. A *graph* is defined algebraically as a collection, $G(V, E)$, of two sets:

- $V(G),\ E(G)$, where $V = V(G)$ is the set of $p > 1$ nodes or vertices of the graph;
- $E = E(G)$ is a set of $q \geq 0$ pairs of nodes.

We say that G has *order* p and *size* q and refer to G as a (p, q) graph. The elements of E are the *edges*, or links of G, and edge $e = (u, v)$ is said to *join*

nodes u and v in G.

Definiton 5.5. A directed graph or *digraph* is a graph $G = G(V, E)$, except that the edge set, $E = E(G)$ consists of *ordered* pairs of distinct nodes. $e = (u, v) \in E(G)$, referred to as $e = uv$ is referred to as the link *from u to v*.

If the presence of a link in a digraph implies the existence of the opposite link ($e = uv \implies \exists\ e' = vu$), then the digraph is *symmetric*, otherwise it is *asymmetric*. A simple and intuitive way of pictorially represent a graph is by means of circles and lines. The nodes $V(G)$ of the graph can be represented by circles (labeled for convenience of reference) and the edges $E(G)$ of the graph can be represented by directed lines joining the pairs of nodes that are members of set $E(G)$. Note that since the definition we have agreed upon for a graph is not specific with regards to order or uniqueness of the node pairs, therefore, we can have loops (which corresponds to $(v, v) \in E(G)$), and directed edges (i.e. $(u, v) \neq (v, u) \in E(G)$). A simple graph with 3 nodes and 3 links is shown as Figure 5.1.

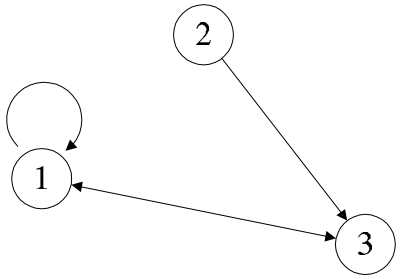

Figure 5.1: A pictorial representation of a graph (algebraic) structure

Thus, we see that such a graph representation is ideal for depicting a mobile ad hoc network, where, as mentioned earlier, the nodes are the transmitting/receiving hosts and the links represent availability of a communication channel. In the context of a general graph, self-loops make no sense for ad hoc

networks, so we exclude these cases from the set of possible $E(G)$. However, directional links are entirely possibly, as would be case with nodes with different transmit powers or receiver sensitivities which would result in unidirectional links. Therefore, we allow this possibility in our model (but exclude node pair repetitions in $E(G)$ which represent parallel edges—two or more edges joining the same pair of nodes). For the preliminary part of our analysis however, we assume symmetric links and identical nodes. Thus, this results in undirected graphs with no self-loops, as pictured in Figure 5.2

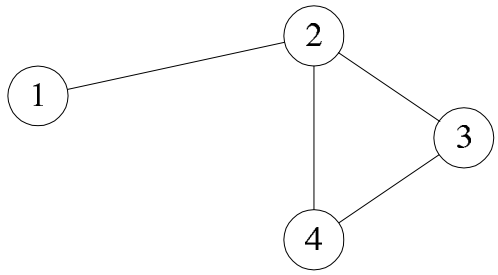

Figure 5.2: Representation of an Ad Hoc Network with Symmetric Links

The edge uv is said to be *incident to* the nodes u and v, which are adjacent nodes or neighbors. *Isolated* nodes are those who have no neighbors, and would correspond to nodes which are not in the radio range of any other node. The degree of a node $v \in V(G)$, denoted by $deg(v)$, is the number of edges the node is incident to, and can be thought of as the total number of ad nodes that are in the neighborhood of that node. Similarly, *indegree* and *outdegree* refer to incidences for directed graphs. We now note some basic facts[2], and some additional terminology:

[2]Standard proofs, given in [100]

Lemma 5.1.

$$\sum_{v \in V(G)} deg(v) = 2q \tag{5.1}$$

$$\sum_{v \in V(G)} indeg(v) = \sum_{v \in V(G)} outdeg(v) = q \tag{5.2}$$

$$\tag{5.3}$$

Definiton 5.6.

$$\delta(G) \triangleq \Delta = \text{Min}\{deg(v) : v \in V(G)\} \tag{5.4}$$

$$\Delta(G) \triangleq \Delta = \text{Max}\{deg(v) : v \in V(G)\} \tag{5.5}$$

Definiton 5.7. Two graphs are *isomorphic* if a one-one correspondence can be set-up between their nodes sets that preserves adjacency.

Thus, the graphs shown in Figure 5.3 are isomorphic. Isomorphic graphs

Figure 5.3: Isomorphic Graphs

are of interest, since any *invariant* measure of a graph G (e.g. the number of nodes, edges, or connectivity, failure measures, etc.) is, by definition, the same for isomorphic graphs. Thus, if we have a technique to calculate a measure of interest for some specific structured graphs, the results would hold for any graph isomorphic to it. From a practical point of view, however, isomorphic

graphs would actually represent different physical arrangements of the nodes in a particular domain, but with the same connectivity properties. Thus, isomorphism allows us to reduce the cardinality of the infinite set of possible geographical locations of the nodes.

Definiton 5.8. A *weighted* graph is graph a $G = G(V, E)$ together with a weight function $w : E(G) \mapsto R$ where $w(e)$ is the weight assigned to the edge $e \in E(G)$.

In applications, this might represent the cost, reliability, or capacity of a communication channel in a radio network. It is also possible to consider graphs with weights assigned to the nodes $v \in V(G)$, representing the costs associated with storage locations, power consumption, or CPU capacity of nodes in a data network. In this manner, a cost function can be built for evaluating metrics such as routing or MAC as a function of link/node formation and breakage. Our main interest will eventually be in communication networks given by a symmetric (i.e. bi-directional capable), or asymmetric (i.e. uni-directional capable) digraph with a weight function $P : E(G) \mapsto [0, 1]$. Here $P(e)$ represents the probability that the communication link $e \in E(G)$ is operational, on a bit, packet, or frame level. Ideally, this probability is allowed to depend on the direction of the link in order to model such things as local jamming, directional antennas for transmission and/or reception, etc.

We next discuss some connectivity concepts that correspond directly to connectivity in multi-hop ad hoc networks. A *walk* in a graph G is an alternating sequence of nodes and edges, e.g. $v_0,\ e_1,\ v1,\ e_2 \ldots, v_n$. It is said to connect nodes v_1 and v_n and has length n. A walk is closed if $v_0 = v_n$, otherwise it is open; it is a *trail* if its edges are distinct, and it is a *path* if its nodes (and hence also its edges) are distinct. A *circuit* is a closed path and a *cycle* is a circuit with distinct nodes (other than the initial node $v_0 = v_n$). Two nodes $u,\ , v \in G$ are said to be

connected if there is a path from u to v, and the graph is said to be *connected graph* if every pair of nodes is so connected.

Thus, we can see that graph connectivity is a fairly exact analogue of the concept behind connectivity in an ad hoc network. What is of paramount interest, however, is some measures by which *connectivity* can be quantified. The usual way to do this is to specify the number of nodes or edges that must be removed in order to disconnect G.

Definiton 5.9. A graph G is n-connected, $n \geq 1$ if it remains connected after the removal of any set of $n-1$ vertices. The *node-connectivity* $\kappa(G)$ is the smallest number of vertices whose removal results in a disconnected graph.

Similarly, an n-edge connected graph and the *edge-connectivity*, $\lambda(G)$, can be defined.

Lemma 5.2.

$$\kappa(G) \leq \lambda(G) \leq \delta(G) \tag{5.6}$$

For the design of reliable communication networks, it is important to ensure large values of $\kappa(G)$ and $\lambda(G)$, since this provides for many alterative paths for communication between nodes. Conversely, small values of these graph invariants imply that the network is relatively vulnerable and can be disabled (i.e. disconnected) by the removal of only a few of its nodes or links. Statistically, for ad hoc networks, it is usually the case that edge failures are far more common than node failures.

We conclude this brief section on graph theory fundamentals by highlighting some classical results in graph connectivity (see [100] for proofs).

Lemma 5.3. For a graph G with p nodes and q edges:

1. If $q < p - 1$ then G is disconnected

2. If $q > frac(p-1)(p-2)2$, then G is connected

3. If $\delta G \geq p/2$ then $\lambda(G) = \delta(G)$

4. If $\delta(G) > frac(p+n-2)2$, where $1 \leq n < p$, then G is n-connected.

Theorem 5.1 (Menger's Theorems). —Node/Edge Version

1. $\kappa(G) \geq n$ iff for each pair $u, v \in V(G)$ of distinct, non-adjacent nodes, there exist at least n node-disjoint paths which connect u and v.

2. $\lambda(G) \geq n$ iff for each pair $u, v \in V(G)$ of distinct nodes there exist at least n edge-disjoint paths which connect u and v.

5.2.2 Probabilistic Graphs and Reliability Measures

We have seen in the previous subsection that a static ad hoc network, such as a sensor network, can be modeled as a deterministic algebraic graph structure. A mobile wireless ad hoc network, on the other hand, can be modeled as a random graph where the nodes of the graph correspond to transmitting or receiving mobile units, and the edges of the graph correspond to the connection state of the network at a particular instant in time. Furthermore, for the links in the network represented by the dynamic edges in the graph, a weighting function, $p : E(G) \mapsto \mathbb{R}$, can be defined. This can be used to represent a variety of link-dependent parameters for the ad hoc network, e.g. the capacity of the link, or the failure probability, etc. We are particularly interested in the case where the weighting function, $P_1 : E(G) \mapsto [0, 1]$, represents the probability of an edge being operational. Similarly, another weighting function, $P_2 : V(G) \mapsto [0, 1]$ can be defined for nodes, which represents the probability of a node itself being oper-

ational. The determination of these functions is system dependent (the physical layer coding being used, network layer protocols being used, geographical terrain of operation, etc.), and is a major area of research that is not the focus of this section. It is assumed that these estimates are obtained by the reliability protocols from lower layer protocols in the network. Numerical issues are discussed, however, in the sections below.

With these metrics, we can consider some 'common sense' notions of reliability. For example, for a given *source* node in the ad hoc network, and a corresponding *target* or destination node, one may ask what is the chance of reliably sending packets from source to target? The myriad protocols for the 7 layer OSI stack are all designed to accomplish the actual connectivity and packet transfer tasks, but by themselves they cannot provide performance bounds when this question is asked from the perspective of a heterogeneous ad hoc network as an end-to-end system[3]. This is illustrative of the complex systems analogy given earlier, where the individual protocols at each layer of the network stack for each node in the network (involved in the transfer process) are working hard to ensure reliable and error free operation, but in so doing are interacting with each other in uncertain ways. As a result, the reliability of the entire process cannot be derived simply from the reliability of any particular protocol layer.

Instead, we note that we can quite naturally abstract the problem and define a probabilistic notion of connectivity to answer the reliability question. Thus, if we can gather the statistics for the individual node and link failure probabilities (or obtain this data from the protocols themselves by inferential or Monte Carlo techniques), then we can apply standard statistical and estimation techniques to obtain connectivity estimates, within confidence-interval limits. This is the

[3]except in some special cases of the grid, star, etc. networks.

essence of statistical dependability analysis as applied to ad hoc network connectivity [101]. The mobility and dynamic nature of the ad hoc network itself can then be averaged as a sort of outer integration akin to $E(X)$ expectation calculations, subject to estimating motion and terrain statistics, and we can thus obtain figures for dependability of the network as a whole, *as experienced by an 'average' node in the network*. A similar approach was taken to estimate the order of the delay in random wireless networks, as shown in Section 3.2.1.

The most common statistic used in the technical literature for the reliability of wired networks is the *K-terminal reliability* [13].

Definiton 5.10. If $K \subset V(G)$ is a subset of the nodes, and the *K-terminal reliability* is the probability that every two vertices in K are connected by a path of operational edges.

In an ad hoc network, we are more interested in a single source-destination pair, known as *st*-reliability, in which case $|K| = 2$, or in the *all-terminal* reliability, for which $K = V(G)$. The former is denoted in the technical literature as $\mathrm{Rel}(G; s, t)$. For an undirected graph, the *st*-reliability is a special case of that for a directed graph (since each edge is then equivalent to two edges with opposite directions but equal probability of success). In that case, the *st*-reliability is simply the probability that the network is connected, or that G contains at least one operational spanning tree or subgraph. This is a useful measure for any ad hoc network, since it indicates the likelihood of a transmitted package to reach a desired destination from the source.

The subtle thing to note here is that the *st*-reliability metric does not measure the probability along *a specific route* that a particular routing/MAC algorithm has discovered, or looked up in a table. Rather, it measures the *possibility* that *any* operational path can be found by the underlying algorithms. This is a more

accurate measure of the network performance, as experienced the data packets in the network. To generalize for a global 'average' for the network as a whole, another common metric is the following.

Definiton 5.11. The *resiliency* of a network is the *st*-reliability averaged over all $p(p-1)$ possible node pairs:

$$\mathrm{Res}(G) = \frac{1}{p(p-1)} \sum_{s,t \in V(G)} \mathrm{Rel}(G; s, t) \tag{5.7}$$

Thus far the emphasis has been on link failures. An important metric can now be defined for node failures in networks, which characterize the *vulnerability* (Section 5.1) of a network. To capture the concept of network vulnerability (e.g. in the face of hostile attacks or cascade failures of nodes), a concept from the wired networks discipline can be extended to the heterogeneous wireless case.

Definiton 5.12. The *Node-connectivity factor* (NCF) represents the average number of nodes that have to be removed from a network in order for the remaining subgraph to be totally disconnected. This quantity can be defined recursively, since if G is disconnected and has components $G_1,\ G_2, \ldots, G_m$, then:

$$\mathrm{NCF}(G) \triangleq \Delta = \sum_{i=1}^{m} \mathrm{NCF}(G_i) \tag{5.8}$$

For a connected graph G, if $\kappa(g)$ is the size of the smallest set $V_o \subset V(G)$ for which $G - V_0$ is disconnected, and if $\chi_1, \ldots, \chi_n$ is the collection of all such node sets V_0, then:

$$\mathrm{NCF}(G) \triangleq \kappa(G) + \frac{1}{n} \sum_{i=1} n\mathrm{NCF}(G - \chi_i) \tag{5.9}$$

The NCF and a similarly defined quantity for links—the *link-connectivity factor* (LCF)— are useful indicators of the relative advantage of one particular resource allocation scheme in a network versus another, and thus aids in optimizing network dependability. In Section 5.3.1, this parameter is used in defining one such

novel optimization protocol for heterogeneous wireless networks. Finally, we consider the diameter of a graph:

Definiton 5.13. The *diameter* of a graph G is the maximum hop distance supported by the edge configuration:

$$\text{Diam}(G) = \max\{\text{dist}(u, v) : u, v \in V(G)\} \tag{5.10}$$

Recently, this metric has been preferred for quantifying forms of the quality of service (QoS) for networks, and is of considerable interest in the context of hybrid ad hoc networks. The basic idea is once again simple: a graph with a small expected diameter is better-connected than one with a large expected diameter.

5.3 Dependability Optimization for Node Failures

We have thus far developed an analytic and optimization framework for investigating the dependability of hybrid ad hoc networks in the face of link failures. In this section, we treat the problem of node failures, and propose a relatively simple network design scheme for optimizing reliability in networks with the possibility of node failures.

Statistically, in mobile ad hoc networks, link failures are orders of magnitude more probable that node failures, because of mobility-induced dynamic topology changes in the network. Thus, optimal network design focusing on routing and MAC techniques for handling link failures are more commonly used than node failure protocols. However, being able to handle node failures is of paramount importance in some specific architectures. For example, in the case study presented in Section 5.4, the NGI architecture involving gateways in heterogeneous

hybrid ad hoc networks is studied. We observed earlier in Chapter 4 that gateways are single points of catastrophic failures for such multi-tiered networks. In this regard, we present distributed, fault tolerant techniques for handling such node failures—primarily by relying on redundancy in the network. A similar situation applies for the case of static sensor networks performing data fusion at centralized or hierarchical nodes (Section 2.2.4).

Single points of failures are a characteristic shortcoming of all hierarchical networks, as discussed at the outset in Section 2.4.1. This is the reason for preferring ad hoc network architectures in NGI networks. We have determined that most of the existing survivability measures that have been reported are inadequate for applying to hybrid networks since the Internet-centric hybrid wireless networks of today are subject to deliberate and/or random attacks [101]. Since the destruction of a network is essentially a process in which the nodes of the network are gradually destroyed until completely disconnected, a dependability measure should be based upon such a process and its definition should reflect the survivability of the network during the entire destruction process. We now propose a dependability measure for hybrid networks to quantify and optimize node failure survivability.

5.3.1 Distributed Node Resilience Criteria (DNRC) for Peer-to-Peer Networks

There have been many measures proposed for the dependability analysis of node (and link) failures in networks [75]. Most of the techniques are based on graph theory metrics, and primarily on minimum-cut tree enumeration techniques. However, these methods generally yield complicated computations, except for well known network structures such as grid or star networks. Especially for the case

of mobile networks, the computation time (and network overhead) required often makes minimum-cut tree calculations moot, since the topology will have changed by the time the dependability results become available for network optimization.

To mitigate this undesirable situation, we now develop a new measure of dependability for hybrid networks. This is based upon the estimation of the network connectivity in a complete destruction process, i. e. the network connectivity is summed with the node of the network being removed one by one until the network becomes disconnected.

Definiton 5.14. For a network G with n nodes, the *Distributed Node Resilience Criteria* (DNRC), denoted by γ, is defined as:

$$\gamma(G) = \sum_{k=0}^{m-1} \xi(k) \tag{5.11}$$

where $\xi(k)$ is the *connectivity measure* of network G_k, which is produced by removing the most important node from the network G_{k-1} for $k = 1, 2, \ldots, m-1$, and m is the number of nodes which have to be removed before the network becomes totally disconnected. Thus, $\xi(0)$ measures the connectivity of the whole network G, and $\xi(k)$ is given by:

$$\xi(k) = \sum_{i=1}^{n-k-1} \sum_{j=i+1}^{n-k} \mathrm{NCF}_k(i,j) \tag{5.12}$$

Here $\mathrm{NCF}_k(i,j)$ is the node connectivity factor, as defined earlier in Equation (5.9), between i and j in the network G_k, and $n-k$ is the number of nodes in the network G_k. $\mathrm{NCF}_k(i,j)$ can be simplified in this case to:

$$\mathrm{NCF}_k(i,j) = \sum_{r=1}^{x} \frac{1}{\mu(r)} \tag{5.13}$$

where r is the number of independent paths between nodes i and j and $\mu(r)$ is the number of hops long the r^{th} independent path between nodes i and j.

Definiton 5.15. The *priority factor* of node i, denoted by $\mathrm{PF}(i)$, is the number of all independent paths that include the node i in the network G_k, for $i = 1, \ldots, (n-k)$.

Since our objective in introducing the DNRC measure is to reduce the complications inherent in computing standard dependability metrics, we assume some plausible node failure characteristics to keep the calculations relatively tractable. These are:

- The nodes are considered to be either operational or non-operational. This removes the necessity of using complex queuing theory models to account for partially operational nodes via Markovian statistics.
- Only a single node is rendered inactive at any instant of time. This eliminates the case of multiple nodes being damaged simultaneously (which is nonetheless possible in tactical deployments of dense sensor networks, e.g., where a group of nodes are targeted for destruction). Without this constraint, the analysis becomes significantly involved, since the network architecture then changes drastically.
- Link failures are not considered in the optimization calculations. This metric is explicitly geared to handle node failures; link failure optimization techniques are assumed to be dealt with the underlying routing/MAC protocol for the wireless network.

An example calculation is illustrated next. Consider the network as pictured in Figure 5.4, for graphs (i)G and (ii)G_1 In the network G, Figure 5.4(i), we can

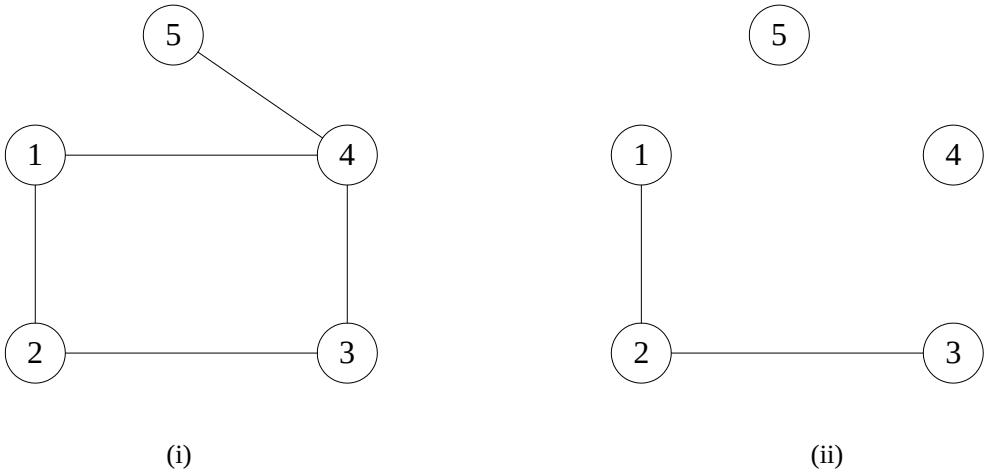

Figure 5.4: $\gamma(G)$ Calculation for a Simple Network.

compute as follows:

$$\begin{aligned} k=0 \quad & \mathrm{NCF}_0(1,2) = 1 + 1/3 \quad \mathrm{NCF}_0(1,3) = 1/2 + 1/2 \\ & \mathrm{NCF}_0(1,4) = 1 + 1/4 \quad \mathrm{NCF}_0(1,5) = 1/2 \\ & \mathrm{NCF}_0(2,3) = 1 + 1/3 \quad \mathrm{NCF}_0(2,4) = 1/2 + 1/2 \\ & \mathrm{NCF}_0(2,5) = 1/3 \\ & \mathrm{NCF}_0(3,4) = 1 + 1/3 \quad \mathrm{NCF}_0(3,5) = 1/2 \\ & \mathrm{NCF}_0(4,5) = 1 \end{aligned} \tag{5.14}$$

thus:

$$\xi(0) = \sum_{i=1}^{4} \sum_{j=i+1}^{5} \mathrm{NCF}_0(i,j) = 9.67 \tag{5.15}$$

The priority factor of each node in the network G is as follows:

$$PF[1] = 11; \quad PF[2] = 10; \quad PF[3] = 11; \quad PF[4] = 13; \quad PF[5] = 4 \tag{5.16}$$

The most important node (with the highest priority factor) is node 4, whose destruction leads to network G_1, Figure 5.4(ii), for which $\xi(1)$ can be calculated

in a similar manner:

$$\xi(1) = 1 + 1/2 + 1 = 2.5 \tag{5.17}$$

In this case, the important node is node 2, with PF[2]=3, and its removal leaves the network completely isolated. Therefore m=2 and thus:

$$\gamma(G) = \xi(0) + \xi(1) = 9.67 + 2.5 = 12.17 \tag{5.18}$$

For the defined DNRC metric, the key idea in the calculations is to remove the most important node first (the node with the highest priority factor), which results in a substantial savings in computation time. This implies that the metric captures the worst case situation of the most important node being destroyed first, and is therefore a lower bound on the dependability of the network as a whole. Furthermore, the metric has also combined the effects of link failures associated with a node failure. It is thus a combined dependability metric that is directly applicable to the NGI network, as discussed in Chapter 4. It is also applicable to other types of data fusing sensor network architectures such as those that are charaterized by having a small number of processing nodes supported by a dense sensor mesh (Chapter 2).

5.3.2 Simulation Study of the DNRC Metric

To verify the proposed dependability metric, we have automated the computations discussed in Section 5.3.1 on the MATLAB simulation environment, with C/C++ customizations. Standard data structures for graph representations, via adjacency and incidency matrices, were programmed for medium scale networks of up to 50 nodes. The scenario consisted of n static nodes representing, for example, sensor nodes in a data fusion architecture, with $n \leq 50$. The subsequent steps in the simulation are summarized below:

Algorithm 5.1 (DNRC Simulation).

- *Input constants (set 'a priori')*: Randomly generated graph G adjacency and incidency matrices.
- *Output*: calculated DNRC and P values (probability that remaining network is still connected).
- *Initialization Step*: Arbitrarily chose $e \gg n$ links among the pairs of the nodes in the network, to simulate a randomly designed, connected, network architecture G, with a small diameter, typically $\text{Diam}(G) \leq 2$ or 3.
- Calculate D_0, the baseline DNRC measure for the random graph.
- $\{$*While iteration count* $\leq$ *1000*$\}$, **DO**:
    1. Randomly disable 50% of the nodes in the graph, along with the adjacent edges.
    2. Determine if the resulting graph is still connected. Update count.
    3. Reset to original graph G and repeat.
- Estimate P, the fraction of the instances when the network was still connected.

The probability value, P, is the Monte Carlo simulated quantification of the likelihood of the network remaining connected despite suffering upto a 50% destruction of its nodes. Higher P values for a particular architecture imply that network is more dependable and fault tolerant of node failures. On the other hand, the DNRC values are computed for the same network as prescribed by Equation (5.11).

On comparing the results for a number of different networks of varying sizes, we observed that for higher DNRC values, the measured P values were correspondingly higher. Thus higher DNRC values implied a more node-failure tolerant, dependable network. Thus a clear correlation was observed experimentally, as expected, and provides validation of the utility of the metric. Table 5.1 summarizes some of the other results that were observed with different network sizes and configurations.

Table 5.1: DNRC and Connectivity Probability Results for Random Networks.

# of Nodes	*Node Configuration*	*P value*	*DNRC value*
10	ring network	39	54
10	partial grid	42	77
20	fully connected	54	341
20	random (Diag(G)=2)	27	113
50	grid network	40	500

Unfortunately, a clear relationship between P and DNRC values is not yet evident, since absolute DNRC values depend on the network size. Normalized values of DNRC[4] are possibly better, but this is a subject of further research. However, for networks of the *same* size, the DNRC offers an intuitive comparative tool when pitted against DNRC values for standard network architectures such as ring, grid, fully connected, etc. As evident from the table, higher DNRC values in these cases correspond to networks with higher node-resiliency, for random vs. structured networks of the same size. Furthermore, since the connectivity probabilities of the standard grid, ring, etc. networks can be obtained a prior, the

[4]normalized with respect to network node size, n

DNRC computations for an arbitrary network can be compared to these standard values to obtain a quick measure of the node-resilience for that network.

Alternatively, given a certain network topology, the DNRC value for that topology can be used to re-design the architecture of the network for yielding higher DNRC values, and hence more dependable networks, tolerant of node failures. This is the idea behind the optimization algorithm for improving network node dependability, as outlined in the next section.

5.3.3 Optimization of Networks Using DNRC Metric, $\gamma(G)$—a Design Flow

As mentioned in the previous section, the DNRC metric, $\gamma(G)$, provides a quick and efficient method for evaluating the dependability of moderately sized networks. It can also be used to optimize the network to enhance the dependability of an existing network. The process for optimization of the network can be formulated as follows:

Algorithm 5.2. 1. Calculate DNRC value, $\gamma(G)$ of the initial network.

2. Determine the two nodes I and J with the maximum $\text{NCF}_k(i,j)$ between them, and remove the link between them

3. Establish a link between the two nodes of the least importance

4. Repeat until $\gamma(G)$ value stops increasing.

This algorithm can easily be applied in the context of hybrid networks like the NGI network, where there are specific data aggregation points such as the gateways or the satellite nodes. The specific DNRC dependability calculations can be aggregated out at these nodes, since the network state information can

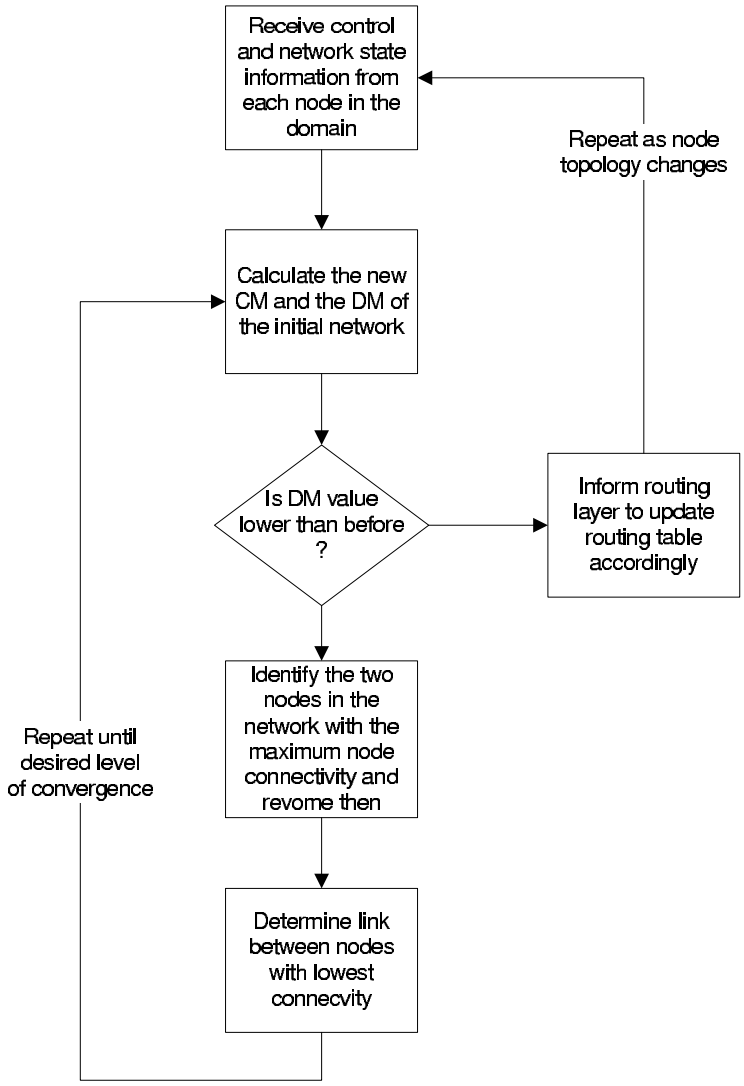

Figure 5.5: Flowchart for CCA or 'Aggregation' Node Local Computation for Optimizing Reliability

be queried most easily from these locations these locations (e.g. position, loading, BER values that are forwarded to the gateway nodes as part of the route update messages). In this manner, the route themselves can be updated and links explicitly instructed to be turned on or off according to the results of the

$\gamma(G)$ calculations. The flowchart of the algorithm can be summarized as given in Figure 5.5.

5.3.4 Concluding Remarks

The technique presented in the previous sections provides a form of *Reliability Based Node/Route Selection Process* for enhancing the dependability of a peer-to-peer wireless network. We have provided a platform model on which to analyze hybrid networks, and defined a novel measure for the dependability of such networks. The computational process involved in calculating the dependability measure for an arbitrary network was seen to be a tractable iterative algorithm which enabled us to create a network design flow (for moderate sized networks) for *improving* the reliability of the network.

5.4 Case Study: Dependability Protocols for the NGI Network

Thus far, we have discussed techniques for the dependability analysis and design of general peer-to-peer networks, and Section 5.3 in particular introduced techniques for node failures in moderately sized (primarily static) networks. In this section, we apply and extend these ideas to the canonical example of the DARPA proposed Next Generation Internet network, which is a hybrid, multi-tiered, heterogeneous "network of wireless networks", as discussed in detail in Chapter 4. We will consider this as a case study for devising fault tolerance techniques that will improve dependability of this hybrid network at the points of its maximum vulnerability—the gateway nodes.

5.4.1 Dependability of the NGI Architecture

We consider the hybrid satellite and multiple-hop wireless network as proposed in the DARPA Next Generation Internet (NGI) initiative[5], Chapter 4, and shown in Figure 5.6. This is a heterogeneous network comprised of different components,

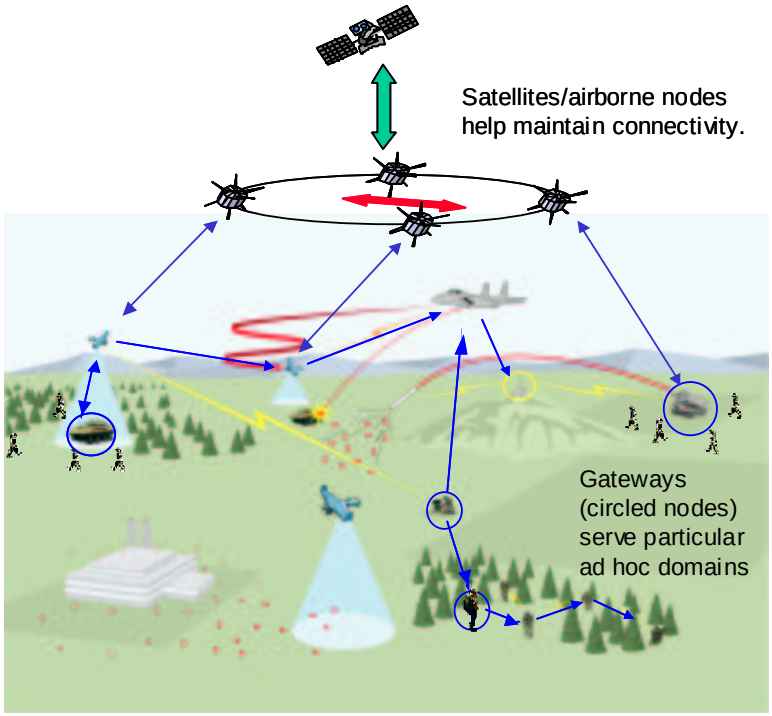

Figure 5.6: NGI Architecture and Failure Points

such as wireless ad hoc networks, low earth orbiting (LEO) or geostationary earth orbiting (GEO) satellites, and portals to the wired core networks. It represents an attempt to enable connectivity among the various types of wireless systems and services that are in use today, and presents a vision of the *ubiquitous networking* concept of future networks.

Our interest is in the wireless mobile ad hoc nodes that enable cross-platform (e.g. ground node to airborne) connectivity. It is assumed that in typical applica-

[5]http://www.darpa.mil/ito/research/ngi/

tions of the NGI network concept, users forming ad hoc networks will be clustered into separate mobile groups or domains, e.g. in troop deployments involving multiple independent platoon units. The objective, then, is to enable reliable, secure communications among the various node elements across different, geographically disparate domains. Relatively powerful mobile *gateway* nodes, which are nodes that are capable of interfacing between the various hybrid components of the network are also deployed in the domain of each cluster of nodes, to facilitate communication with a LEO or GEO satellite, or with each other. Hence, these gateway nodes serve to extend the range and connectivity of the clusters of mobile nodes. The environment is characterized by the presence of terrain blockages or other severe channel impairments.

Since the gateways are expected to be the only nodes among the (ground-based) nodes to be equipped with multi-protocol capable radios, consequently, all the communication from the ad hoc network to other components of the NGI extended network have to be routed through these gateway nodes. Any node in an ad hoc network that wishes to send data to a node in another ad hoc network that is part of the NGI infrastructure has to first route the data packets to the gateway. The gateway then forwards them to the appropriate gateway in the destination ad hoc network, which in turn forwards the packets to the destination node. However, for this chain of events to execute successfully, each node in the extended network needs to be configured with the information for identifying the gateway that it is affiliated with. If only a single gateway is present in each ad hoc network, then the job of configuring each node with gateway information is not difficult. As in wired LANs, the mobile nodes can be configured either manually or during IP address assignment using DHCP.

However, from a fault tolerance point of view, a gateway based architecture

is undesirable because it has catastrophic single points of failure. If each network domain has just one gateway, then the failure of that gateway will disconnect that ad hoc network from the NGI infrastructure. The disadvantages of using a hierarchical network, such as NGI, are well known for distributed systems in general, and was discussed in Section 2.4.1. However, for the NGI architecture, there are no other viable alternatives, since the communication systems involved rely on different physical and network layer protocols, so *gateway* or bridge nodes are the only portal through which common data can be exchanged. This introduces a serious bottleneck at the gateway node, both from a network load performance and from a dependability perspective. As discussed in Section 4.3.2, a viable, simple solution to this dilemma is to introduce redundant gateways in each sub-domain of the network. The redundancy provides a measure of fault tolerance, and eases the load on any one gateway node—provided that the network signalling and management operations for coordinating multiple gateways are handled efficiently. Furthermore, all gateways may not be able to communicate with LEO satellites at all times for various reasons (e.g. terrain blockages, buffer overflows, etc.). In such cases, at least one other gateway-capable node needs to take over the responsibility of being the gateway for the group, and the mobile nodes within its purview need to be informed to route their *inter*-domain packets to the new gateway. Unfortunately, standard Internet inter-domain routing protocols like BGP do not apply in these cases because of the high frequency of changes that is likely in an ad hoc network.

Thus, one of the end objectives of this section is to determine a group of protocols that enable *gateways* to efficiently incorporate security and a measure of dependability and distributed fault tolerance during their normal operations. In the following sections, we demonstrate mechanisms by which the mobile gateways can alternate amongst themselves the task of being the primary gateway

providing cross layer connectivity, and thus ensure fault tolerance in case of gateway failures. These techniques also address the security problem associated with gateway-centric operations in tactical applications, when critical gateways may be commandeered or maliciously controlled to disrupt the network. To avoid this situation, the responsibility of being the gateway should be rotated among the gateway-capable nodes at pseudo-random times, in an authenticated, distributed manner. This serves to protect the gateway resource from hostile attacks on the system that may compromise the connectivity or data integrity of the network. Time-out conditions can also be set so that if a gateway fails and thus cannot communicate its status, then the other gateway-enabled nodes would take over the responsibility autonomously.

In the following sections, these issues are addressed systematically for dependability optimization of gateway-centric, hybrid, multi-tiered networks. In particular, the following are novel applications of the dependability concepts, as discussed in this chapter:

- Fault tolerance algorithms by which the network can detect failures among gateways, and can coordinate the selection and initiation of backup gateways.

- A technique for enabling a random and secure mechanism for rotating the responsibility of being a gateway among a group of gateway-capable nodes. This is done by modifying and reducing to practice standard distributed processing techniques based on generalized Byzantine fault-tolerance algorithms. The algorithms have been specialized to apply to the case of mobile, wireless gateways.

- The design of the relevant network control features such as routing and

medium access control messaging, to implement the secure and distributed fault tolerance.

- An implementation and simulations study of the proposed algorithms, utilizing standard DSDV and DSR routing algorithms [79] to estimate overhead and throughput performance.

5.4.2 Prior Research

It is somewhat surprising that scant literature exists that addresses distributed fault tolerance and reliability design for ad hoc networks explicitly. A wealth of literature exists regarding *centralized* wireless schemes, such as cellular telephony, but peer-to-peer network reliability has been relegated to the best efforts of the routing and MAC layer protocols, with no particular system design approach for hybrid networks.

Several routing protocols, for e.g., DSR [41], DSDV [68], and TORA [66] have been proposed for ad hoc networks. Almost all of these algorithms are targeted for routing within the ad hoc network only. They do not extend to heterogeneous networks containing a mix of both ad hoc components and infrastructure-based networks. As discussed earlier, such a setup has multiple gateways present within network that are capable of routing packets between the ad hoc network and the other types of fixed or airborne networks. Anycasting is a possible solution where all the gateways can be grouped into a single anycast address and mobile nodes can use this address as the router for communicating with hosts not in their ad hoc network. But not all routing protocols support such anycast mechanisms. Vaidya et al. [45] propose extensions to TORA to support anycasting and Dante, et al. [38] propose a sink based anycast routing protocol.

Haas, et al. [104] discuss the security related issues related to routing in ad hoc networks and propose that ad hoc networks should have a distributed architecture with no central entities to achieve high survivability. They propose to use $(n, t+1)$ *threshold cryptographic* methods to decentralize any cryptographic operation (e.g., signing a key), where at least $t+1$ nodes have to collaborate to perform the operation and no t nodes can achieve the same result, even operating in collusion.

Any fault tolerant algorithm has to be inherently distributed. The complexity of such algorithms in ad hoc networks is elevated due to the highly dynamic nature of such networks and the unreliability of the communication medium. Hatzis, et al. [35] propose two distributed leader election algorithms for ad hoc networks. Their algorithms require that all nodes know the coordinates of the space in advance. In contrast, Malpani, et al. [57] propose distributed leader election algorithms that are based on TORA. However, both these works consider only the dynamic and multi-hop nature of the mobile ad hoc network and assume that the communication medium is reliable and that the nodes are 'well-behaved'. Other distributed problems solved for the mobile ad hoc networks are *mutual exclusion* [98], and *reliable broadcast* [65]; however, none of these deals with the case of misbehaving or *malicious* nodes.

In contrast to these efforts, we propose distinct algorithms for handling distributed gateway failures in a hierarchical setting, and study the overhead and throughput performance in a detailed simulation environment (Section 5.4.5.

5.4.3 Failure Recovery Modes for Gateways in Hybrid Network

We now discuss the general techniques for recovering from gateway failures in a multi-tiered wireless ad hoc network. In general, hierarchical or centralized

controllers for distributed processing can face a variety of failure conditions during their normal operations. In the context of wireless gateways in a mobile ad hoc network setting, these failures can be classified into the following three broad groups:

(i) *Fail-stop*: This happens when a gateway is destroyed or otherwise incapacitated.

(ii) *Intermittent:* This can occur when gateways are disabled temporarily because of terrain blockages, etc.

(iii) *Byzantine:* This happens when a gateway is confiscated and is being manipulated by the enemy.

In the following subsections, we propose combined algorithms that can handle the first two types of failures mentioned above, and also formulate a distributed algorithm for the third type of failure, based on the standard *Byzantine Generals* algorithm, for gateway selection.

5.4.3.1 Fail-stop Behavior and Intermittent failures

We consider both case **(i)**: fail-stop failures, and case **(ii)**: intermittent failures together, since they can be handled by similar techniques. In fail-stop failures, the number of gateway-enabled nodes that are available in an ad hoc group is permanently reduced (through destruction or power/operational outage). In contrast, for intermittent type failures, gateway nodes are temporarily disabled as these nodes experience intermittent satellite connectivity (or terrain blockages). But in this case, the total number of simple nodes (non-satellite connected nodes) increase since the gateways then become simple nodes.

There are several solutions by which these failures can be handled. These solutions can be broadly divided into two categories:

(i) *Mobile node initiated.*

(ii) *Gateway initiated.*

The solutions for each category can also be sub-divided into two types:

1. *Reactive*

2. *Proactive*

Thus there are four possible combinations of solutions, and the algorithms corresponding to these solutions are outlined in Section 5.4.4.1, and presented in flowchart and algorithmic form in Section 5.4.7. For estimating the numerical order for the control messages generated for each strategy, we assume a network consisting of m mobile nodes and n gateway nodes, per ad hoc network domain.

5.4.3.2 Gateway Byzantine Faults

The more difficult fault arises when a gateway is confiscated and is being manipulated by enemy. The confiscated/manipulated node is henceforth referred to as the malicious gateway. The fundamental problem that arises as a result is that if this node manages to assume the role of the gateway for the network, either during the regular gateway rotation step or by subterfuge, then all communication in the system is rendered insecure, or worse. The ideal fault tolerance goal for this scenario will be to prevent the malicious node from becoming the gateway, or interfering in the gateway selection or data routing tasks.

A gateway, when active, attracts a lot of communication from nearby mobile nodes, as part of it regular function of routing inter-domain packets. However, an analysis of the traffic pattern might reveal the position information of the active gateway, and thus making it susceptible to attacks or confiscation. To avoid these problems (as well as for power conservation reasons), it is clear that no gateway should be active for long periods. Instead, the task of being the gateway in a domain should be rotated among the gateway-capable nodes in that local network. In addition, no mobile node should use same gateway for communication for long periods. This is to avoid a malicious node claiming to be active and manipulating the packets from that mobile node. This also avoids the collection of contiguous encrypted packets of data by the malicious gateway, where the contiguity may become helpful in decrypting the data or revealing of the key used to encrypt the data.

Thus, an algorithm is required that assigns the role of the gateway among the gateway-enabled nodes in a pseudo-random fashion. Any such algorithm that selects active gateways and configures the nodes with at least one gateway should follow three criteria:

(i) No gateway should be allowed to be active for long period of time.

(ii) No mobile node should use a single gateway for long period of time.

(iii) The order in which gateways become active should not be pre-decided or predictable.

If the selection sequence is pre-decided, then a confiscation of a gateway will allow the malicious node to know in advance exactly the times when each gateway becomes active. It will also know which gateway is being used by each mobile node. To avoid this vulnerability, any gateway selection algorithm should follow

the following third criterion: i.e., gateways that will become active for a time period should be selected just prior to that time period, by at least a majority of gateways participating in the selection procedure. This is a distributed majority selection problem, which is a modified form of the 'Byzantine Generals Problem' that has been extensively studied [49]. For the case of gateways in mobile ad hoc networks, we have modified the general algorithm and reduced to practice the essential algorithm that satisfies all the three criteria mentioned above. This is discussed in Section 5.4.4.2, and flowcharts and pseudo-code are also included in Section 5.4.7.

5.4.4 Algorithms for NGI Gateway Fault Tolerance and Security

We now discuss the algorithms for the four possible combinations of solutions for recovering from gateway failures, as mentioned earlier, as well as from Byzantine-type faults for a multi-tiered wireless ad hoc network. The fault tolerance algorithms that we propose are designed to be implemented as hardware or software protocols on computing devices operating as part of a wireless network. They rely on underlying and overlying systems and software modules for complete system operation. As such, our assumptions regarding the NGI system architecture are as follows:

Assumptions Regarding NGI Architecture for Fault Tolerant System Design

- The fault tolerance algorithms are network programs residing at the network layer and/or transport layer of the 7 layer OSI stack. They will interact with the routing protocol to direct the formation/destruction of routes within the ad hoc network.
- The network employs standard MAC, Routing or Physical Layer protocols.

- The network *may* have application level or packet level encryption/data authentication, but strong encryption and mutual authentication is an absolute necessity for passing the control messages among the gateways executing the fault tolerance algorithms. This is to ensure against Byzantine attacks from unauthorized nodes.

- During initial deployment, all the gateway-capable nodes in an ad hoc network are aware of each other's identity (e.g. authentication public key, network address etc.).

- During normal operations, only one gateway per ad hoc group is active as the primary gateway for that domain. Backup gateways are activated (one at any given time) in response to failures or for security/alteration reasons. There is no system architectural reason that prevents multiple gateways from operating simultaneously. However, simulation studies have suggested that the overhead necessary (in handoff, tracking, power conservation etc. for the mobile nodes in the group) for multiple operational gateways per group outweighs the benefits.

- During initialization and boot-up, the gateway node with the lowest network ID assumes the task of being the gateway and floods the network with this identification information. Standard underlying protocols are assumed to ensure authentication (e.g. by digital signatures, etc.) for the other nodes in the network.

- Since gateway nodes are expected to be GPS enabled, a common synchronization clock is assumed to be available to all the gateway nodes for reconciling time-out clocks and timers for the distributed algorithms.

5.4.4.1 Fail-stop or Intermittent Failure Recovery Algorithms

Mobile initiated and Reactive

In this case, each mobile node starts a search for a gateway only if it finds its current gateway to be inactive. The detailed algorithm is shown in flowchart form as Figure 5.7 with the accompanying pseudo-code, in Section 5.4.7[6]. To check the 'liveness' status of its current gateway (which indicates whether it is operating or dead), the mobile node pings the gateway periodically. Each mobile node periodically sends gateway_STATUS query message to its current gateway and expects a KEEP_ALIVE message as the reply within an RTT amount of time. If no such reply is received, the mobile node broadcasts a gateway_SOLICIT message and waits for a reply from any active gateway for T_2 time. The mobile node keeps retrying for every T_2 time until an active gateway sends a gateway_AD message. This whole procedure is repeated every T_1 units of time. The broadcast messages need to reach beyond a single hop in the ad hoc network; consequently all nodes rebroadcast a broadcast message upon receiving one for the first time.

The number of messages that flow in the network as a result is $n \cdot 1/T_1$ unicast messages per unit time. However, when a gateway is determined to be dead, there will be a flood of n gateway_SOLICIT broadcast messages in the worst case, and n/m broadcast messages for the average case. The replies to gateway_SOLICIT messages are unicast messages and can be up to $(m-1) \cdot n$ messages in the worst case and $(m-1) \cdot n/m$ for the average case.

Mobile initiated and Proactive:

In this case, each mobile node keeps track of all active gateways by periodically pinging all the gateways. When the current gateway that the node is affiliated

[6]All the figures and algorithm pseudo-codes for this section are listed together in Section 5.4.7 towards the end of this chapter.

with does not respond to ping messages, the node switches to one of the gateways that recently responded to its ping messages. The detailed description of this approach is Figure 5.8. This algorithm is similar to the reactive case presented earlier. The main difference lies in the fact that the mobile nodes have to keep track of all the gateways in the network. This is useful for fast recovery, if the current gateway becomes inactive, but at the cost of an increase in the overhead.

The number of messages in the normal case is $m \cdot n \cdot 1/T_1$ unicast messages per unit time. When a failure is detected, the switch-over time to a new gateway is close to zero in most cases, and is the same as for the reactive case when no other gateway responds to gateway_STATUS query message.

Gateway initiated and Reactive:

For the gateway initiated reactive case, each gateway keeps track of the active gateways through a periodic KEEP_ALIVE message exchange between themselves. Each gateway responds to another gateway's KEEP_ALIVE message with a KEEP_ALIVE_ACK message. A particular gateway is labeled as dead only if this gateway does not respond to fixed number of multiple retries of the KEEP_ALIVE message transmissions. When a gateway is found to be inactive, all gateways broadcast a gateway_AD message that includes information about all known gateways that are alive. The nodes previously using the dead gateway, on collecting such messages, can subsequently switch to a new active gateway in its radio range.

In the reactive case, there is a flow of $m \cdot (m-1) \cdot 1/T_1$ unicast messages per unit time and m broadcast messages upon detection of a faulty gateway. The detection time is $T_1/2$ seconds, on the average.

Gateway initiated and Proactive:

For the gateway initiated proactive case, all gateway enabled nodes period-

ically broadcast a gateway_AD message. Each mobile node that receives such a message from a gateway-enabled node decides whether to switch to this new gateway based upon on two parameters: *(i)* hop count, and *(ii)* last time, T_{last} at which a gateway_AD message was received from the current gateway. If the hop count to the new gateway-enabled node is less than the current gateway's hop count to which it is affiliated, or if the condition:

$$T_{last} < 2 \cdot \frac{1}{(\text{gateway\_ad\_frequency})}$$

is true, then the mobile sets the 'better' gateway-enabled node as its new gateway. Figure 5.10 and the associated pseudo-code explains the actions of the mobile node and gateways in further detail.

The proactive solution requires $m \cdot 1/T_1$ number of broadcast messages per unit time, where $1/T_1$ is the frequency of gateway advertisement broadcasts. The switchover time upon detection of a faulty gateway is negligible, but the detection time itself is $T_1 \cdot (3/2)$ units, on the average.

The overhead required for the four solutions presented above can be summarized as shown in Table 5.2.

Table 5.2: Overhead Requirements for Gateway Fault Tolerance Algorithms.

	Mobile Node Initiated	*Gateway Initiated*
Reactive	$(m-1) \cdot \frac{n}{m}$ unicast, $\frac{n}{m}$ broadcast	$m \cdot (m-1) \cdot \frac{1}{T_1}$ unicast, m broadcast
Proactive	$m \cdot n \cdot \frac{1}{T_1}$ unicast	$m \cdot \frac{1}{T_1}$ broadcast

Best Solution:

Based on the number of messages exchanged, detection time and switchover times, for each of the four algorithms presented above, the gateway initiated

solutions are better since they can make use of the connectivity through the overhead satellite network, and also present low overhead on the communication infrastructure in the ground-based ad hoc network. In particular, all other things being equal, the gateway initiated and reactive solution is best because all the messaging is only between gateways.

5.4.4.2 Gateway Selection Algorithm

The general flowchart of the gateway selection algorithm is shown as Figure 5.11. The precondition for this algorithm is that every gateway knows the list of all other gateways in the ad hoc network. We will use the same KEEP_ALIVE based mechanism as described in previous sections. The properties of the gateways we assume in this system are: (i) the gateways are synchronized (almost synchronized because of accessibility to GPS for all gateways), (ii) the messages may be lost, and (iii) not all gateways are reliable.

The algorithm has two phases. During the initialization phase, each gateway-enabled node looks up its list of all other known gateway-enabled node in its domain and queries them regarding status (of being dead or alive). Based upon the status, each gateway-enabled node forms the list of the 'voting gateways'. In case there is a malicious gateway among this group of voting nodes, then at least 4 gateway nodes are required to be alive, otherwise the impossibility situation of the Byzantine Generals Problem occurs [49]. In this case, the default operation mode would be for each gateway-enabled node to assume the role of being the gateway to all the nodes within one hop distance of itself (nodes that are within one hop distance of more than one gateway decides locally which gateway node's domain to join). The domain is thus partitioned in this mode (to minimize the data manipulation activities of the malicious node), and the command and

control center, and all the nodes in the domain are informed of the impossibility situation. If necessary, then depending upon the application scenario, the nodes may choose to stop transmitting all sensitive inter-domain data altogether.

If the impossibility situation mentioned above does not occur, and more than 4 gateway nodes are alive, then each gateway-enabled node waits a random time, T, between 0 to T_{1max} It then generates its vote as to who the next gateway should be from among the list of the voting gateways, excluding the current gateway as a possible candidate. This vote is then signed using the underlying encryption technology that is being used, and transmitted to all the other gateways in the voting list. This is done for all the members of the voting list and each gateway sends its message and collects the messages sent by other gateways. Since the communication path between gateways can have intermittent failures, ACK-based mechanism is used (e.g. IEEE 802.11) to increase the reliability of message exchange in each round. If more than two-thirds of the votes are received by each node before timeout condition T_{2max}, then each gateway performs a count of the majority vote and if a clear winner emerges, then switches to this new gateway. All the mobile nodes are then informed of this decision by all the gateways broadcasting SELECTED_gateway messages. If either a timeout occurs or no majority winner emerges, then the algorithm is repeated (up to a software settable number of times, depending upon the application requirements). However, if two-thirds of the votes are not received by the timeout condition, then once again, the impossibility situation of the Byzantine Generals Problem occurs and the domain-partitioned default mode of operation is invoked.

The gateway selection algorithm is periodically executed (with large enough timeout periods $T_{1max}$and T_{2max} to prevent too frequent gateway changes). Since all gateways are almost synchronized, each gateway knows when the selection

algorithm needs to be (re)started. It is interesting to note that this algorithm may also be used if the currently active gateway is destroyed or its satellite link is blocked. Thus, it can be used as another variation (albeit inefficient) of the gateway fault recovery system described earlier.

5.4.5 Implementation of the Gateway Reliability Algorithms

We have simulated the algorithms for gateway fault tolerance, outlined in Section 5.4.3, on the ***ns-2*** network simulation platform, version 2.1b8 [95]. This version contains modules that implement different wireless network protocols, such as DSDV, DSR, TORA and AODV. Our simulation setup is similar to that shown in Figure 5.12, where the ad hoc network component consisted of five mobile nodes. Some of the mobile nodes are designated as gateways, and are labeled as BSx in the figure. They serve to connect wireless ad hoc network to the other components of NGI. The Control Operations Center (COC) is divided into several components, labeled as COCx. Each normal mobile node, labeled as Nx, has a TCP connection with corresponding COCx. Depending on the gateway BSx that a mobile node Nx is using at any time instant, all links from COCx to all gateways in ad hoc network are disabled, except the link to BSx. This forces the packets from COCx to Nx to flow through the gateway BSx. A single COC cannot be used in our ***ns-2*** setup because packets to different mobile nodes need to be routed through different gateways. In a real-life setup, the COCs would normally be low earth orbiting satellites, and would have some satellite network handoff mechanism to decide about which gateway to use.

For preliminary experiments, we performed gateway selection in a round-robin fashion. In this method, every node in the ad hoc component knows which gateway will be active at any time instance in advance, according to a fixed time

look-up table. There is no data overhead incurred in gateway selection and no overhead in the mobile node configuration. The only overhead is due to the abrupt change in switching from old to new gateway, during which time some packets may be lost as new routes are established. We compared this case with *best base-station* approach. In best base-station approach, each mobile node is periodically configured to use the nearest gateway (nearest in number of hops).

The experimental setup contained 100 mobile nodes in the ad hoc network, moving in a rectangular, bounded region of 700mx700m size, at an average speed of 40 m/s. We generated the movement scenario file using CMU's *setdest* tool. We implemented *fault-tolerant agents*, referred to as *ftagents*, which runs on each node of the ad hoc network. The ftagents on gateways keep track of other gateways, participate in gateway selection and perform the task of informing all mobile nodes. The ftagents on nodes receive messages from gateways and configure the node to use the selected gateway for communicating with the nodes not in same domain.

The overall throughput for different numbers of gateways, with DSDV as the underlying routing protocol, is shown in Figures 5.13 and for DSR is shown in Figure 5.14. In each of the results graphs, we have compared the performance of the *best base-station* approach to the *round-robin* case. The main point to note from the plots is that the change in the overall throughput is not very significant for the fault tolerance algorithms, implying that the overhead and computational load incurred are bearable for the fault tolerance objectives for this system.

5.4.6 Concluding Remarks

Thus far, we have outlined algorithms that provide a measure of fault tolerance and security in the operation of gateways in a hybrid satellite, mobile multi-hop

network environment. Based on the number of messages exchanged, detection time and switchover times, for each of the four algorithms presented above, the gateway initiated solutions are better since they can make use of the connectivity through the overhead satellite network, and also present low overhead on the communication infrastructure in the ground-based ad hoc network.

Simulation results obtained by studying some representative cases on the ***ns-2*** simulation platform implied that the algorithms are practical and implement-able, with only a marginal increase in overhead and complexity.

In summary, we have thus demonstrated that the reliability and dependability of critical information processing nodes in hierarchical networks such as the NGI can be enhanced by means of relatively simple fault tolerant, secure protocols. The techniques employed also enabled the network to be self-recoverable, and to have a graceful degradation in the network performance in the face of failures—either unintentional or malicious. These techniques can also be applied to any hybrid satellite/mobile ad hoc network that is deployed in a terrain with blockages and communication impairments (e.g. tactical applications involving divisions of troops communicating with mobile gateways in harsh fading/jammed communication environments with blockages and high failure hazards). Other applications can be in distributed robotic platforms, e.g. in terrestrial or extraterrestrial exploration, where mobile nodes have to operate in possibly inhospitable terrain (outer space, planetary or earth remote locations) for remote sensing, data collection, control applications, etc. The gateways in such applications can serve as the central data collection point for various mobile nodes, which are then relayed to appropriate LEO or GEO points. Thus the gateways in these scenarios have to be ultra-reliable and secure.

Commercial applications are also possible in industrial settings where mobile

sensors or systems are needed to monitor or control a distributed process (e.g. physical packages being automatically routed and delivered, raw materials being handled remotely in the process line-up of a manufacturing plant etc.) or for security applications, and situations where the information from mobile robotic sensors are aggregated at a single concentrator node or gateway. Dependability through redundant gateways (and hence the associated gateway dependability algorithms) are crucial in such applications as well.

Future work involves determining the response time of the algorithms to failure events for a variety of settings. In particular, for large scale or sense sensor networks (greater than hundreds of nodes), the scaling behavior of the algorithms are of interest. Some form of hybrid authentication/security and fault tolerance schemes should also be explored to enable efficient, resilient network architectures.

5.4.7 Flowcharts and Figures

Algorithm 5.3. Mobile Initiated Reactive

```
1: if Mobile Node then
2:    number of tries = 0
3:    Send a GATEWAY_STATUS message to current gateway
4:    Set timer for RTT time
5:    Receive a timeout interrupt or message
6:    if KEEP_ALIVE message comes from current gateway then
7:       Cancel timer
8:       Goto step 26
9:    else if timeout interrupt then
10:      if number of tries < max number of tries then
11:         number of tries ++
12:         goto step 3
13:      else
14:         current gateway = NONE
```

```
15:         Broadcast a GATEWAY_SOLICIT message
16:         Set timer for T2 time
17:         Receive a message or timeout interrupt
18:         if  GATEWAY_AD received from a gateway G then
19:           current gateway = G
20:           Cancel timer
21:         else if timeout interrupt then
22:           goto step 15
23:         end if
24:       end if
25:     end if
26:     Sleep for T1 time
27:     goto step 2
28: end if
```

```
1: if Gateway then
2:   Send KEEP_ALIVE message to the sender of GATEWAY_STATUS mes-
     sage
3:   Send GATEWAY_AD message to the sender of GATEWAY_SOLICIT mes-
     sage
4: end if
```

Algorithm 5.4. Mobile Initiated Proactive

```
1: if Mobile Node then
2:   number of tries = 0
3:   Set of responding gateways G = NULL
4:   Send a GATEWAY_STATUS message to all known gateways
5:   Set timer for RTT time
6:   Receive a timeout interrupt or message
7:   if KEEP_ALIVE message comes from gateway g then
8:     if current gateway == g then
9:       Cancel timer
10:      Goto Step 38
11:    else
12:      G = G ⋃ {g}
13:      Goto Step 6
14:    end if
15:  else if timeout interrupt then
16:    if number of tries < max number of tries then
```

```
17:         number of tries ++
18:         Send a GATEWAY_STATUS message to current gateway
19:         goto step 4
20:       else
21:         current gateway = NONE
22:         if S ≠ NULL  then
23:           current gateway = choose one from set S
24:           Goto step 38
25:         else
26:           Broadcast a GATEWAY_SOLICIT message
27:           Set timer for T2 time
28:           Receive a message or timeout interrupt
29:           if  GATEWAY_AD received from a gateway G then
30:             current gateway = G
31:             Cancel timer
32:           else if timeout interrupt then
33:             goto step 26
34:           end if
35:         end if
36:       end if
37:     end if
38:     Sleep for T1 time
39:     goto step 2
40: end if
```

```
1: if Gateway then
2:   Send KEEP_ALIVE message to the sender of GATEWAY_STATUS mes-
     sage
3:   Send GATEWAY_AD message to the sender of GATEWAY_SOLICIT mes-
     sage
4: end if
```

Algorithm 5.5. Gateway Initiated Reactive

```
1: if Gateway then
2:   S = set of all gateways
3:   number of tries = 0
4:   Set of gateways that acknowledge S' = NULL
5:   Send a KEEP_ALIVE message to all gateways in set S
6:   Set timer for T2 time
```

```
7:    Receive a message or timeout interrupt
8:    if received a KEEP_ALIVE_ACK message from gateway g then
9:      S' = S' ⋃ {g}
10:     Goto Step 7
11:   else if timeout interrupt then
12:     if S == S' then
13:       Goto step 23
14:     else if number of tries < max number of tries then
15:       number of tries ++
16:       Send a KEEP_ACK message to gateways in set (S-S')
17:       goto step 5
18:     else
19:       S = S'
20:       Broadcast a GATEWAY_AD message including G in the message
21:     end if
22:   end if
23:   Sleep for T1 time
24:   goto step 3
25: end if
```

```
1: if MobileNode then
2:   Receive a GATEWAY_AD message from gateway G with gateway set S
3:   if  current gateway ∈ S then
4:     Goto step 2
5:   else
6:     current gateway = G
7:   end if
8:   Goto step 2
9: end if
```

Algorithm 5.6. Gateway Initiated Proactive

```
1: if Gateway then
2:   Broadcast a GATEWAY_AD message
3:   Sleep for T1 units of time
4:   Goto step 2
5: end if
```

```
1: if Mobile Node then
2:   Receive a GATEWAY_AD message from gateway G
```

3: **if** G == current gateway **then**
4: T_{last} =current time
5: **else if** (hopcount(G) < hopcount(present gateway)) OR (T_{last} < current time $-2 * T1$) **then**
6: current gateway = G
7: **end if**
8: Goto step 2
9: **end if**

Algorithm 5.7. Send Collect Msgs (message m, Gateway Set S)

1: Collected messages set C = NULL
2: start_time = present time
3: Set of gateways that acked S' = NULL
4: C = C $\bigcup$ {m}
5: Send m to all gateways in S
6: Set timer for T1
7: Receive a timeout interrupt or message
8: **if** received a message m' from gateway g' **then**
9: C = C $\bigcup$ {m'}
10: **if** C contains messages from all gateways in S **A**ND S == S' **then**
11: return S, C
12: **end if**
13: Goto Step 7
14: **else if** received an ack message from gateway g' **then**
15: S' = S' $\bigcup$ {g'}
16: **if** S' == S **then**
17: Cancel the timer
18: **if** C contains messages from all gateways in S **then**
19: return S, C
20: **else**
21: Set timer: (start_time + T2 - present time); Goto Step 7
22: **end if**
23: **end if**
24: **else if** timeout interrupt **then**
25: **if** S − S' ≠ NULL **then**
26: **if** number of tries < max number of tries **then**
27: number of tries ++
28: Send m to all gateways in (S − S'); Goto step 6
29: **else**

```
30:         Label all gateways in set (S − S') as dead; S = S'
31:         if C contains messages from all gateways in S then
32:           return S, C
33:         end if
34:         Goto Step 7
35:       end if
36:     else
37:       Label all gateways whose messages does not appear in C to be dead
38:       S = S - {dead gateways}; return S, C
39:     end if
40: end if
```

Algorithm 5.8. Check Consistency (message set C)

```
1: Suppose C = {C_1, C_2, C_3, ..., C_m}
2: G = NULL
3: R = NULL
4: for all gateway g s.t. g's message is in at least one c ∈ C do
5:   if g's message is in at least f * |C| number of c's in C then
6:     G = G ⋃ {g}
7:     R = R ⋃ {g's message}
8:   end if
9: end for
10: return G, R
```

Algorithm 5.9. Select Gateway (gateway set G)

```
1: Choose a random number r
2: Calculate MD5 hash(r) = h
3: G,c = SendCollectMsgs(h, G) /* Phase 1 Round 1 */
4: G,C' = SendCollectMsgs(c, G) /* Phase 1 Round 2 */
5: G, H_SET = CheckConsistency(C')
6: G,D = SendCollectMsgs(R, G) /* Phase 2 Round 1 */
7: G,D' = SendCollectMsgd(D, G) /* Phase 2 Round 2 */
8: G, R_SET = CheckConsistency(D')
9: for all g ∈ G do
10:   if MD5Hash(r_g) ≠ h_g then
11:     G = G - {g};
12:   end if
```

13: **end for**
14: g_{sel}=SelectionFunction(G, R_SET)
15: Broadcast ELECTED_GATEWAY message including g_{sel}
16: Wait for T time
17: Goto step 1

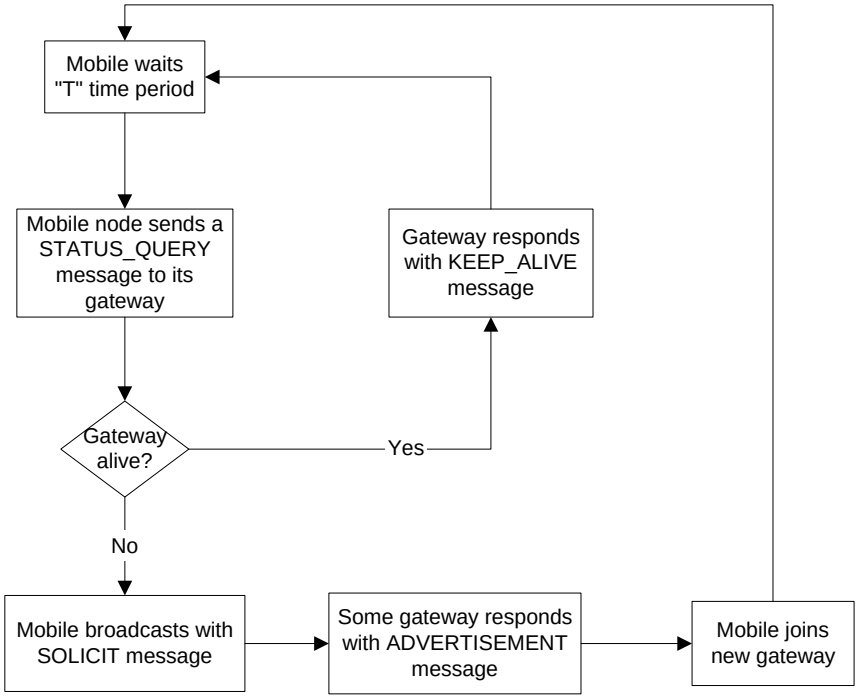

Figure 5.7: Flowchart for mobile initiated reactive fault-tolerance scheme

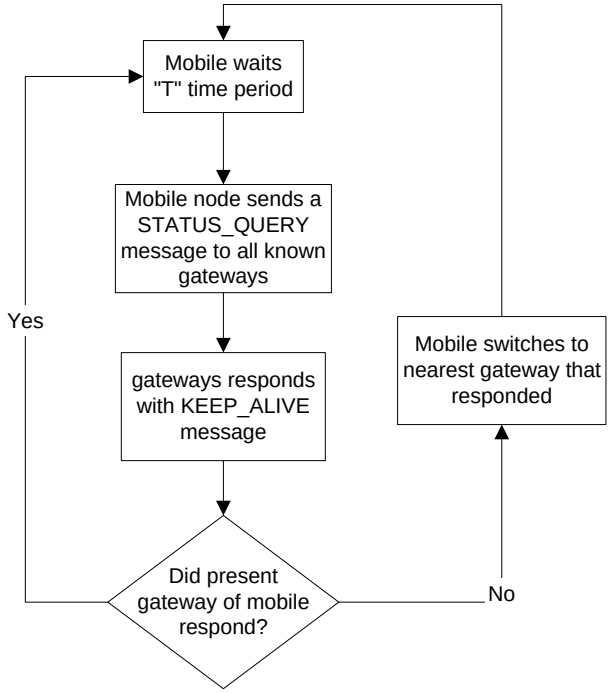

Figure 5.8: Flowchart for mobile initiated proactive fault-tolerance scheme

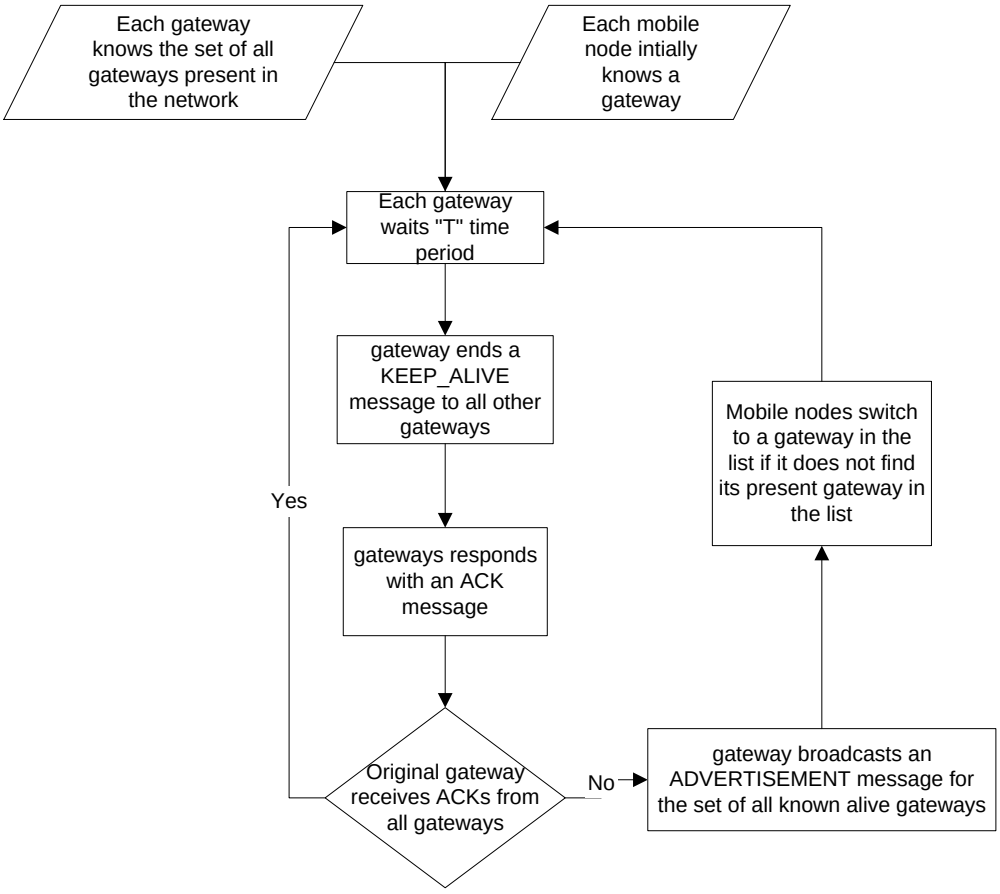

Figure 5.9: Flowchart for gateway initiated, reactive fault-tolerance scheme

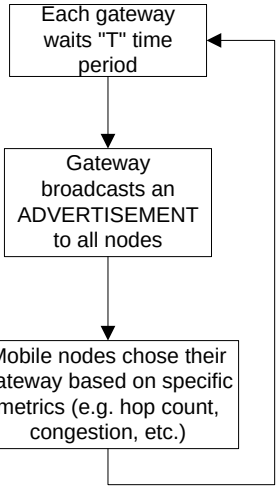

Figure 5.10: Flowchart for gateway initiated, proactive fault-tolerance scheme.

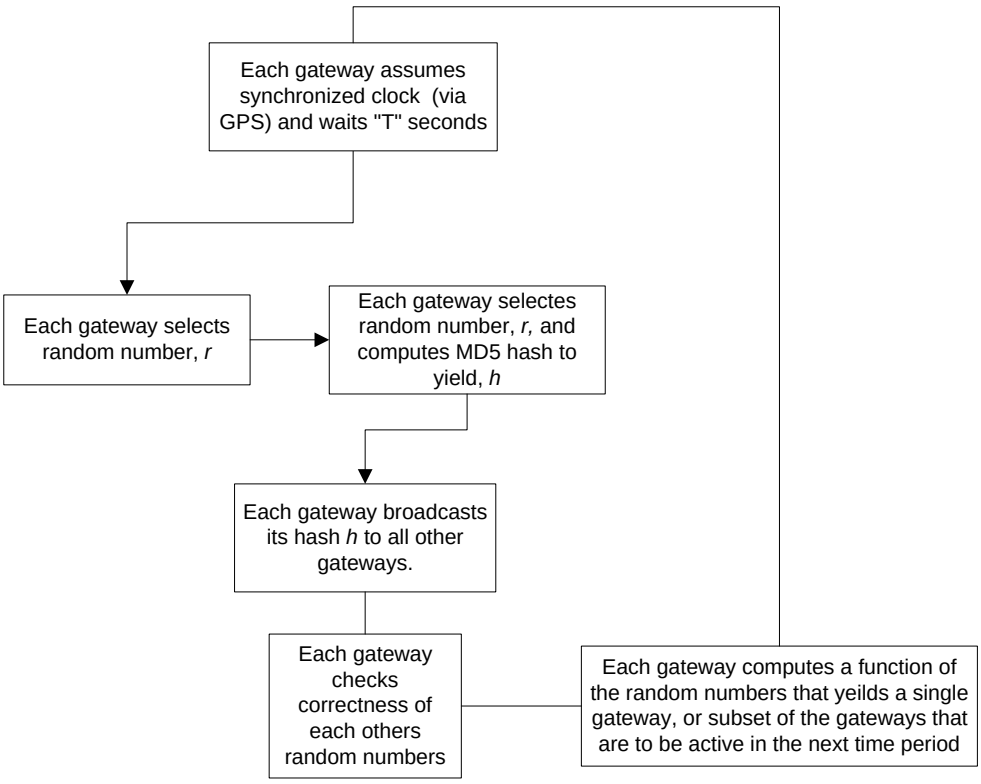

Figure 5.11: Secure gateway selection algorithm.

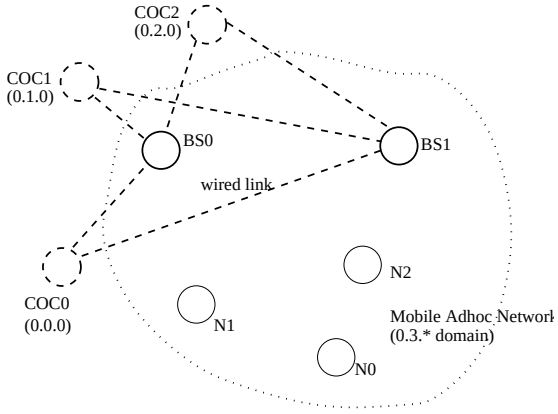

Figure 5.12: Node/gateway layout used for testing fault-tolerance algorithms.

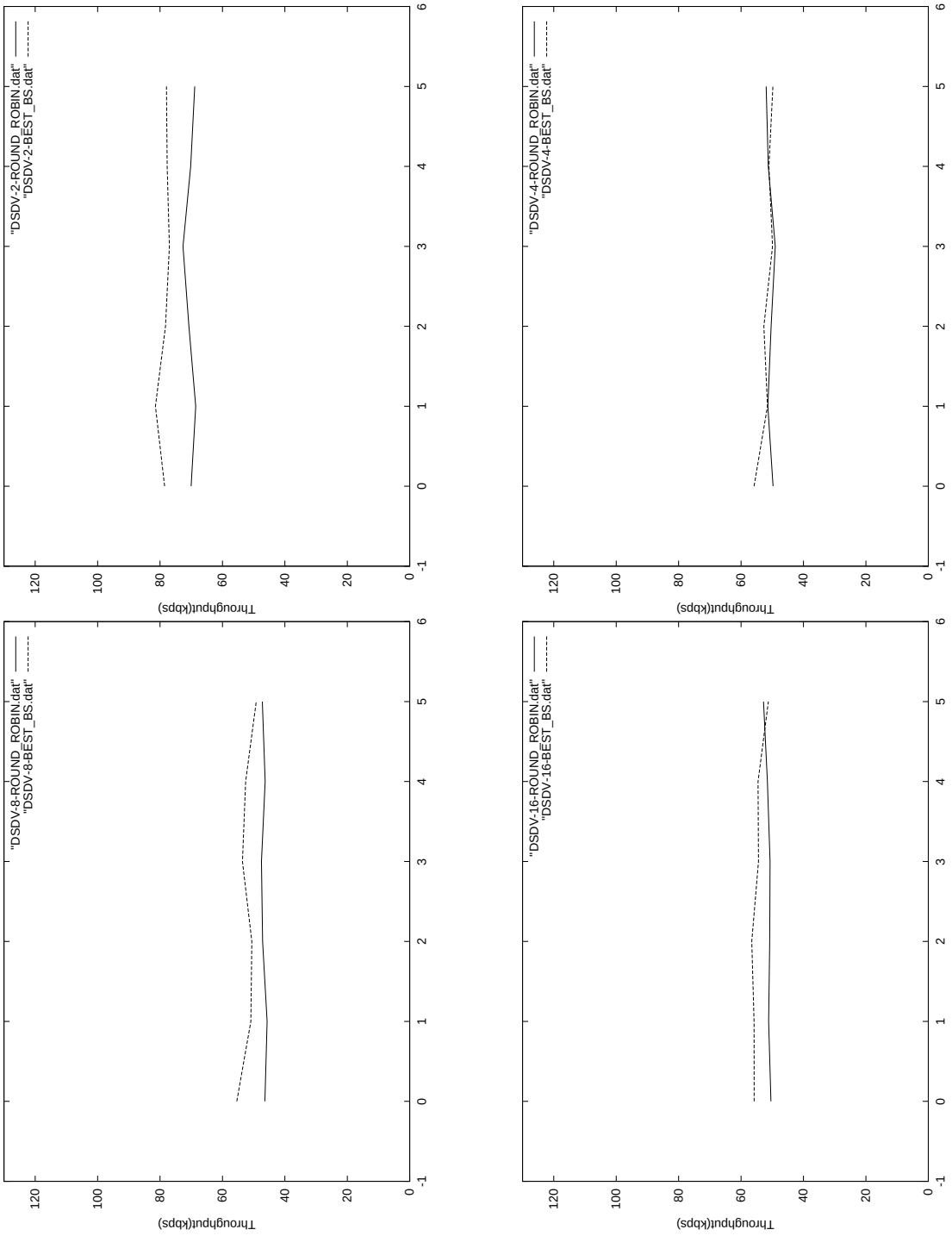

Figure 5.13: Overall throughput in best basestation vs. round-robin gateway selection algorithms for the DSDV routing protocol, as a function of number of gateways in domain.

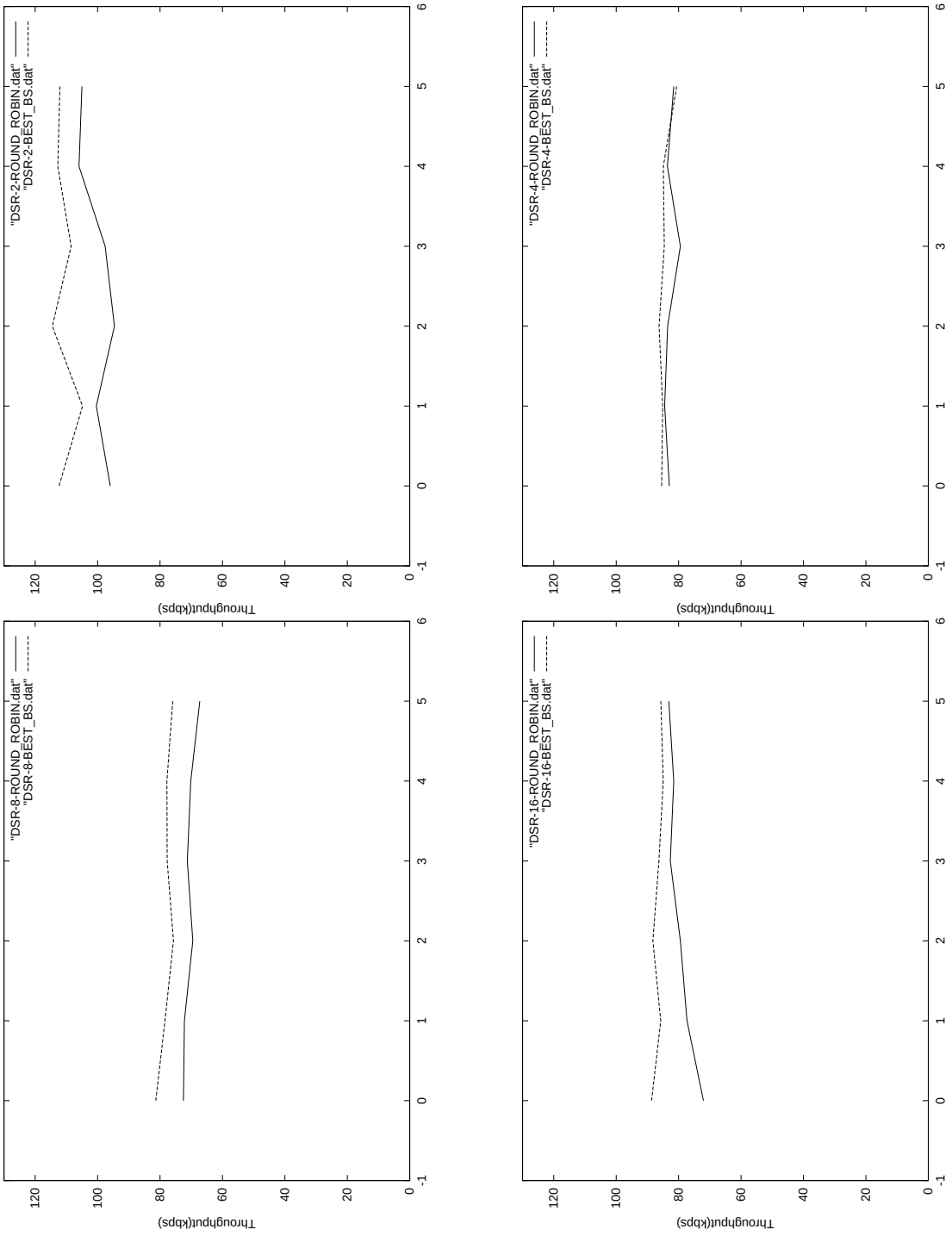

Figure 5.14: Overall throughput in best basestation vs. round-robin gateway selection algorithms for DSR routing protocol, as a function of number of gateways in domain.

CHAPTER 6

Rate Adaptive MIMO OFDM LDPC transceiver

The final chapter of this thesis deals with a physical layer issue for wireless nodes: the design of a rate adaptive transceiver based on multi-input, multi-output (MIMO) antenna technology, together with some modern modulation and coding support. The interesting issue about this development is regarding how information is processed in such a device. It is seen that, in the spirit of Chapter 2 of this thesis, the fundamental techniques that allow signals to be separated in such devices, and thus allows MIMO to realize its full potential, is a form of data fusion. The multiple transmit/receive channels can be thought of as independent sensor nodes observing independent data, and the challenge is, again, one of efficient data fusion. Even more interestingly, the techniques that have been developed over the years in this discipline, and the modified technique that is proposed, are seen to be case specific implementations of the likelihood fusion principles discussed in earlier chapters.

The introductory section below outlines the objectives and approaches undertaken for designing the new radio, and provides a roadway for the subsequent sections.

6.1 Introduction

It was mentioned at the outset that the objective of communication devices can be generalized to be the acquisition, processing and dissemination of information. However, with the exponential rise in the use of wireless devices, the time, space and frequency dimensions that are necessary to enable all these devices to operate is becoming crowded. Secondly, as discussed earlier, if the maximum benefit is to be obtained from these systems, then they need to be networked in a cooperative communication configuration, and this in turn is placing huge demands on the available bandwidth. Finally, for most applications involving wireless devices nowadays, the channel conditions under which these devices are supposed to operate severely limit the effectiveness of the underlying physical layer technology. Most wireless channels have very detrimental effects on radio frequency propagation of communication signals, and essentially constricts effective bandwidth.

All these constraints have necessitated the development of highly bandwidth efficient techniques for the physical data transmission/reception functions for wireless systems of all varieties (sensors, ad hoc node, etc.). In this chapter of the thesis, an attempt is made to alleviate these problem by presenting a novel approach that maximizes the raw spectral efficiency of transceivers. This is accomplished by using a combination of three technologies to form a space-time coded transceiver that can be adaptively optimized to extract the maximum diversity from the wireless channel. These technologies are: multi-input, multi-output antenna technology (MIMO), orthogonal frequency domain multiplexing (OFDM), and the powerful low density parity check channel codes (LDPC)—outlined in Sections 6.2.1, 6.2.2, and 6.2.3, respectively.

It has been shown that MIMO technology is the only means by which to

improve upon the Shannon's hard limit [81] for the capacity of a single antenna system, also known as single input, single output (SISO) . Over the last few decades, steady progress has been made in digital communication theory and practice to the point where the Shannon limit has already been approached using modern single antenna based coding and modulation techniques [22]. At this stage, to extract further spectral gains from the wireless channel, MIMO appears to be the only viable solution. OFDM, on the other hand, is a multi-carrier modulation technique which has only recently become practically useful with the advent of powerful FFT and IFFT chipsets [4]. This allows independent data bits to be sent over smaller individual bins in the frequency spectrum, and thus counteracts the frequency selective fading effects of the wireless channel, without the expense and complexity of equalizers. Using OFDM, it then becomes possible to undo the severe effects of the Rayleigh channel and make the channel appear more like an AWGN channel. Finally, the newly re-discovered LDPC codes are powerful channel codes, with performance similar to (or better than) the best Turbo codes, but having much simpler coding/decoding structures by nature of being a block code. Over harsh fading channels, it has been shown to achieve impressive coding gains [56].

Our novel approach has been to combine the best elements of these three technologies to present a smart, 'spectrum-aware', software defined transceiver. The basic block diagram of the transceiver is shown as Figure 6.1.

In particular, a novel multi-antenna signal separation scheme is proposed, which uses likelihood decoding of LDPC streams that are transmitted over independent antennas (Section 6.4.2). Good convergence has been observed in simulation studies to date. This is coupled with a simplified MIMO channel estimation scheme that can be incorporated as part of a robust system design

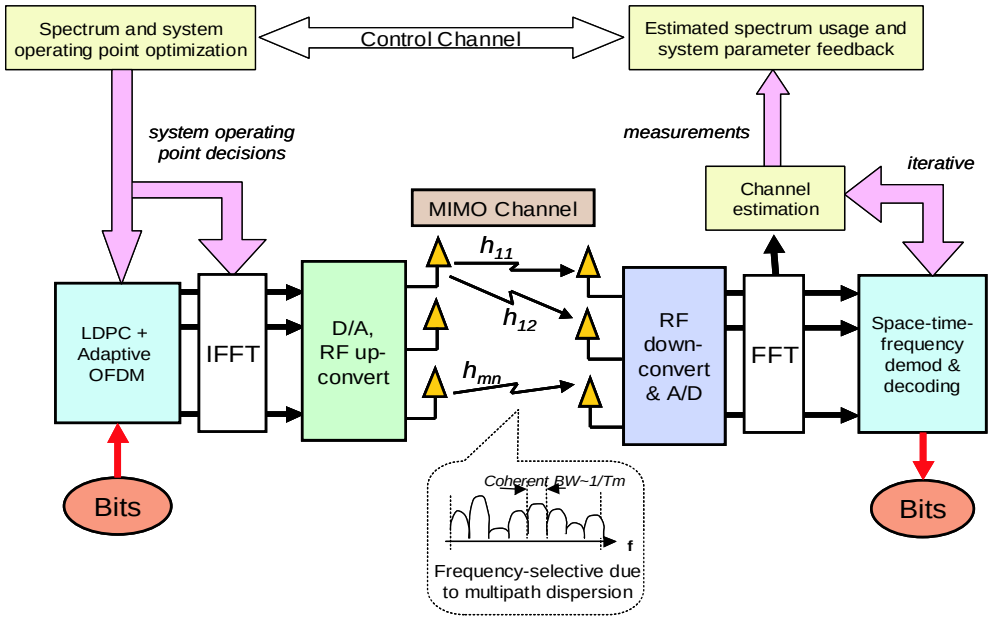

Figure 6.1: Block diagram of proposed spectrally efficient MIMO-OFDM-LDPC transceiver.

(Section 6.3). The modifications that would be necessary to enable extensions of the technique to higher order signal constellations are also discussed (Section 6.4.3). Finally, an outline is provided regarding techniques that can be used to make the system "spectrum-aware", i.e. adaptive to changes in the channel and network conditions (Section 6.5), and some issues regarding system design and implementation are also discussed with some simulation results.

To begin with, in Section 6.2, a brief introductory background is provided on the current state-of-the-art in LDPC, OFDM and MIMO technologies.

6.2 Primer on MIMO, OFDM and LDPC

6.2.1 Multiple Input, Multiple Output (MIMO) Systems

Broadband connectivity has long been considered the future of the telecommunications industry. 'Broadband' nowadays typically refers to data services operating at data rates in excess of 1.544 Megabits per second (Mbps), which is known as the *T-1* rate. However, with the explosive growth in the availability and use of the wireless channel, the available radio frequency spectrum for wireless applications has gradually become crowded. Thus, innovative technologies are now required for optimum spectrum utilization and interference endurance, while maintaining low-complexity for deployable wireless systems.

This is especially true for mobile ad hoc network (MANET), where low-power, resource allocation and re-use are of prime concern. In light of these developments, it has recently been recognized that Multi-input-multi-output (MIMO) antenna-based radio systems provide marked capacity gain and interference-combating potential, with manageable system complexity. This is because MIMO systems with appropriate modulation and coding can better exploit the Space-Time-Frequency diversity of the wireless channel for maximum spectral efficiency (measured as the user capacity: bits/sec/Hz/Area, or some physical layer attribute such as bit error rate (BER), outage rate, etc.).

There are also sound system engineering reasons for preferring MIMO systems as the backbone technology for broadband applications. This is because higher data rates through single antenna, single carrier systems require expensive equalizers/codecs to combat the effects of intersymbol interference (ISI). Multi-carrier MIMO systems, however, can overcome this constraint, but at the expense of increased RF processing costs. But historically, the cost of RF processing has

grown slower than baseband processing cost as data rates have increased. In addition, spatio-temporal modems can operate at rates well below their theoretical limit, while still providing the required throughput, whereas traditional modems have already been forced to operate in close-to-theoretical operation regions (Figure 6.3). Therefore, for these combined reasons, MIMO based spatial-temporal technology is inevitable for high bandwidth wireless systems.

It has been well known for decades that multiple *receive* antennas can provide improved resolution and detection estimates/decisions than a single antenna receiver. As a matter of fact, this feature is simply a special case of the information processing concepts that were explored in depth in Chapter 2 in the context of wireless sensor networks. It was seen there that data pooled from multiple sources (in this case, from multiple antennas) and appropriately combined yield superior observation and detection statistics. However, the fact that using multiple *transmit* antennas—in addition to multiple receive antennas—can dramatically improve channel capacity is a fairly recent practical discovery [22]. The pioneering work demonstrating that a transmitter with N transmit antennas and a receiver with M receive antennas ($M \geq N$) can achieve a $min\{N, M\}$-fold capacity increase, without any increase in bandwidth occupation and signal power, was shown by researchers at AT&T Bell Labs in 1995 with their BLAST MIMO system [23]. This has, in turn, spurred a huge research interest in space-time processing transceivers.

The basic theory is intuitive. Transmitted signals from N multiple antennas form a wavefied, $s(t, \boldsymbol{r})$ in time and space (Figure 6.2). Properly designed MIMO systems can sample this wavefield space at multiple spatial and temporal points (the M receive antennas), and thus achieve diversity order up to $N \times M$. But the sampled signals are uncorrelated only when there are radio frequency scatterers in

the domain. This ensures that the M receive antennas observe wavefronts that have arrived from randomly different paths, and that are sufficiently rich and different from each other to enable the original N data streams to be separated at the receiver. Thus, this technology is ideal for dense, urban settings where RF scatters abound. This is also the only way to overcome the Shannon capacity bound for SISO systems operating over AGWN channels [81] (Figure 6.3).

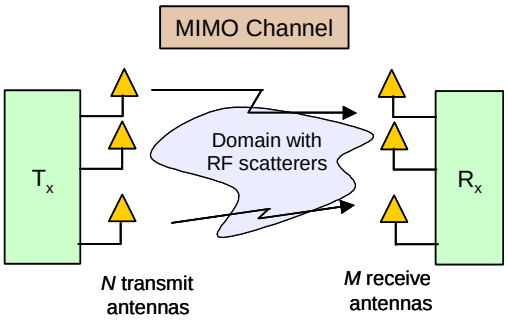

Figure 6.2: MIMO transceiver in the presence of RF scatters.

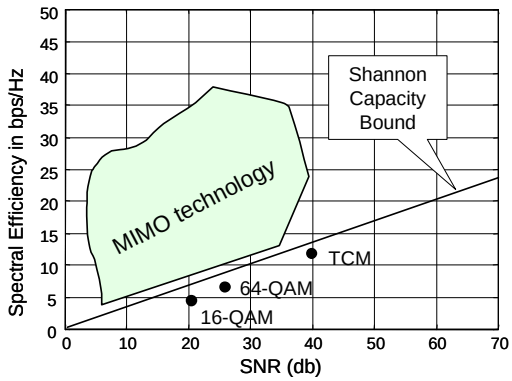

Figure 6.3: Capacity potential for MIMO transceivers in RF cluttered environment.

There is extensive mathematical and quantitative verification of the intuitive theory mentioned above as to why MIMO systems yield capacity improvements over SISO systems. In order not to digress, a detailed discussion of the mechanics of MIMO capacity computations is omitted from this thesis, and the reader is referred to the voluminous published literature in the field [12, 22]. We only highlight the main result from this research. We recall that for a SISO system operating in an AWGN channel, the capacity is given by the well-known Shannon's formula for channel capacity in bits/sec/Hz [81]:

$$C = \log_2(1 + \Gamma) \tag{6.1}$$

where Γ is the signal to noise ratio. However, in a MIMO system with N transmit and M receive antennas, the capacity increases to:

$$C_{MN} = \log_2 \det\left(\mathsf{I}_N + \frac{\Gamma}{N}\mathsf{H}\mathsf{H}^*\right) \tag{6.2}$$

$$= \log_2 \det\left(\mathsf{I}_M + \frac{\Gamma}{N}\mathsf{H}^*\mathsf{H}\right) \tag{6.3}$$

$$= \sum_{i=1}^{N} \log_2\left(1 + \frac{\Gamma}{N}\lambda_i\right) \tag{6.4}$$

where H is the $M \times N$ channel propagation (mixing) matrix (discussed in further detail in Section 6.3), λ_i are the eigenvalues of the $M \times M$ matrix $\mathsf{H}\mathsf{H}^*$, I is the identity matrix, and $\det(A)$ is the determinant of matrix A. We note that the capacity of an $M \times N$ system is the same as an $N \times M$ and, therefore, this expression be used for overloaded arrays where there are fewer receiving antennas than transmitting antennas. For symmetric systems with $M = N$, the capacity is thus a function of N, the SNR, and the eigenvalues λ_i.

If the propagation is line of sight (LOS) and there is little or no multipath, then the majority of the eigenvalues λ_i will be insignificant. In the limiting case of one dominant eigenvalue, the MIMO capacity expression simplifies to the following lower bound:

$$C_{NN}^{min} = \log_2(1 + N\Gamma) \tag{6.5}$$

which represents the case when there is maximum correlation between the receiving antennas and H has only a single degree of freedom. In this case, the capacity increases only logarithmically with the number of antennas.

However, if the domain of the MIMO system has many scattering surfaces, then many multipath components are generated as a result and the receiving antennas observe signals from many different angles. In this case, upto N of the eigenvalues λ_i can be significant, and the channel matrix H can have upto N

degrees of freedom. The correlation between the receiving antennas is thus low. In the limit, if H is an orthogonal matrix with HH^* being the identity matrix, then all the N eigenvalues are equal to N, and the resulting upper bound on capacity is given by:

$$C_{NN}^{max} = N \log_2(1 + \Gamma) \tag{6.6}$$

So, for multipath rich environments, the theoretical limiting MIMO capacity increases linearly with the number of antennas (while the total amount of transmitter power is fixed and divided equally among the transmitting antennas). This is the fundamental motivation that for using a MIMO based system for our proposed transceiver.

6.2.2 Orthogonal Frequency Domain Multiplexing (OFDM)

As mentioned earlier, the mobile radio channel is characterized by multipath fading environment. The signal at the receiver contains a large number of reflected radio waves that arrive at the receiver at different times. However, for broadband multimedia mobile communication systems, it is nowadays necessary to use high-bit-rate transmission of at least several megabits per second. Unfortunately, if digital data is transmitted at this rate, the delay time of the reflected, delayed waves exceeds 1 symbol time for most urban settings. These reflected waves interfere with the direct line of sight reception (if there is one) and causes ISI, or frequency selective fading. Such frequency-selective fading is a dominant impairment in mobile communications. Fading reduces receive signal-to-noise ratio and degrades the bit-error-rate (BER).

To combat frequency-selective fading, the effects of ISI must be eliminated. There are several ways to accomplish this. Adaptive *equalization* techniques at the receiver can be used, but in practice, achieving this equalization at several

megabits per second is difficult and expensive. The alternative technique is to use *multicarrier* transmission techniques, in particular orthogonal frequency domain multiplexing (OFDM) [4].

The basic idea of OFDM is to divide the available spectrum into several subchannels, and to transmit independent carriers in each subchannel. By making all the subchannels narrowband, they experience almost flat fading, which makes equalization unnecessary or very simple. However, if traditional "brick-wall" [74] type filters are used to separate the subchannels, then there is considerable inefficiency in packing the subchannels, with their guard bands, into the available bandwidth. Instead, to obtain high spectral efficiency, the frequency response of the subchannels are chosen to be orthogonal so that they may overlap in the frequency domain. Hence the name OFDM. The orthogonality is maintained even when the composite signal passes through a time-dispersive channel by introducing a cyclic prefix.

The frequency domain overlapping is most easily accomplished by means of digital signal processing techniques, specifically fast Fourier transforms (FFT) and inverse FFTs, rather than by frequency synthesizers.[1] So, if $D_0, D_1, \ldots, D_{N-1}$ are the data symbols, then the discrete Fourier transform (DFT) can be used as a linear transformation to map the complex data symbols to OFDM symbols $d_0, d_1, \ldots, d_{N-1}$ such that:

$$d_k = \sum_{n=0}^{N-1} D_n \mathrm{e}^{j2\pi n \frac{k}{N}} \tag{6.7}$$

[1]Incidentally, it is only recently that very fast FFT hardware processors that enable real-time OFDM operations have become widely available; and this is the primary reason why multicarrier systems have gained in popularity.

The linear mapping can represented in matrix form as:

$$\boldsymbol{d} = \mathsf{W}\boldsymbol{D} \tag{6.8}$$

$$\text{where} \quad \mathsf{W} = \begin{pmatrix} 1 & 1 & \cdots & 1 \\ 1 & W & \cdots & W^{N-1} \\ 1 & W^2 & \cdots & W^{2(N-1)} \\ \vdots & & & \\ 1 & W^N & \cdots & W^{N(N-1)} \end{pmatrix} \tag{6.9}$$

$$\text{and} \quad W = \mathrm{e}^{j2\pi\frac{1}{N}} \tag{6.10}$$

It can be shown that the time domain representation of the OFDM signal including windowing effect is [4]:

$$x(t) = \sum_{l=-\infty}^{\infty} \sum_{k==k_1}^{N=k_2} \sum_{n=0}^{N-1} \left\{ D_{nl} \mathrm{e}^{j2\pi\frac{n}{N}k} \right\} w\left(t - \frac{k}{f_s} - lT\right) \tag{6.11}$$

where D_{nl} represents the n^{th} data symbol transmitted during the l^{th} OFDM block, f_s is the D/A data rate, $w(t)$ is the pulse shaping window and k_1, k_2 are the cyclic pre/postfix lengths. A diagram illustrating the operation and the resulting time and frequency domain waveforms are shown in Figure 6.4

The demodulation operation is essentially a forward Fourier transform, together with some signal processing for cycle prefix removal, synchronization, etc. We can write the received signal for a time-varying random channel as:

$$r(t) = \int_0^{\infty} x(t-\tau)h(t,\tau)\,\mathrm{d}\tau + n(t) \tag{6.12}$$

The received signal is sampled at $t = k/f$ for $k = \{-k_1, \ldots, N + k - 2 - 1\}$. With non inter-block interference, and assuming the windowing function satisfies

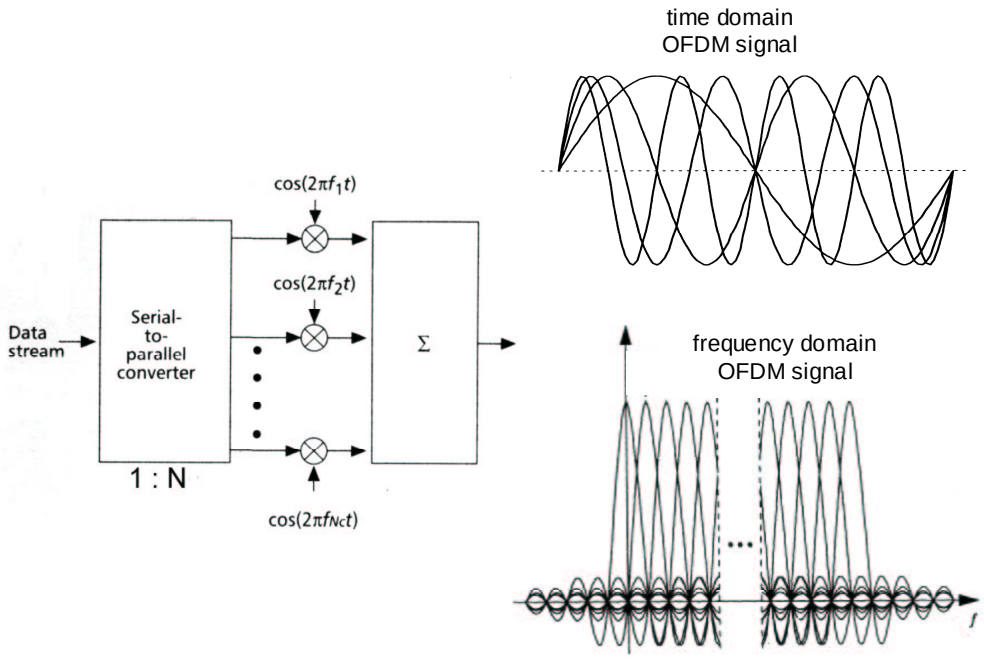

Figure 6.4: OFDM time and frequency domain waveforms.

$w(n-l) = \delta_{nl}$, the output of the FFT block at the receiver is given by:

$$\tilde{D}_m = \frac{1}{N}\sum_{k=0}^{N-1} r_k \mathrm{e}^{-j2\pi n\frac{k}{2N}} \tag{6.13}$$

$$\text{where} \quad r_k = \sum_{n=0}^{N-1} H_n D_n \mathrm{e}^{j2\pi\frac{nk}{2N}} + n(k) \tag{6.14}$$

Representing the frequency response of the time-invariant channel $h(t-\tau)$ at frequency n/T by H_n, and white noise samples as $N(n)$, the estimate of the data symbol is then:

$$\tilde{D}_m = \begin{cases} H_n D_n + N(n), & n = m \\ N(n), & n \neq m \end{cases} \tag{6.15}$$

A block diagram summarizing the OFDM modulation/demondulation scheme is presented in Figure 6.5. For further details about the analysis of OFDM in multipath Rayleigh fading channels, the reader is referred to [4].

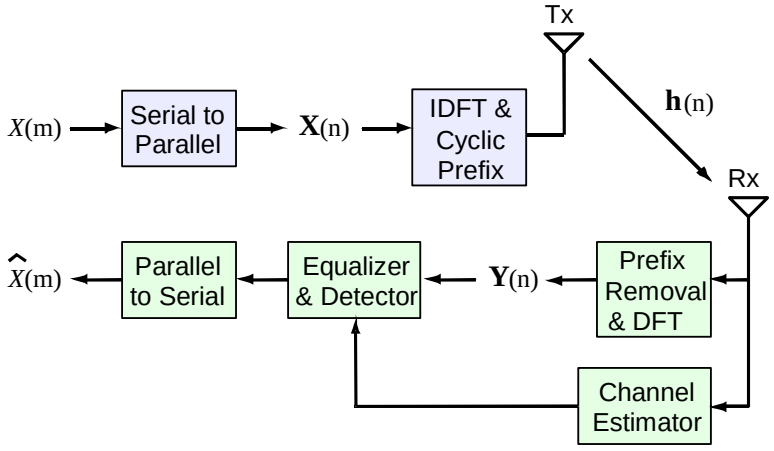

Figure 6.5: OFDM transmitter/receiver block diagram.

6.2.3 Low Density Parity Check (LDPC) Channel Codes

Channel coding, more generally also known as *error control coding*, is the technique of providing a structure for the signal that is transmitted over a channel (wired, wireless, or recording media) so that transmission errors are recognized (or corrected) at the receiver. This enables communication with a lower probability of error for a given channel signal to noise ratio than that possible without using channel coding. Or, equivalently, it allows communication at a lower signal to noise ratio for a given probability of error. There are many types of channel codes (block codes, convolutional codes, turbo etc.) with differing constraints and performance characteristics over a variety of channels. The reader is referred to standard texts [74] for a detailed survey.

Shannon proved that for any channel, there exist families of block codes that achieve arbitrarily small probability of error at any communication rate up to the capacity of the channel [81]. His proof was non-constructive, and there has been intense research activity in the last half century to determine the "best" codes that approach the limit (Figure 6.3). The best codes discovered thus far have

been the well known Turbo Codes [7], and the recently re-discovered low density parity check codes (LDPC) [56, 26].

For our proposed high data rate transceiver, we have determined LDPC as the channel code of choice because of two reasons. First, LDPC codes are regular block codes generated by sparse matrices, and thus have very desirable and relatively simple implementation structures. The decoding algorithms are particularly attractive and tractable. Second, the MIMO-OFDM configuration that forms the front end for this transceiver relies on the underlying LDPC channel code for signal separation, by means of a novel soft metric iterative calculation. This is discussed in detail in Section 6.4.2. In the next few paragraphs of this section, a brief background regarding the key features of LDPC codes are provided.

Definiton 6.1. A (n, k) *linear block code*, $\mathcal{C}$, with data word length k and codeword n is a k-dimensional subspace of the binary n-dimensional vector space $\mathbf{F}_2^n$.

There are 2^k datawords $\boldsymbol{u} = [u_0, u_1, \ldots, u_{k-1}]$ and 2^k corresponding codewords $\boldsymbol{c} = [c_0, c_1, \ldots, c_{n-1}]$ in the code $\mathcal{C}$. In matrix-vector notation:

$$\boldsymbol{c} = u_0\boldsymbol{g_0} + u_1\boldsymbol{g_1} + \cdots u_{k-1}\boldsymbol{g_{k\ 1}} \tag{6.16}$$

$$\boldsymbol{c} = \boldsymbol{u}\mathsf{G} \tag{6.17}$$

where

$$\mathsf{G} = \begin{bmatrix} \boldsymbol{g_0} \\ \boldsymbol{g_1} \\ \vdots \\ \boldsymbol{g_{k\ 1}} \end{bmatrix} \tag{6.18}$$

is the *generator matrix* incorporating the k linearly independent basis vectors of the code $\mathcal{C}$.

It is the codeword $\boldsymbol{c}$ instead of the data word $\boldsymbol{u}$ that is transmitted over the channel. The received signal, $\boldsymbol{r}$, at the receiver is then:

$$\boldsymbol{r} = \boldsymbol{c} + \boldsymbol{e} \tag{6.19}$$

which is the original codeword $\boldsymbol{c}$ corrupted by $\boldsymbol{e}$, the error induced by the channel. The function of the channel decoder is then to determine the original data word $\boldsymbol{u}$ from the received, corrupted codeword. The standard technique is to consider the nullspace $\mathcal{C}^{\perp}$ of the code $\mathcal{C}$ which is spanned by $(n-k)$ linearly independent vectors, $\boldsymbol{h}_0, \boldsymbol{h}_1, \ldots, \boldsymbol{h}_{n-k-1}$ and use the property that the codewords $\boldsymbol{c}$ are orthogonal to these vectors. Thus:

$$\begin{aligned} \boldsymbol{c}\,\boldsymbol{h}_i^T &= 0 \quad \forall i & (6.20)\\ \Longrightarrow \boldsymbol{c}\,\mathsf{H}^T &= \boldsymbol{0} & (6.21) \end{aligned}$$

where H is known as the *parity check matrix*. So, for the received signal $\boldsymbol{r}$:

$$\begin{aligned} \boldsymbol{s} &\triangleq \boldsymbol{y}\,\mathsf{H}^T = \boldsymbol{u}\,\mathsf{H}^T + \boldsymbol{e}\,\mathsf{H}^T & (6.22)\\ \boldsymbol{s} &= \boldsymbol{0} + \boldsymbol{e}\,\mathsf{H}^T & (6.23) \end{aligned}$$

where $\boldsymbol{s}$ is the *syndrome*. In syndrome decoding, a finite table of syndromes is constructed via Equation (6.22), according to the code structure. This is then used to guess the most likely error which has occurred, $\hat{\boldsymbol{e}}$, by means of one of the estimation techniques mentioned in Section 2.2.3 (e.g. maximum likelihood). Finally, the original transmitted codeword is recovered from this error estimate by:

$$\begin{aligned} \hat{\boldsymbol{c}} &= \boldsymbol{r} + \hat{\boldsymbol{e}} & (6.24)\\ &= \boldsymbol{c} + \boldsymbol{e} + \hat{\boldsymbol{e}} = \begin{cases} \boldsymbol{c} & \text{if no decoding error has occurred} \\ \neq \boldsymbol{c} & \text{in case of decoding error} \end{cases} & (6.25) \end{aligned}$$

which is successful in recovering the original codeword with a specified probability of error, depending on the noise statistics and code design (G).

Definiton 6.2. A *regular* (n, k) LDPC code is a linear block code who's parity check matrix, H contains exactly w_c 1's per column and $w_r = w_c \cdot (n/m)$ 1's per row, where $w_c \ll w_m$.

If the number of 1's per column or row are not constant, then the code is an *irregular* LDPC code. LDPC codes were invented by Gallager at MIT in the 1960's [26], and rediscovered by Makcay in the 1990's [56]. Since the generator matrix (and hence the parity check matrix) is sparse, the minimum code distance, d_{min}, which is the minimum number of columns of the parity check matrix that adds up to $\mathbf{0}$, is large. This is precisely what gives LDPC codes the ability to correct a large number of channel errors. Another interesting feature of LDPC codes shown by Gallager is that over an ensemble of regular LDPC codes, the distribution function for d_{min} for a randomly selected member of the ensemble resembles a step function, for d_{min}, greater than a small threshold, δn, where δ is a function of w_c, w_r. This implies that almost any randomly selected sparse generator matrix, G, for relatively large block lengths is automatically a "good" LDPC code. This remarkable fact makes the design of LDPC codes straightforward.

For the LDPC code for our transceiver design, we simply used the semi-random construction technique [56], whereby H is generated with weight-w_c columns, weight-w_r rows, and no two columns are allowed to have an overlap greater than one. In addition, short cycles are avoided and $\mathsf{H} = [\mathsf{H}_1 \vdots \mathsf{H}_2$ constrained so that H_2 is full rank. The algorithm is summarized as Algorithm 6.1.

Algorithm 6.1. Construction of LDPC Codes:

1. Chose parameters n, k, w_c, w_r and L_{min} = minimum cycle length. H will

then be an $m \times n$ matrix ($m = n - k$) with w_c 1's per column and w_r 1's per row.

2. Set column counter $i_c = 0$

3. Generate a weight-w_c column vector and place in column i_c of the H matrix that is being generated.

4. In the H matrix at this point, if the weight of each row is $\leq W_r$, if the overlap between any two columns is ≤ 1, and if all the cycle lengths are $\geq L_{min}$, then $i_c = i_c + 1$.

5. if $i_c = n$, STOP; else goto 3.

Because the algorithm is semi-random, the algorithm may not converge all the time. In such cases, it should be restarted with a new seed. A few extra steps may be necessary to ensure that the row weights are uniformly equal to w_r, but this is usually unnecessary since irregular LDPC codes have been shown to be better than regular LDPC codes. Once H is obtained, it is straightforward to obtain the generator matrix, G and perform the codeword encoding operation: $\boldsymbol{c} = \boldsymbol{u}\,\mathsf{G}$.

The decoding algorithms for LDPC codes are elegant iterative techniques, originally proposed by Gallager [26], and are effectively optimal. It iteratively computes distributions of variables in graph-based models and come under different names, depending upon the context:

- sum-product algorithm (all graphical models)
- min-sum algorithm (approximation to the sum-product algorithm)
- forward-backward algorithm, BCJR algorithm, APP or MAP algorithm (trellis-based graphical models)

- belief-propagation algorithm, message-passing algorithm (machine learning, AI, Bayesian networks.

The "sum-product" or "message-passing" algorithms are the most commonly applied names for the algorithm applied to the decoding of LDPC codes. A detailed discussion of the decoding algorithms is not provided further in this thesis, since it is amply illustrated in the published literature. Rather, a brief intuitive outline is included, which serves to highlight the connection between the belief-passing, graph-based algorithm and the Likelihood Opinion Pool data fusion technique, discussed in Section 2.2.4. It is seen that, after all, LDPC decoding is a form of multi-sensor data fusion, and that the concepts outlined in Chapter 2, dealing with likelihood function representations of data, are directly applicable.

It was Tanner [90] who originally considered LDPC codes and showed that they may be represented effectively by a bipartite graph [100] (also known as a *Tanner* graph), where *check* and *bit* nodes represent the structure of the parity check matrix. A check node j in the row of H is connected to the bit node i in the column of H, whenever element h_{ji} in H is a '1'. Graphically, this is shown in Figure 6.2.3

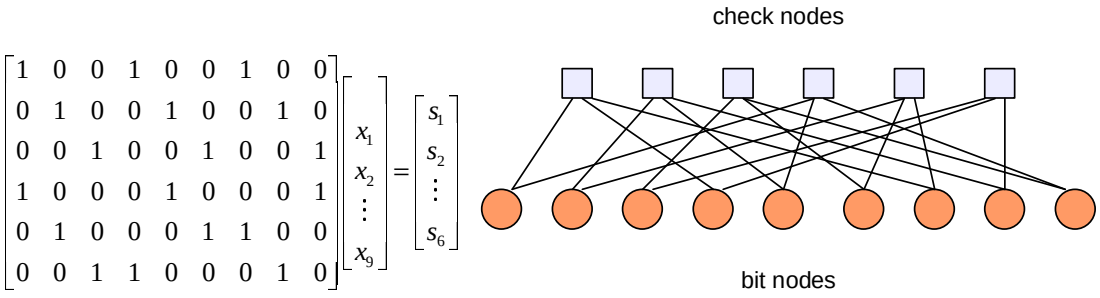

Figure 6.6: Tanner graph representation of the parity check matrix of an LDPC code

During decoding, the received codeword is matched to the bit nodes, and the structure of the parity check matrix then causes the check nodes to check the parity of the resulting node connections from the check to the bit nodes. If the parity of the connections at each check not does not match, it implies that the received codeword is in error. In this case, the decoding algorithm then uses bipartite graph and attempts to determine which of the bit nodes contain the error. The check nodes provides "guesses" regarding which nodes it suspects are in error (due to parity check failures) to the bit nodes. This is a form of extrinsic information, and in the soft-decoding version of the algorithm, involves likelihood ratios (similar to the observations of multiple sensors in a distributed sensor network, Section 2.2.4), as in Equation (6.26).

$$\text{Log-Likelihood Ratio (LRR)} = \log\left\{\frac{\Pr(c_i = 0 \mid \boldsymbol{r}, \boldsymbol{s_i})}{\Pr(c_i = 1 \mid \boldsymbol{r}, \boldsymbol{s_i})}\right\} \tag{6.26}$$

where, $\boldsymbol{r}$ is the received codeword, $\boldsymbol{s_i}$ is the event that the bits in the codeword $\boldsymbol{c}$ satisfy the w_c parity check equations involving c_i. In the second half of the iteration, the aggregated likelihood metrics and messages from the bit nodes are then passed back to the check nodes. Decoding is stopped after a maximum number of iterations is reached or before that, provided $\hat{\boldsymbol{c}}\,\mathsf{H}^T = \mathbf{0}$, else the codeword error in unrecoverable. The algorithm is summarized as Algorithm

Algorithm 6.2. General LDPC Decoding Algorithm:

1. Initialize bit/check nodes.
2. Pass messages from bit nodes to check nodes for parity check.
3. Pass messages (parity check likelihood ratios) from check nodes to bit nodes.
4. Find $\hat{\boldsymbol{x}}$ and $\hat{\boldsymbol{c}}$ from the probabilistic information residing at the nodes $\{x_i\}$, attempting to satisfy $\hat{\boldsymbol{c}}\,\mathsf{H}^T = \mathbf{0}$ within a maximum number of iterations, then goto 2; else STOP - declare error, goto 2.

The algorithm is thus a form of the belief-propagation system [67], which is also a characteristic in trellis-based searches. It is shown in graphical form in Figure 6.7.

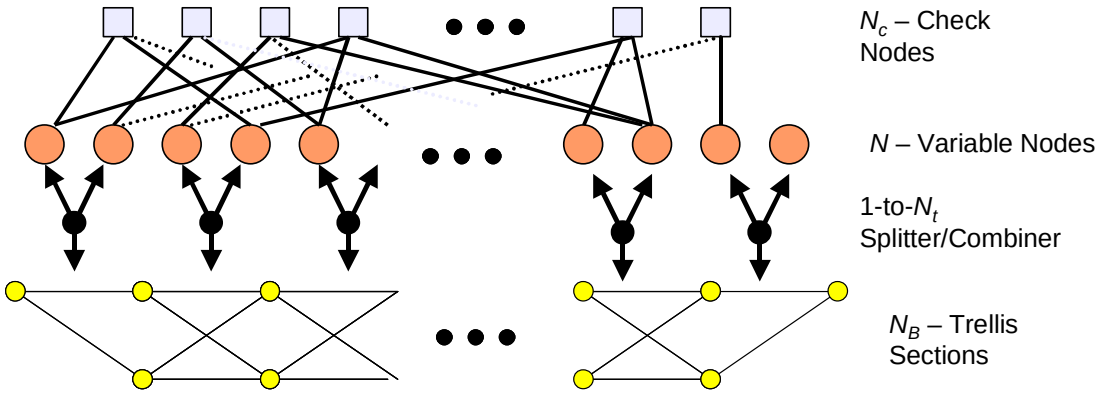

Figure 6.7: Likelihood ratio decoding of LDPC codes.

6.3 MIMO-OFDM Channel Estimation

In general, a *communication channel* can refer to any medium that is in between the data source and sink. Therefore, it can be the vacuum of space (space communications), water (for underwater communications), air (most of the types of terrestrial wireless communications) solids (wireline communications over cables, etc.), or a combination. Among them, the air channel, or more commonly referred to as *wireless* radio frequency channel (RF), is the most varied, since it encompasses a variety of physical environments: open spaces, dense urban areas, foliage cover, etc. Each of these physical environments influences the propagation of the radio signal from the transmitter to the receiver operating in this region. Unfortunately, the RF characteristics in each of these environments is unique and frequency dependent, so a single mathematical characterization of the channel is not universally valid.

The dense urban area, specifically, with multiple physical structures, RF reflecting/absorbing areas, and strong sources of interference, is a particularly harsh RF environment. Also referred to as the mobile communication channel, this channel is characterized as being the most difficult environment to conduct wireless communications over. Because of multiple signal reflections (and random adding/subtracting of the resulting signal paths), this channel is prone to multipath fading, which takes the form of nonlinear channel responses in the frequency spectrum.

To successfully design radio devices for the mobile RF channel (for detection, demodulation, decoding), it is necessary to be able to estimate the effects of the channel in the time/frequency domain. The estimation is, by definition, only an approximation since the channel state is infinite dimensional. The traditional approach for single-input single-output systems (1 transmit and 1 receive antenna) has been to be *model* the effects of the channel by means of a large finite-state machine (FSM). The engineering version of this model is in the form of a Finite Impulse Response (FIR) filter together with a AWGN source. The filter's impulse response is then tuned to approximate the impulse response of the channel under consideration, which corresponds to the noise contaminated multipath behavior of the RF channel.

The magnitude, phase and time of arrival of the multipath components, in particular, are the critical factors that are of importance to the radio designer. The FIR structure can model these as a fractionally sampled filter approximating the discrete-time sampled version of the infinite-dimensional mobile channel. The problem of estimating the channel parameters (*channel estimation*) then simply reduces to estimating the time-varying FIR filter coefficients, and the delay spacing of the taps. Thus, for a SISO system, the end result is a tapped delay

line channel model where the filter coefficients for a $(2n+1)$ tap filter can be represented by means of a vector $\boldsymbol{h}$, as shown in Equation (6.27).

$$\boldsymbol{h} = \left(h_{-n}, h_{-(n-1)}, \ldots, h_{-1}, h_0, h_1, h_2, \ldots, h_n\right) \tag{6.27}$$

Channel estimation then involves estimating (and time-updating) the coefficients h_0, h_1, etc. Various methods have been devised for this endeavor [36]. The basic technique has been to employ channel sounding pulses, or known *training sequences* that are transmitted over the channel and deconvolved at the receiver to estimate the filter taps at specific instants of time. Various adaptive schemes are then possible to approximately track the channel effects and periodically update the filter coefficients as required.

For a MIMO channel, however, the problem is more complex. RF transmissions from *each* of the antenna elements of the transmitter are distorted by the channel, and a mixed, noise corrupted RF signal is observed by *each* of the receive antennas (Figure 6.8). If m transmit and n receive antennas are used, then each of the mn transmit-receive antenna pairs is effectively a SISO channel by itself, and thus has to be modeled by a sperate FIR filter. The coefficient vector $\boldsymbol{h}_{ij}$ would then represent the channel between transmit antenna i and receive antenna j. The filter coefficients of the various transmit-receive channel permutations can be conveniently organized by means of the *channel matrix*, H, which captures the complete effect of the MIMO channel, Equation (6.28).

$$\text{Channel Matrix} = \mathsf{H} = \begin{pmatrix} \boldsymbol{h_{11}} & \boldsymbol{h_{12}} & \cdots & \boldsymbol{h_{1n}} \\ \boldsymbol{h_{21}} & \boldsymbol{h_{22}} & \cdots & \boldsymbol{h_{2n}} \\ \vdots & \vdots & \ddots & \vdots \\ \boldsymbol{h_{m1}} & \boldsymbol{h_{m2}} & \cdots & \boldsymbol{h_{mn}} \end{pmatrix} \tag{6.28}$$

The problem in this instance to estimate the vectors $\boldsymbol{h}_{ij}$. Normal SISO training sequences are not sufficient, since the transmissions from each antenna ele-

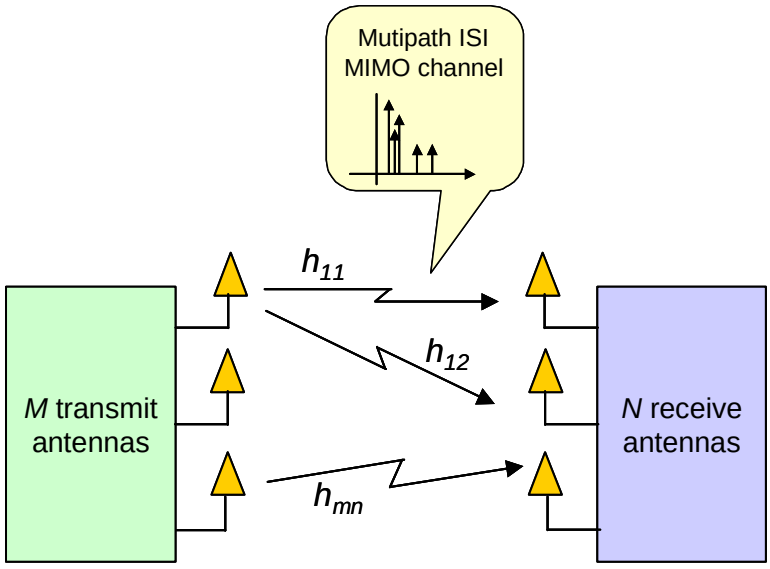

Figure 6.8: Multi-input, multi-output radio frequency channel.

ment interferes with the transmissions from the other antenna elements of the device. Instead, training sequences should be designed to decouple the inter-antenna interference for channel estimation. Several such schemes have been proposed of late [53], where the focus has been to design mutually *orthogonal* training sequences. In this thesis, one such simplified scheme is presented for the case of LDPC coded MIMO-OFDM channels.

6.3.1 Simplified Channel Estimation for LDPC coded MIMO-OFDM Channels: Code Design

Without loss of generality, we consider the mathematical development for a 2×1 case only for convenience, since 2×2 and higher versions are straightforward extensions of the same concept. For modeling individual transmit-receive antenna pairs, a three-tap FIR filter ($N_c = 3$) is assumed. The extension to higher filter lengths is also straightforward. The block diagram of the channel model is

shown in Figure 6.9. We now derive some necessary constraints for the training sequences that can be used to estimate the channel parameters of this simplified MIMO channel. At time instant k, if the transmitted signals from antennas #1

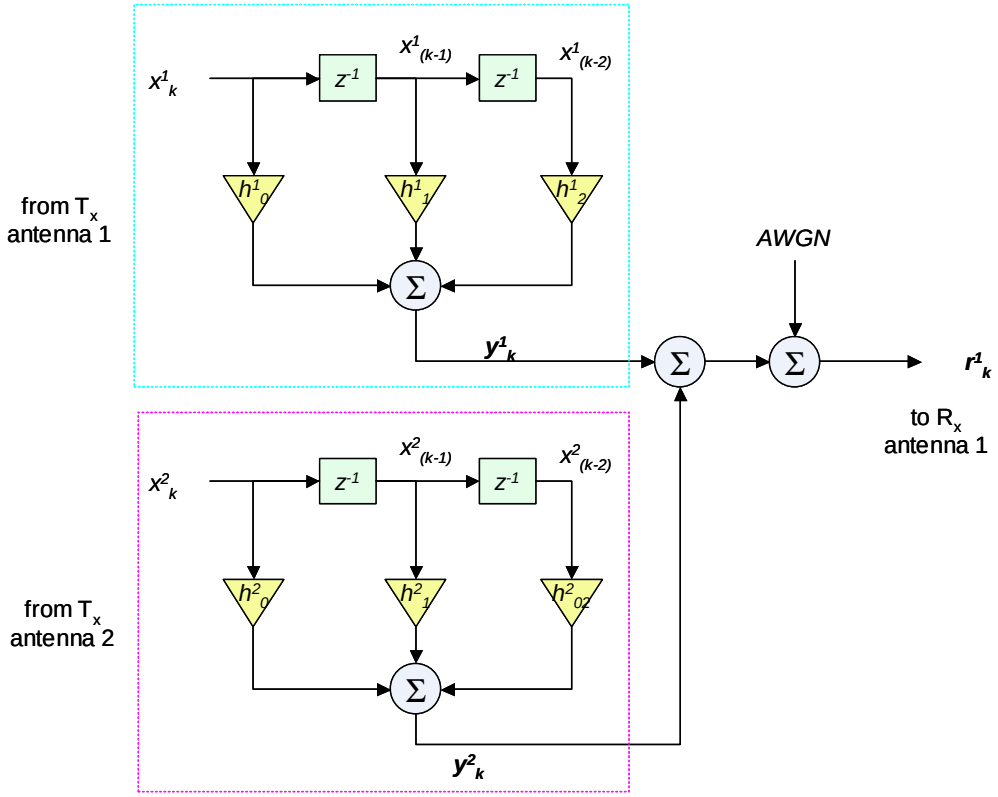

Figure 6.9: Three-tap FIR filter model for a 2×1 MIMO channel.

and #2 are x_k^1 and x_k^2 respectively, then the MIMO channel represented by the FIR filters performs a weighted summation of the signals x_k^i, x_{k-1}^i, and x_{k-2}^i for each of the transmitted streams to produce the signals y_k^1, y_k^2 which are then further mixed with additive white Gaussian noise (AWGN). The received signal at receive antenna #1 is therefore:

$$\begin{aligned} r_k &= y_k^1 + y_k^2 + n_k \\ \text{where} \qquad y_k^j &= \sum_{i=0}^{2} h_i^j x_{k-1}^j + n_k \end{aligned} \tag{6.29}$$

For the purpose of estimating the channel filter coefficients with training symbols, we now consider collecting y_0^j, $y_1^j, \ldots, y_{N-1}^j$ for N time samples. Then, the time series vector of the signals distorted by the channel coefficients can be written as :

$$\begin{pmatrix} y_0^j \\ y_1^j \\ \vdots \\ y_{N-1}^j \end{pmatrix} = \underbrace{\begin{pmatrix} x_0^j & x_{-1}^j & x_{-2}^j \\ x_1^j & x_0^j & x_{-1}^j \\ x_2^j & x_1^j & x_0^j \\ \vdots & \vdots & \vdots \end{pmatrix}}_{\triangleq \mathsf{X}^j \text{ : a Toeplitz matrix}} \begin{pmatrix} h_0^j \\ h_1^j \\ h_2^j \end{pmatrix} \tag{6.30}$$

Constraint 1. The time-series transformation matrix, X^j, describing the training sequence (or matrix) input-output relationship of the channel must be a Toeplitz matrix as shown in Equation (6.30).

The matrix-vector relationships for antenna #1 and antenna #2 can then we written as:

$$\boldsymbol{Y}^1 = \mathsf{X}^1 \boldsymbol{h}^1 \tag{6.31}$$

$$\boldsymbol{Y}^2 = \mathsf{X}^2 \boldsymbol{h}^2 \tag{6.32}$$

where $\boldsymbol{X}^j$, as defined in Equation (6.30), is the matrix of the training symbols transmitted from antenna j, with dimension $(N \times 3)$, and $\boldsymbol{h}^j$ is the corresponding unknown channel coefficients for that transmit to receive channel (dimension 3×1). Therefore, Equation (6.29) can be re-written as:

$$\boldsymbol{r} = \mathsf{X}^1 \boldsymbol{h}^1 + \mathsf{X}^2 \boldsymbol{h}^2 + \boldsymbol{n}. \tag{6.33}$$

Constraint 2. Ideally, the training sequences should have the following correla-

tion property:

$$(\mathsf{X}^1)^T(\mathsf{X}^1) = \begin{pmatrix} N\sigma^2 & 00 & \\ 0 & N\sigma^2 & 0 \\ 0 & 0 & N\sigma^2 \end{pmatrix} \quad \text{where} \quad \sigma^2 = E\{x^2\} \tag{6.34}$$

$$(\mathsf{X}^i)^T(\mathsf{X}^j) = \begin{pmatrix} 0 & 0 & 0 \\ 0 & 0 & 0 \\ 0 & 0 & 0 \end{pmatrix} \qquad i \neq j \tag{6.35}$$

The interesting thing to note here is that if the x's are randomly chosen for the matrix X, then the requirement of Equations (6.34) and (6.35) can be easily satisfied. For example, consider a binary random variable, $x \in \{-1, 1\}$ drawn with equally likely probability. Then for the sample sequence $\boldsymbol{x}^i = (x_1^i, x_2^i, \ldots, x_N^i)$ where the elements are drawn iid, we have:

$$\begin{aligned} \boldsymbol{x}^i \cdot \boldsymbol{x}^j &= \sum_{p=1}^{N} x_p^i x_p^j \\ &= \begin{cases} N, & \text{if} \quad i = j \\ 0, & \text{if} \quad i \neq j \end{cases} \end{aligned} \tag{6.36}$$

Thus random selection of the training sequence elements will satisfy constraint 2. But the selections must also satisfy constraint 1, the Toeplitz condition. Therefore we propose the following sequence:

$$\boldsymbol{x} = \underbrace{(x_1, x_2, \ldots, x_{N+2N_c-1})}_{\text{elements draw iid, taking -1 or +1 with equally likely probability}} \tag{6.37}$$

Then, consecutive N symbols are chosen for the columns of the X matrices. For the simplified case under consideration, we thus need training matrices X^1 and X^2 that are Toeplitz and whose columns are orthogonal to each other. An example is considered next.

Given the iid sequence in Figure 6.10, we can select the columns of X^1 and X^2 as shown in Figure 6.10 and Equation (6.38),

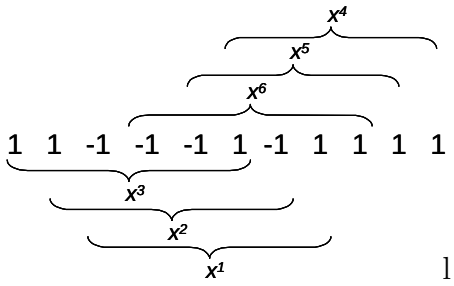

Figure 6.10: Orthogonal training sequence elements

$$\mathsf{X}^1 = \left[x^1 x^2 x^3\right] = \begin{pmatrix} -1 & 1 & 1 \\ -1 & -1 & 1 \\ -1 & -1 & -1 \\ 1 & -1 & -1 \\ -1 & 1 & -1 \\ 1 & -1 & 1 \end{pmatrix} \tag{6.38}$$

$$\mathsf{X}^2 = \left[x^4 x^5 x^6\right] = \begin{pmatrix} 1 & -1 & -1 \\ -1 & 1 & -1 \\ 1 & -1 & 1 \\ 1 & 1 & -1 \\ 1 & 1 & 1 \\ 1 & 1 & 1 \end{pmatrix} \tag{6.39}$$

where both X^1 and X^2 are now Toeplitz matrices, as required by constraint 1. We also note that:

$$(\mathsf{X}^i)^T(\mathsf{X}^i) = \begin{pmatrix} \sigma \cdot 1 & 0 & 0 \\ 0 & \sigma \cdot 1 & 0 \\ 0 & 0 & \sigma \cdot 1 \end{pmatrix}, \quad \text{and} \tag{6.40}$$

$$(\mathsf{X}^1)^T(\mathsf{X}^2) = \begin{pmatrix} 0 & 0 & 0 \\ 0 & 0 & 0 \\ 0 & 0 & 0 \end{pmatrix}, \tag{6.41}$$

as required by constraint 2. These are the desired matrix of training sequences that can be transmitted from the respective antennas. The computed autocorrelation for a length-6 sequence is shown in Figure 6.11, where the orthogonal and hence impulse autocorrelation nature of the sequence are observed, as required by Equations (6.40) and (6.41).

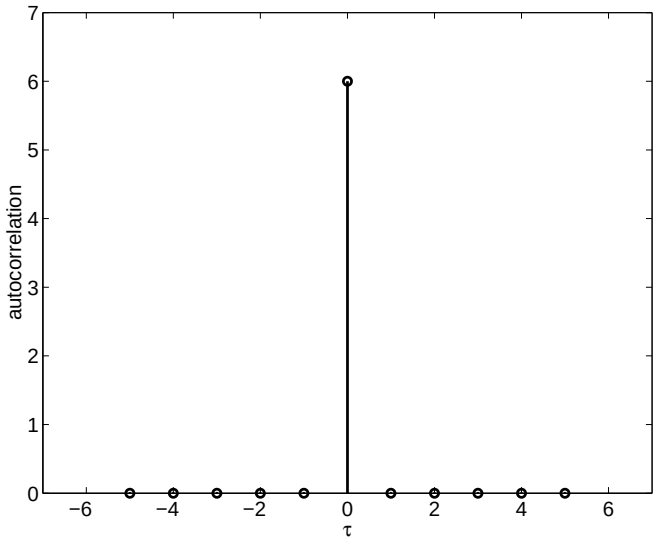

Figure 6.11: Autocorrelation of length-6 perfect sequence training symbols.

6.3.2 Channel Matrix Estimation

In the previous section, the matrices of the training sequences for each antenna were designed. They were seen to be have desirable correlation properties and a Toeplitz structure. At the receiving antenna(s), the individual channel parameters can then be estimated by the received versions of these training pulses, as per the following theorem.

Theorem 6.1. The optimum estimate of the channel matrix $\boldsymbol{h}^j$ for a MIMO channel, given the training sequence matrices as in Equations (6.40) and (6.41), is given by:

$$\hat{\boldsymbol{h}^j} = \left[(\mathsf{X}^j)^T(\mathsf{X}^j)\right]^{-1}(\mathsf{X}^j)^T\boldsymbol{r} \tag{6.42}$$

Proof. The total received signal for the 2×1 system is given by:

$$\boldsymbol{r} = \mathsf{X}^1\boldsymbol{h}^1 + \mathsf{X}^2\boldsymbol{h}^2 + \boldsymbol{n} \tag{6.43}$$

Therefore:

$$(\mathsf{X}^1)^T\boldsymbol{r} = \left[(\mathsf{X}^1)^T(\mathsf{X}^1)\right]\boldsymbol{h}^1 + \overbrace{\left[(\mathsf{X}^1)^T(\mathsf{X}^2)\right]\boldsymbol{h}^2}^{=0} + (\mathsf{X}^1)^T\boldsymbol{n} \tag{6.44}$$

Thus, the Least Squares Estimate of $\boldsymbol{h}^1$, by the Gauss-Markov theorem [88], can be shown to be:

$$\hat{\boldsymbol{h}^1} = \left[(\mathsf{X}^1)^T(\mathsf{X}^1)\right]^{-1}(\mathsf{X}^1)^T\boldsymbol{r} \tag{6.45}$$

A similar symmetric relationship holds for $\boldsymbol{h}^2$. □

We note that the matrix $\left[(\mathsf{X}^1)^T(\mathsf{X}^1)\right]$ is diagonal, so the estimation algorithm can be efficiently implemented. The sequence in Figure 6.10 that we considered for constructing the training matrix is a *perfect* sequence, and as we have seen, it can be obtained by randomly selecting samples from binary random variables $\in \{+1, -1\}$.

6.4 Combining LDPC and OFDM with MIMO

Given the structure of the LDPC coding and decoding algorithm (Section 6.2.3), and the OFDM particulars (Section 6.2.2), we now describe the manner in which these technologies can be merged to yield an LDPC coded multicarrier system. We initially consider the case of a 1×1 system (single transmitter and receiver) for outlining the procedure, assuming BPSK as the front-end modulation for the OFDM multicarriers. Then, in Section 6.4.2, we describe how this technique can be extended to the 2×1 MIMO case involving 2 transmit and 1 receive antennas. We see that it results in a simplified signal separation scheme that itself can be further extended to higher order MIMO systems. In Section 6.4.3, we also derive the soft metric calculations for higher order modulation support for the OFDM multicarriers: in particular for 4-PSK modulation.

6.4.1 Likelihood Metrics for OFDM modulated LDPC

We consider the OFDM modulated LDPC system as shown in Figure 6.12 For

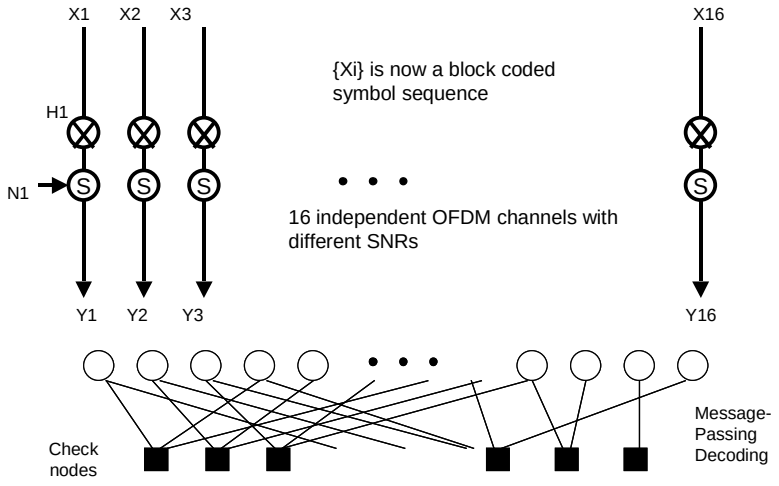

Figure 6.12: LDPC coded OFDM for a 1×1 system with 16 OFDM carriers.

this 1×1 system, in order to decode the LDPC channel code, we require a suitable likelihood probability that can be used for the bit nodes of the LDPC decoder. The result is stated as Theorem and derived below. We see that the result turns out to be a relatively simple Gaussian dominated soft metric, if we use the orthogonal bipolar symbolling scheme similar to the channel estimation techniques discussed in Section 6.3. We assume that we have used the channel estimation techniques described in that Section and thus have an estimate of the channel matrix (in this case a vector of the channel coefficients, $\boldsymbol{H}$).

Theorem 6.2. The optimum log-likelihood ratio soft metric for LDPC decoding of an 1×1 OFDM multicarrier system is:

$$\frac{1}{\sigma^2}\left(h_i + h_i^*\right) \tag{6.46}$$

where h_i is the channel coefficient for the i^{th} individual OFDM carrier bin.

Proof. At the receiver of the system, we observe:

$$\underbrace{Y_i}_{\text{observation}} = \underbrace{h_i}_{\text{known}} \cdot \underbrace{X_i}_{\text{unknown}} + \underbrace{N_i}_{\sim \mathcal{N}(0,\sigma^2)} \tag{6.47}$$

Thus, the apriori probability yields:

$$\begin{aligned} \Pr\left[Y_i = y \mid X_i = x \in \{-1, +1\}\right] &= \Pr\left\{N_i = y - x\right\} && (6.48) \\ &= \frac{1}{\sqrt{2\pi}\sigma} \cdot \mathrm{e}^{-\frac{|y - h_i x|^2}{2\sigma^2}} && (6.49) \end{aligned}$$

from where the likelihood ratio can be computed to be:

$$\begin{aligned} \frac{\Pr\left[Y_i = y \mid X_i = 1\right]}{\Pr\left[Y_i = y \mid X_i = -1\right]} &= \frac{\exp\left\{-\frac{y^2 - (h_i + h_i^*) + \|h_i\|^2}{2\sigma^2}\right\}}{\exp\left\{-\frac{y^2 + (h_i + h_i^*) + \|h_i\|^2}{2\sigma^2}\right\}} && (6.50) \\ &= \exp\left\{\frac{1}{2\sigma^2} \cdot 2(h_i + h_i^*)\right\} && (6.51) \end{aligned}$$

Taking logarithms:

$$\log\left\{\frac{\Pr\left[Y_i = y \mid X_i = 1\right]}{\Pr\left[Y_i = y \mid X_i = -1\right]}\right\} = \frac{1}{\sigma^2}\left(h_i + h_i^*\right) \tag{6.52}$$

□

6.4.2 Signal Separation of LDPC coded OFDM Signals in MIMO Systems

The technique for combining the advantages of LDPC channel codes with the ISI-combating OFDM multicarrier technology has been described for a 1×1 in the previous subsection. However, a major problem occurs when trying to incorporate the diversity benefits of a MIMO system on top of these techniques. This is the issue of signal separation. For example, for a 2 MIMO system, both the transmit antennas will be transmitting signals simultaneously (in the same time-frequency dimension), so the receiver will observe a mixture of the two transmitted signals. This is the same problem that we encountered in Section 6.3.1 when we were attempting to estimate the MIMO channel parameters. However, in that case, we designed *known*, special orthogonal training codes that could be easily separated. For regular data transmissions, the symbols are unknown. What is required is a technique that will enable the simultaneous transmission of independent data streams on each of the MIMO antennas, and yet enable separation of the received, mixed signals at the receiver.

There has been significant research activity in this area since the seminal paper by Siavash Alamouti [1]. In the Alamouti scheme, a repetition coding approach is taken for MIMO signal separation (for a $2 \times n$ system). In summary, a symbol is first transmitted from one antenna, and the conjugate of the same symbol is then re-transmitted from the second antenna, but with a unit delay. The conjugate structure allows some selective processing at the receive antennas

to eventually separate the transmitted streams. The Alamouti scheme is mainly used to achieve transmit diversity of order 2, and thus reduces the probability of error for the overall system. However, since the constraint length of the code is only 2, it does not make a good code. Further work has extended the Alamouti scheme to more than 2 transmit antennas. Our contention is that almost all of the diversity benefits, together with added coding benefits can be achieved with simple LDPC+OFDM embedded onto MIMO, since the LDPC code is a much better code than a simple repetition code. In addition, the interference problem can be solved with Turbo code-like iterations.

The proposed scheme is as follows (considered for a 2×1 system). There are two possible methods.

Method 1: Two streams of LDPC coded symbols are transmitted over the two transmit antennas. For example, the odd indexed symbols can be transmitted via the first antenna and the even indexed symbols can be transmitted over the second antenna. The scheme can be extended in an analogous manner for more than 2 antenna systems. Considering the BPSK example discussed in previous sections, if $x_1, x_2, \ldots, x_N \in \{+1, -1\}$, where N is the number of OFDM bins (e.g. 1024), then $x_1, x_3, x_5, \ldots$ would be transmitted from antenna 1, and $x_2, x_4, x_6, \ldots$ from antenna 2. At the receiver, the signals would mutually interfere, but the LDPC code structure that the signals have been embedded with can now be exploited. In principle, Baye's rule can be applied to extract likelihood probabilities for the odd and even sequences from the received signal $r_k = x_k^1 + x_k^2 + n$. The question is in how to generate the likelihood probabilities for the even and odd sequences in practice.

We propose that a simple scheme is to use the posterior probabilities from

the odd sequence as the prior probability estimates for generating the likelihood probability for the even sequence, and vice versa. In this manner, the LDPC decoding algorithm would thus be able to compute the forward-backward iterations and separate the signal mixture into the even and odd streams.

Mathematically, we consider the first frequency bin (e.g. out of a total of 1024) for the LDPC coded OFDM system. Then, rewriting Equation (6.33), we have:

$$\boldsymbol{r_1} = \mathsf{H}_1^1 \boldsymbol{x}_1^1 + \mathsf{H}_1^2 \boldsymbol{x}_1^2 + \boldsymbol{n}_1. \tag{6.53}$$

We drop the subscripts indices for brevity. Then the likelihood metric for the first antenna is given by:

$$\begin{aligned} \Pr\{\boldsymbol{r} \mid \boldsymbol{x}^1\} &= \sum_{\{\boldsymbol{x}^2\}} \Pr\{\boldsymbol{r}, \boldsymbol{x}^2 \mid \boldsymbol{x}^1\} && (6.54) \\ &= \sum_{\{\boldsymbol{x}^2\}} \Pr\{\boldsymbol{r} \mid \boldsymbol{x}^2, \boldsymbol{x}^1\} \Pr\{\boldsymbol{x}^2 \mid \boldsymbol{x}^1\} && (6.55) \\ &= \sum_{\{\boldsymbol{x}^2\}} \Pr\{\boldsymbol{r} \mid \boldsymbol{x}^2, \boldsymbol{x}^1\} \Pr\{\boldsymbol{x}^2\} && (6.56) \end{aligned}$$

since the odd and even streams $\{\boldsymbol{x}^1, \boldsymbol{x}^2\}$ are in general independent. The interesting thing to note here is that Equation (6.56) is computable. The posterior probability $\Pr\{\boldsymbol{x}^2 \mid \boldsymbol{r}\}$ over the whole sequence is used in place of $\Pr\{\boldsymbol{x}^2\}$, which is obtained from a similar operation on the second antenna.

Finally, with the likelihood probability $\Pr\{\boldsymbol{r} \mid \boldsymbol{x}^1\}$, the posterior probability $\Pr\{\boldsymbol{x}^1 \mid \boldsymbol{r}\}$ itself can be obtained. This routine is then repeated at every iteration for convergence to the correct estimate of the transmitted symbol.

Method 2: A variation on Method 1 can be achieved by sending two independent blocks of LDPC codes over the two antennas. For example, if half of the bits from an LDPC code word is transmitted on the first antenna then the other

half can be sent on the second antenna, for an overall bit rate of 1 bps. The advantage in this case is that the independence assumption of Equation (6.56) is then perfectly valid. The same decoding scheme as discussed for Method 1 can also be applied here, with an expectation of better convergence results.

We note that both these schemes are scalable to many transmit antennas. A simulation study and results are presented in later sections of this chapter.

6.4.3 LDPC Soft Bit-Metrics for Decoding M-ary Symbols in OFDM

As mentioned earlier, the decoding development that has been considered in the previous sections has been for the case of BPSK modulated OFDM carriers with LDPC channel codes. In this scheme, the soft values of the transmitted bits (not symbols), together with the likelihood values, are what are required to perform the forward-backward iterations for LDPC decoding (Section 6.2.3). In this subsection, we now consider the case of LDPC decoding for OFDM with higher constellation modulations, e.g 4-QAM or 16-QAM. Ostensibly, for 4-QAM, the problem can be treated as two independent BPSK's, but for the other constellations, either a hard decision decoding can be performed (mapping the received symbols to their guessed symbols and then LDPC decoding), or a soft decision metric has to be derived. Hard decision decoding in this case is inherently a poor choice since it can lead to instability in the forward-backward algorithm and higher decoding errors. Thus, a soft decision metric is derived below.

We consider the case of the symbols being drawn from a 16-QAM signal constellation. That is, $X_k \in$ 16-QAM. Then there are four bits representing each symbol in the constellation. For the symbol sequence $X_1, X_2, \ldots, X_N$, we need to the extract the binary soft-decision metric for the bit sequence of length $4N$. For OFDM which produces no ISI, it is sufficient to show the procedure for a

particular $k \in \{1, 2, \ldots, N\}$, since the operations are the same the for all k.

We have four bits per symbol, which are denoted by:

$$b_k^1, b_k^2, b_k^3 \text{ and } b_k^4 \text{ for each symbol } Y_k \tag{6.57}$$

The received signal is $Y_k = X_k + N_k$ where N_k is AWGN and $X_k \in$ 16-QAM. On each Y_k, we can generate 16 likelihood functions, which are basically 16 Gaussian pdfs, each having a mean value of the 16 QAM symbols and a variance of σ^2:

$$f_k(i) = \Pr\{Y_k \mid X_k(i) = x(i)\} \tag{6.58}$$

Alternatively, we can take the 15 ratios: $f1/f1, f2/f1, \ldots$ etc., all with respect to f_1 if we wish to avoid performing an exact calculation of the pdfs.

Next, the 16 QAM symbols in the constellation have to be mapped with a specific rule to their 4 bit representations. It is an open question at this point as to the optimum mapping rule (Gray code, anti-Gray code etc.) with respect to LDPC decoding, but we have assumed the natural mapping rule assigning symbol-indices to bits:

$$\begin{aligned} X_k(0) &= (0000) \\ X_k(1) &= (0001) \\ X_k(2) &= (0010), \quad \text{etc.} \end{aligned} \tag{6.59}$$

Denoting the posterior probability of the first bit being a "1" as π_k^1, we have:

$$\pi_k^1 = \Pr\left\{b_k^1 = 1 \mid Y_k\right\} \tag{6.60}$$

There are 8 possibilities that the first bit is zero, i.e.$(1 * * *)$, $2^3 = 8$, so adding all the 8 likelihoods and dividing out the sum of all the 16 likelihoods, we have:

$$\pi_k^1 = \frac{\text{sum of the 8 likelihoods}}{\text{sum of all the 16 likelihoods}} \tag{6.61}$$

The exact same routine holds for the other three positions of the "1", such as (* 1 * *), (* * 1 *), and (* * * 1), and symmetric expressions can be derived for the other π_k^i's. These binary π's are the required soft metrics that are then fed to the LDPC decoding subroutine.

6.5 Adaptivity for LDPC coded MIMO-OFDM and System Design

One of the aims of this effort has been to enable the proposed transceiver system to be adaptive to the channel and network conditions. If additional bandwidth becomes available, or SNR improves, etc., the radio units in communication should be able to dynamically change their operating parameters to exploit the resource that has become available. This is the basic concept behind *software defined radios* [94].

Specifically, one parameter of frequent interest is frequency domain (spectrum) usage. For the proposed transceiver, this can be exploited by using an adaptive form of OFDM, whereby the number of carriers are dynamically altered in response to the available bandwidth. This is in response to studies that have shown that a large portion of the bandwidth allocation for most types of networks wireless services (with the possible exception of commercial cellular service) often remain idle. Examples include wireless LANs such as IEEE802.11a, military radio systems, etc. This technology can attempt to fill those 'frequency holes' by using multi-carrier modulation with a variable number of carriers. An example for the IEEE802.11a standard is shown in Figure 6.13.

However, the adaptivity operation requires a feedback mechanism from the receiver to transmitter over a common control channel that relays updated channel

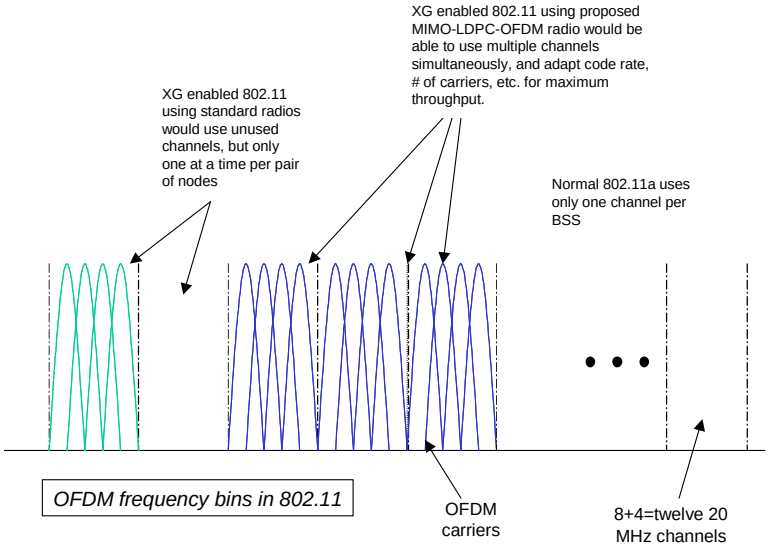

Figure 6.13: Using adaptive OFDM to increase spectral efficiency in IEEE802.11a.

and system operating parameters. In addition , an optimization/decision block is also required that would pool similar operating point data from the other users in the network and then perform an optimization to determine the best use of the spectrum and radio resources. Referring back to Figure 6.1 on page 155, these blocks and data pathways are shown as high level objects implementing the feedback mechanism for the transceiver. Unfortunately, a detailed study of these distributed optimization and negotiation protocols are beyond the scope of this thesis. Research in these areas are currently being spearheaded by DARPA as part of the multi-year Next Generation Communications (XG) project [18]. However, we now briefly discuss the manner in which these spectrum optimization protocols would fit with the proposed transceiver and thus enable rate adaptivity.

Referring to Figure 6.1, the transceiver has a space-time-frequency coding and modulation block at the transmitter that is fed with a set of tunable transceiver parameter information from a spectrum monitoring node or feedback channel.

The exact nature and availability of this channel would be application dependent: e.g. it could be one of the 20 common channels for the IEEE 802.11a standard that is pre-negotiated. In this case, the modified 802.11a standard would enable adaptive tuning of standard OFDM parameters. These parameters can include the transmission bandwidth W, the carrier frequency fc, the number of frequency bins Q, and the set of transmission rates $\{R_i\}_{i=1,...,Q}$ in the individual frequency bins $\{f_i\}_{i=1,...,Q}$, which the spectrum monitoring/optimizing algorithm has determined as being supportable for a given scenario. It should be noted that in some frequency bins, the rate R_i can be selected to be zero, implying that no information will be carried at those particular frequency locations. This would be case if, in that portion of the RF spectrum, either there is on-going communications, or there is too much jamming-interference present. This ability of making frequency-agile transmitted waveforms is one of the unique advantages of using the OFDM modulation scheme which the proposed transceiver can exploits.

In addition, the transmission power and the number of bits per frequency bin, can also be adaptively assigned, based on the general principle of the classic 'water-filling' solution in the frequency domain. This can be done in several ways: by adjusting the code rate of the low-density parity check codes, or by manipulating the constellation size for the encoded information that is mapped to the OFDM modulation symbol sets. Finally, adaptivity can also be obtained by controlling the individual data streams that are transmitted from the independent antenna elements. Repetition coding techniques can be used, like the Alamouti scheme [1] discussed in Section 6.4.2, or the novel signal separation scheme as discussed in Section 6.4.2.

An implementation block diagram of the transmitter and receiver portions of the radio, the data packet structure, and a plot of the performance simulated

via Monte Carlo techniques are given below (Figures 6.14 to 6.18). It is seen that in normal SISO operations in frequency selective fading channels, the BER degrades very rapidly as channel conditions worsen. But the incorporation of LDPC coded OFDM counteracts most of the effects of multipath channel and makes the channel appear as an AWGN channel to the transceiver. The addition of MIMO further enables us to go beyond SISO hard bounds.

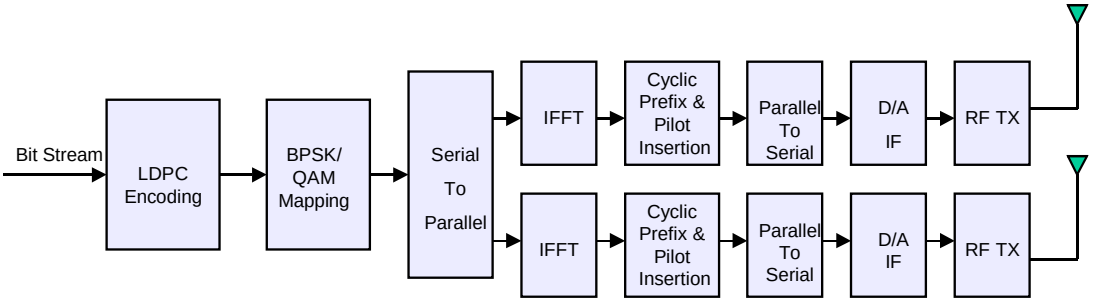

Figure 6.14: MIMO-OFDM-LDPC transmitter block diagram.

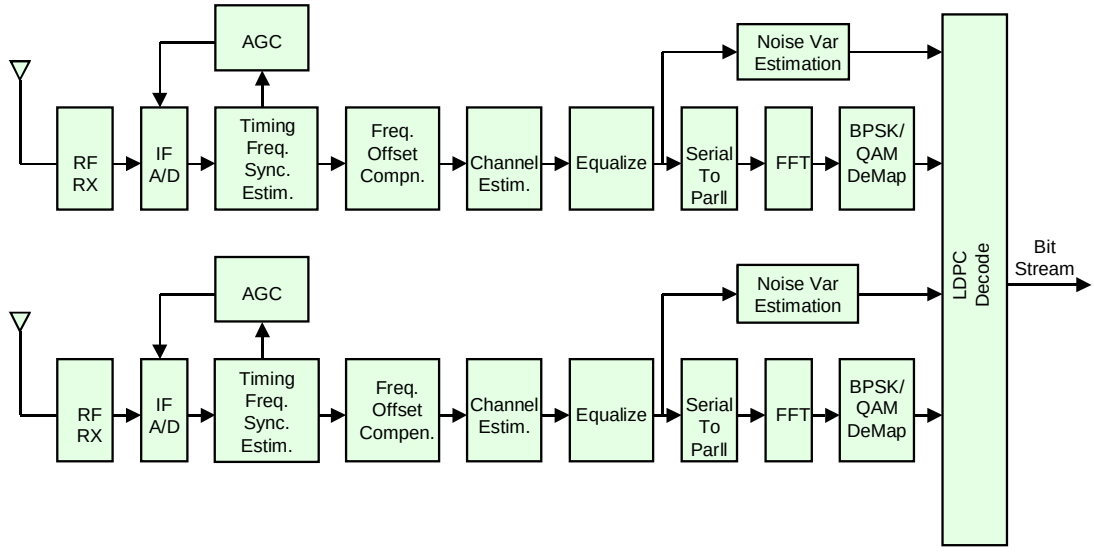

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Figure 6.15: MIMO-OFDM-LDPC receiver block diagram.

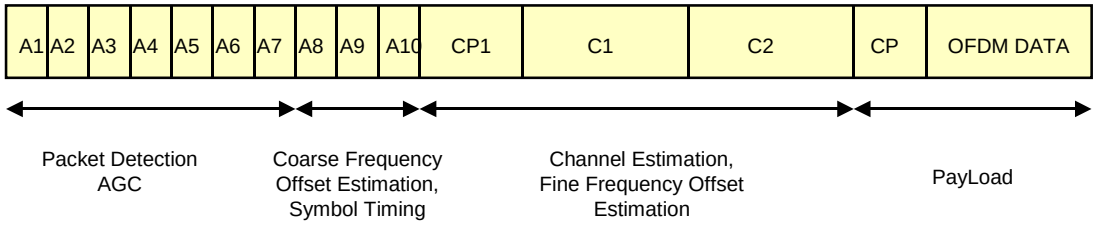

Figure 6.16: Data packet structure.

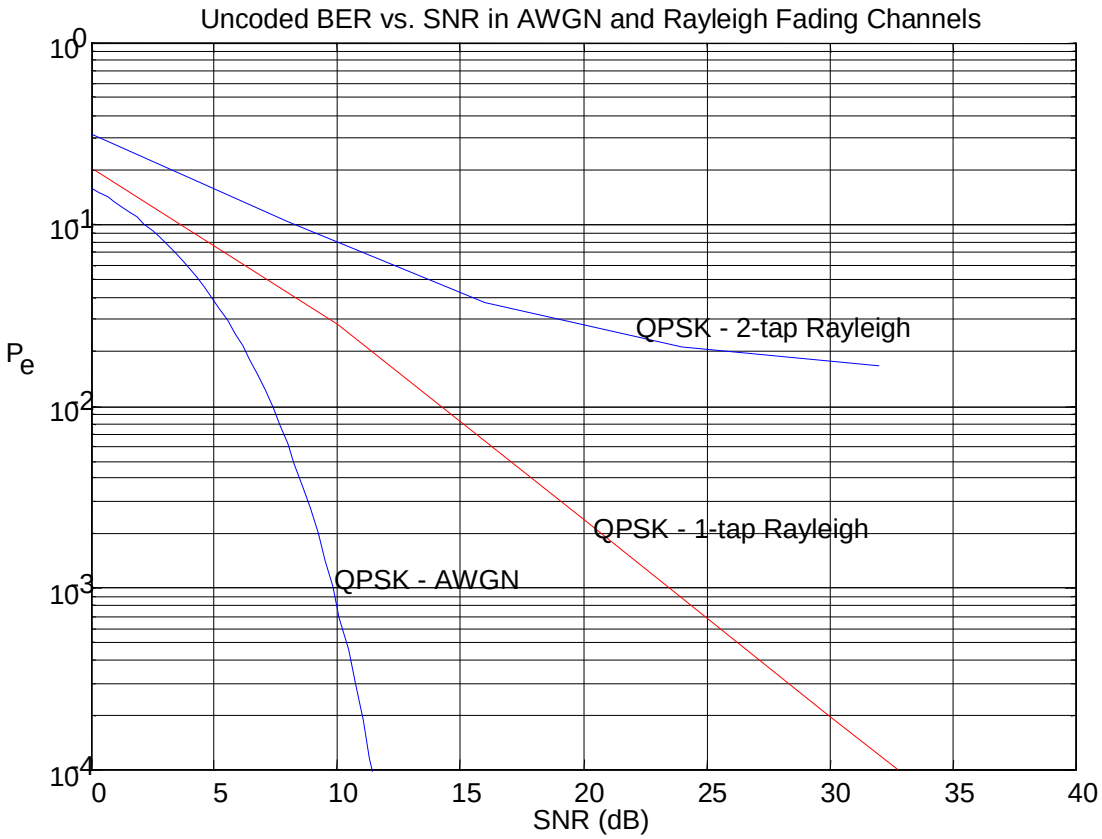

Figure 6.17: Simulation results: SISO bounds.

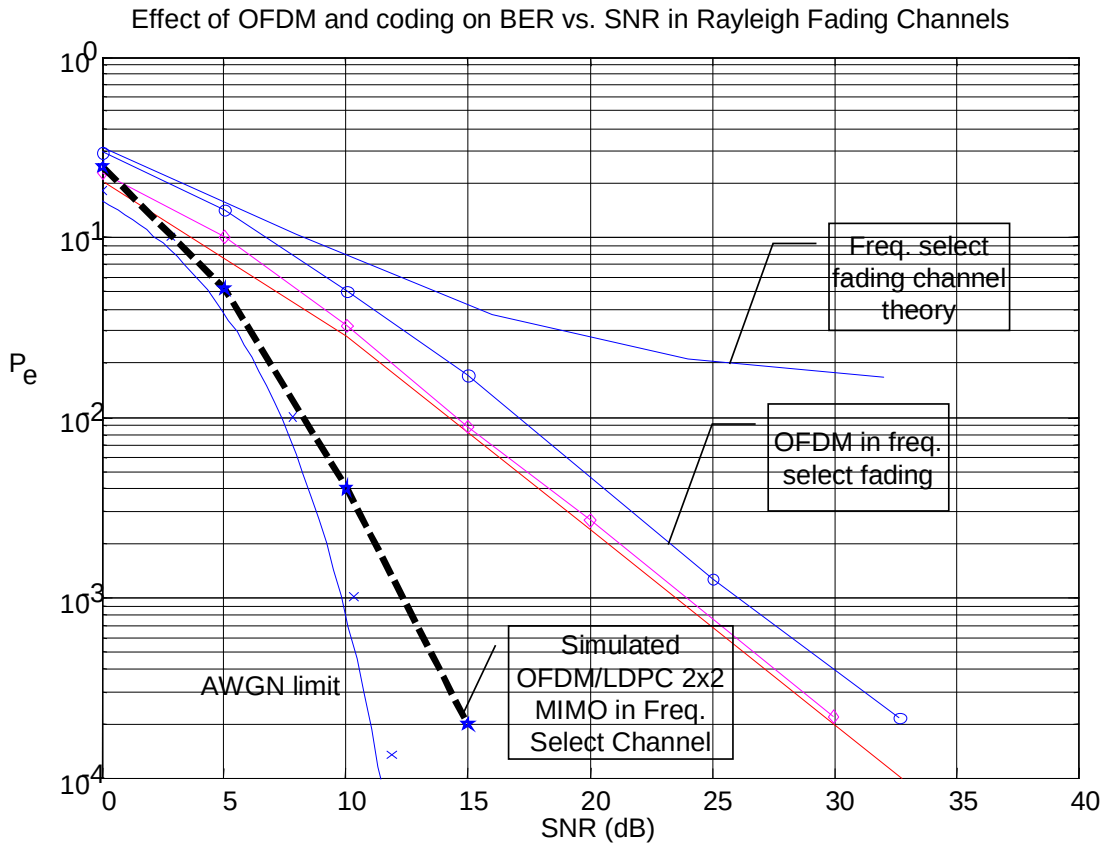

Figure 6.18: Simulation results: effects of LDPC-OFDM.

CHAPTER 7

Conclusion

This research effort has dealt with several inter-related problems relating to the processing of information in wireless networks, and has presented a collection of analysis, techniques and results designed to optimize performance in hybrid networks. Sensors were used as the starting point of the study, and rules were developed for a unified, consistent and efficient data fusion methodology, in heterogeneous, decentralized settings. The need for fundamental limits on performance in such networks was realized, and some bounds on the data rate and delay in idealized configurations were derived. Next, some practical issues dealing with the administration of wireless resources, for sensor networking, for example, was studied. In particular, a hybrid structure for connectivity among disparate, heterogeneous wireless nodes was presented, and optimization techniques were developed for efficient network performance under a variety of constraints. The dependability of such networks was also analyzed, and protocols that improved the dependability of the network were presented. Finally, bandwidth efficient techniques based on the latest developments in digital communication theory were proposed, that were able to achieve maximum spectral efficiency for node to node communications in the decentralized network.

The overall unifying objective that is common to these topics is the analysis of factors that affect performance when scaling the number of nodes in a sensor system from a few (when combinatorial methods for system performance may be

tractable) to many (when statistical methods are the only options). The goal is to determine intelligent unifying techniques from these varying analysis viewpoints, which can be applied to quantify and answer some of the basic performance limits questions for distributed sensing systems. In this regard, we believe this study is novel, and yields insights into the many facets of design for heterogeneous scalable, networked systems.

7.1 Future Directions

Several topics of further research and development, in each of the topics discussed in this thesis, can be gleaned from the work that has been completed to date. The broad goal will be as before: the development of tractable theoretical, modeling and simulation tools that can assist in the 'best-case' design and deployment of heterogeneous networked systems for particular mission objectives.

In the case of data fusion techniques, the Bayesian approach was seen as the central unifying tool. However, it may be mentioned at this stage that the basic philosophy behind the Bayesian probabilistic paradigm is not universally accepted, and there exists alternative theories for decision making. Two of these alternatives are *possibility theory*, also known as *fuzzy logic*, and *Dempster-Schafer theory* [99]. Both have reached a level of maturity and a measure of success to warrant their comparisons with the historically older probability theory. Such a comparative analysis is likely to be very useful for distributed information processing applications, such as sensor networking. It will yield invaluable insight as to the utility of the various methods in application specific cases, and may suggest ways to combine the methods and design hybrid techniques that can extract the benefit from all the tools that are avaialable to the system designer.

In the case of information theoretic bounds for sensor networks, many unanswered questions remain, unfortunately. The n-helper type bounds for non-Gaussian channels is of immediate interest, as is the case when the nodes are mobile. The development of the actual data compression and channel codes for these distributed, cooperating sensor platforms is another huge area of practical interest. It is not at all clear if the codes that have currently been developed and optimized for SISO systems are in fact the optimum choice for distributed, cooperating systems that can tolerate distortions. A thorough analytical and simulation study is desired to determine the tradeoff parameters and performance region. The general problem of satisfactorily combining networking and information theory is also a major unsolved area of research [19]. Delay bounds, similar to those calculated in Chapter 3, are required for practical sized networks with real queueing models. They also have to be derived for the mobile cases, which is where the QoS issues are of real concern.

For hybrid networks, there are a plethora of issues to be solved to enable true wireless connectivity, "anywhere, anytime." The approach taken in Chapter 4 is not the only option that researchers have considered. A comprehensive end-to-end system simulation study is necessary to weigh the relative merits of gateway centric approaches, as opposed to completely ad hoc or "random" schemes. Mobile wireless standards are still contentious issues, both commercially and from an academic point of view, and it would be most helpful to have a unifying analytical framework with which to analyze competing standards exhaustively. Needless to say, many higher layer protocol issues are still major driving forces in the wireless communications and networking community.

Reliability and dependability issues for ad hoc wireless nodes and networks, as discussed in Chapter 5, unfortunately, are still under-appreciated topics, often

coming as after-thoughts to the main network design. At best, the attitude that prevails is for enabling routing, MAC and other protocols to recover once failures have occurred (link breakages, node failures, etc.), and not necessarily designing networks that are fault tolerant and very reliable to begin with. A lack of usable analytical and computational/predictive tools and techniques are primarily to blame, and this thesis has attempted to address this shortcoming. Further work remains to be done to enable more comprehensive techniques that are applicable to real network systems, such as being able to incorporate protocol specific metrics for the various network stack operations. A modular approach, as is the case for cellular telephony systems, is called for. As mentioned in Chapter 5, the problems may turn out to be intractable from a theoretical or computational complexity point of view, but engineering approximations should be feasible in most cases of practical interest.

Finally, bandwidth efficient MIMO techniques are an exciting new arena in digital communications research. Its potential for precipitating a revolution in sensor networking cannot be underestimated. Especially in tactical and military applications, high data rate sensing/processing capabilities have become the central focus and of prime importance for enabling network centric and operational theater visualization applications. As illustrated in Chapter 6, software adaptive radios that can efficiently utilize the time-frequency-space dimensions are the current design challenges, and streamlined design/evaluation specifications are desired. Despite large theoretical hurdles (unified LDPC decoding characterization, efficient MIMO signalling techniques, joint source/channel coding for MIMO systems etc.), rapid progress is being made. Unfortunately, commercial interest has not yet caught on which has hindered the development of practical systems. Actual results from prototypes and field measurements are lacking, which may invalidate some cherished assumptions that many of use take for granted in the

design and simulation of these types of systems during pen and paper or software development (e.g. accurate MIMO channel models, frequency compensation performance for high data rate OFDM, true effect of mobility on MIMO performance, etc.). Also, comprehensive comparative evaluations of the many proposed schemes for OFDM and LDPC, and combinations thereof, are few. Work remains in this regard.

It is heartening to note, however, that the recent progress and projected advances in semiconductor technology, nano-electronics, wireless communications and system design tools are remarkable. These will no doubt enable the practical development, deployment and use, on a mass scale, of the types of wireless distributed information processing systems that this thesis is about. New questions and issues will inevitably arise. But, by necessity, the research questions of today will be tackled and hopefully lead to the solutions of tomorrow, and make the goal of ubiquitous wireless connectivity a reality.

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