Cross-Layer Optimization of Message Broadcast in MANETS

(REPORT BY GROUP 1039A)

Development of a model-based cross-layer optimization approach

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Synopsis:

In this work a model-based cross-layer optimization for reliable message broadcasting in mobile ad-hoc networks has been developed. The basis for this work is a Distributed Car-data for Assisted Driving end-user service, which has strict requirements on reliability in terms of the message failure probability metric P_{mf}, and performance in terms of the end-to-end delay metric D_{e2e}.
Analysis of resilience aspects in relation to broadcasting in mobile ad-hoc networks, has led to the selection of the layer parameters *transmission power*, *PHY mode* and *FEC code rate* for performing cross-layer optimization. These layer 1 and layer 2 parameters provide different resilience mechanisms for improving the metrics P_{mf} and D_{e2e} in the inherently hostile wireless environment.
Models of the physical layer, MAC layer and flooding broadcast mechanism are combined to achieve a means for performing optimization. The physical channel accurately models the 1 and 2 $Mbit/s$ modes of 802.11b, while taking into consideration the effects of path-loss and multi-path fading. The MAC layer sub-model is based on existing work. Both a model of non-persistent CSMA of that includes the effect of hidden nodes in ad-hoc networks, and the Bianchi model of the 802.11 exponential backoff scheme are considered. Finally a model has been developed for describing the behavior of flooding broadcast.
The proposed model has been tested in ns-2 2.29 enhanced with a realistic PHY layer. The model was found to suggest an optimal transmission distance of 210 m, while simulation results showed optimal distances of 160 m and 200 m for the considered high-contention scenario.

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Preface

This project has been produced on the 10th semester under the master program Distributed Application Engineering at Aalborg University in the period February 5th to June 18th 2007. The main theme of the semester is "Distributed Real Time Systems".

On this basis, a cross-layer optimization approach for message broadcasting in mobile ad-hoc networks has been developed.

The report is targeted at an audience with the same academic provision as the project author.

Report organization

The report is divided into chapters describing the steps performed in the development of the cross-layer optimization.

Structure

Chapter 1 is a general introduction to the concepts and definitions regarding resilience, dependability and cross-layer optimization used in this work. Chapter 2 is the preanalysis that introduces the considered service case, surveys existing work in cross-layer optimization and presents the challenges in cross-layer optimization. Chapter 3 is a specific resilience analysis of the considered service case, that clarifies the challenges herein. Chapter 4 is the modeling chapter that presents the overall model and sub-models that is used to perform cross-layer optimization. Chapter 5 presents the functionality of the evaluation framework used for simulating the considered service case. Chapter 6 contains verifications of the used models and tests are used to establish the cross-layer optimization. Chapter 7 is a discussion of the properties of the proposed cross-layer solution and possible deployment aspects. Chapter 8 contains the conclusion and outlook on future work and possible improvements.

Appendices

In supplement to the documentation, appendices have been worked out in order to document in-depth specifications of certain parts of the development process and the system as well as detailed test-results. These are presented in Appendix A-F.

Reading guide

Throughout the report references such as [Nickelsen and Grønbæk, 2006] are made to entries in the bibliography contained in the back. Abbreviations may be found in the list in the back of this report. In addition to the report, a CD-ROM is enclosed containing simulation traces, MATLAB scripts, source code and copies of downloaded references from the bibliography.

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Definitions

Optimization variables - The goal of an optimization is to maximize or minimize a selected set of metrics. An example might be to minimize the delay and probability of service loss, while maximizing the throughput. These metrics are denoted optimization variables.

Condition - Defines a factor which influences the operation of the considered networking system. A condition can be the amount of radio noise or the distance between a transmitter and a receiver.

Layer parameter - A specific parameter in a specific layer that can be adjusted to improve performance under given conditions. A layer parameter could be a timeout duration.

Parameter vector - A vector containing one or more layer parameters from one or more layers.

End-user service - A service located in a service providing node providing functionality to end-user client nodes with user applications.

User application - User application which makes use of the end-user services.

Ad-hoc domain - Ad-hoc network of nodes that are able to communicate in a single hop or multiple hops without using infra-structure.

Chapter 1

Introduction

The design of complex communication systems is divided into layers based on the ISO/OSI reference network model. Each layer represents a certain functionality, such as medium access, routing and transport. This layered approach reduces design and implementation complexity and allows individual layers to be replaced independently of the remaining layers, as long as interface definitions are maintained. For instance the application in the upper layer is and should be capable of running without modification whether the physical link is Ethernet, ADSL, wired or wireless. Within each of these layers much work has been committed to optimize communication protocols for performance in terms of primarily throughput and reliability. However, local single-layer optimizations do not necessarily lead to overall optimization of the entire protocol stack. Further, wireless networks become increasingly popular especially in mobility applications. Inherently wireless communication is unreliable. This increases the need to consider new approaches in such environments to ensure available and reliable, i.e. dependable, services with high resilience to faults. These points motivate research in CL optimizations.
Various studies show how the overall performance can be improved by introducing CL optimization. For instance the work in [Kellerer et al., 2003] seeks to optimize application layer video quality across multiple nodes by adjusting link layer parameters such as assigned bandwidth. In [Garcia, 2003] and [Nickelsen and Grønbæk, 2006] a classical problem with the inability of TCP to distinct between causes of packet loss is considered from a CL perspective. Also, CL approaches have also been applied to optimize physical layer power consumption in mobile devices [Zhao et al., 2003] while also considering application throughput. In addition to the enhancements that have been obtained in these specific examples, other aims of CL optimizations could be to:

- Reduce jitter and latency and improve performance
- Decrease timely overhead during a handover
- Improve resilience in relation to faults

With respect to the latter point, an initiative to investigate future High Dependability (HD) solutions is the EU funded HIDENETS research project[1].

[1] www.hidenets.aau.dk

In HIDENETS, HD concepts are investigated in ad-hoc car-to-car and car-to-infrastructure communication scenarios where services across wired and wireless networks motivate the investigation of fundamental end-to-end resilience problems.
In this work CL optimizations are specifically considered in these communication scenarios to find techniques for improving dependability, resilience and performance. Particularly in wireless scenarios node mobility and transmission dynamics increase the challenge of maintaining service dependability. Operating conditions continually vary, requiring adjustments of parameters in the entire protocol stack to improve resilience and performance. A specific prerequisite of the HIDENETS project is the use of Commercial Off-The-Shelf (COTS) hardware and software solutions. Consequently this work will take its starting point in optimizing existing protocols and mechanisms in a likely deployment scenario.

1.1 Resilience and dependability

In the context of dependable services in HIDENETS, a definition of *resilience* from [Svinnset et al., 2006b] is given: "Resilience in communication systems is the ability of the network or the system to provide and maintain an acceptable level of service while facing various challenges in normal operation." Thus the aim is to handle *potential* and *occurring* faults by adding resilience mechanisms to ensure system robustness in various cases. The concept of resilience both cover fault prevention (when a fault could occur) as well as fault tolerance (when a fault has occurred). In this work a *fault* is considered as some malfunction in a subsystem of the entire networking system considering nodes and links. Thus a fault may lead to another fault which eventually may lead to a *failure*. A failure in the remainder of this work refers to an end-user service failing to meet its requirements of performance and dependability.
The definition of resilience may be considered as in Figure 1.1. Faults may occur due to changing conditions external to a mobile node. E.g. mobility or noise can cause a link to fail causing a fault toward the node. Similarly a fault may occur inside a node due to a software or hardware fault. Resilience toward faults is handled by resilience mechanisms in the protocol stack like for instance Automatic Repeat reQuest (ARQ) and Forward Error Correction (FEC). Such mechanisms offer means to protect the end-user service from the faults and ensure the required performance and dependability. However, resilience often also comes at a price of adding data overhead and consume timely and hardware resources.

Thus *resilience optimization* considers the task of optimizing resilience to improve dependability, performance and typically resource consumption.

With the inherent unreliability of communication between wireless mobile nodes, obtaining dependable end-user services in this environment by adding resilience is an essential challenge. Figure 1.2 presents dependability from the end-user perspective for a service that requires some level of Quality of Service (QoS). Over time the *provided QoS* varies with the dynamic elements of the network. Depending on the properties of the service being used, the QoS requirements may also vary over time (not illustrated). For instance, if the QoS requirement is specified in terms of throughput, a Variable Bit-Rate (VBR) video stream would have varying QoS requirements.

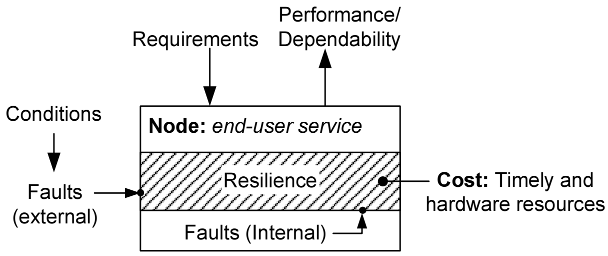

Figure 1.1: *Relations between conditions, faults, resilience, resources and performance/dependability.*

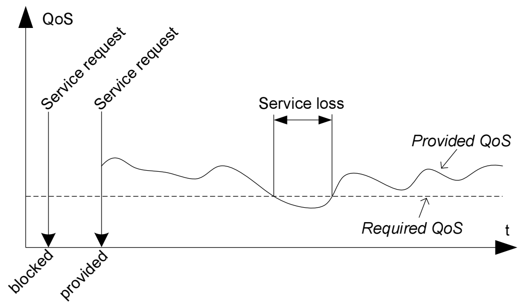

Figure 1.2: *Dependability concepts. A service is blocked if for some reason it cannot be provided upon request. When a service is in use, a service loss occurs if the provided QoS level drops below the required level.*

When a service is requested from a mobile node, the request can either be accepted or rejected, resulting in respectively *provisioning* or *blocking*, of the given service. A service might be blocked if the QoS required by the end-user service cannot be provided by the network. An example could be insufficient bandwidth or too high end-to-end delay for a video conference. The QoS requirement is highly dependent on the given service. For a service that is being used, the conditions might change in a way that renders the network unable to provide the required QoS. Typically, this would be due to an increased amount of *faults* such as packet losses, lack of resources or broken links. Failing to recover properly from these faults results in a *service loss*. The use of CL approaches, to optimize detection and recovery mechanisms in the network, might enable a recovery of the service by raising the QoS to the required level.

1.2 Cross-layer optimization

A general consideration of cross layer optimization is given in Figure 1.3. The protocol stack represents a mobile node. The mobility and use of wireless communication entails that many variable factors affect a node's communication capabilities. Examples of such variable factors are: channel noise, physical topology and available node resources.
These variable factors are not directly controllable and thereby define the *conditions* of the environment in which the node must operate. Depending on the

conditions, a node may benefit from adjustment of parameters in its protocol stack. For example, in a noisy environment the percentage of successful packet transmissions could potentially be raised by changing the modulation and coding schemes [Haratcherev et al., 2006] in the physical layer. The parameters that can be adjusted in the protocol stack are represented generally by the *layer parameters* in Figure 1.3. These *layer parameters* are gathered in parameter vectors $\bar{\theta}_i = \{\alpha_1^i, \alpha_2^i, ..., \alpha_n^i\}$ and $\Theta = \{\bar{\theta}_1, \bar{\theta}_2, ...\bar{\theta}_7\}$.
Optimization by settings of layer parameters is performed in relation to one or more *optimization variables*. In this work these variables are sorted in *performance metrics*, *resilience metrics* and *dependability metrics* enabling optimization in relation to both performance and dependability in service provisioning. The performance, resilience and dependability metrics that are referred to in this work are defined in Appendix A.1.
Observations in the layers of the protocol stack first and foremost deliver information to appraise the optimization variables for evaluation purposes. Additionally, observations contain information to estimate states of communication, which can be used when determining which resilience strategy is optimal in a given state. It applies to layer parameters and observations that they can be considered in either the local node solely or from a *cross-node* perspective. An observation of link congestion on a remote link in a remote node is an example of a cross-node observation.

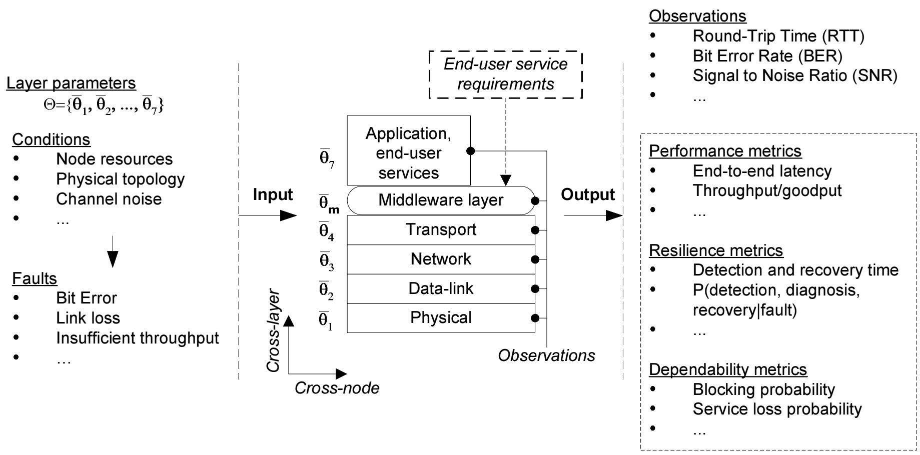

Figure 1.3: *Cross-layer optimization overview.*

The protocol stack is considered containing a set of specific end-user services in the application layer as well as a middleware layer. The middleware layer contains services to provide high dependability features and mobility functionality to end-user services. From the end-user services, requirements on performance are given. Depending on the end-user service type, different requirements to the QoS may exist, which could be expressed as specific performance requirements of for instance throughput, jitter and latencies. Sometimes the term *QoS* refers to the control mechanisms that ensure fulfillment of the requirements, but here it should be understood as specific performance requirements.
The main focus of this work is to consider the improvements of dependability by CL optimizations of resilience metrics. This also implies considering optimiza-

tion of the performance metrics as they, together with dependability metrics, describe the provided QoS and thus must be considered to evaluate the dependability (See Figure 1.2).
In order to ensure the highest degree of resilience, optimization of resilience metrics should be performed by adjusting the layer parameters appropriately. For some layer parameters an optimal choice might be a constant value that is suitable for all the considered states of the communication system. For instance this could be the case for the network layer packet length in a channel with fairly stable fault conditions. However, due to changing conditions some layer parameter types may require a continuous adjustment to provide optimality. A dynamic adjustment would obviously be necessary for the choice of coding and modulation scheme in an environment with varying levels of noise.

In addition to determining whether a static or dynamic optimization is suitable for a given layer parameter, the scope of optimization should also be considered. Some optimizations may not need adjustment of parameters and usage of observations across layer boundaries. This makes them *single layer* optimizations. *Cross-layer* optimizations, on the other hand, rely on observations from multiple layers and/or adjustments of parameters in multiple layers. In addition, optimizations may target improvement of certain aspects of resilience and performance in the *local* node only, based on local and cross-node observations. However, if the objective is an improvement of a complete ad-hoc network domain and fairness to all nodes, this is a *global* optimization. The possible scopes of CL optimizations in the layered network model are shown in Table 1.1.

Layer parameters and used observations are:	
Local node single-layer	Global ad-hoc domain single-layer
Local node cross-layer	Global ad-hoc domain cross-layer

Table 1.1: *Scopes of protocol optimizations.*

1.3 Aims in CL optimization

The overall goal of this work is to investigate the possibilities for Cross-Layer (CL) resilience optimization in the context of the HIDENETS framework. The HIDENETS context implies that the domain of relevance concerns volatile wireless ad-hoc networks with mobile nodes. In such networks the volatility introduces potential faults that entail an increased need for enhancing resilience mechanisms. Due to limited link capacity congestion is common in wireless networks [Corson and Macker, 1999]. As a consequence there is a limited free capacity for resilience mechanisms in the communication. Thus any excess overhead should be avoided while providing enough resilience to cope with occurring faults and ensuring a required QoS.
Specifically, the focus is on optimizing the use of resilience mechanisms, which includes improvement of dependability of end-user services. Resilience is defined as the ability to withstand potential and occurring faults to fulfill performance requirements of an end-user services. To enable this perspective a representative end-user service case is defined. The case must specify the used protocols and include performance requirements. Additionally fault models are specified to

describe which faults to be resilient toward.

In order to adjust resilience mechanisms in the used protocols, it is necessary to identify adjustable layer parameters and their impact on resilience and performance. The layer parameters should be identified within the scope of protocol stacks similar to the defined case.

Finally, as this work is in the context of HIDENETS it should be investigated how actual CL optimizations can be realized in a Commercial Off-The-Shelf (COTS) based system. An important consideration is the impact from enabling CL optimizations. Since the CL perspective potentially breaks the independence of the layers in the layered network model, this may increase complexity significantly. I.e. protocols in different layers may become increasingly difficult to design and implement as they rely on each other. Consequently architectural considerations are highly relevant to make when designing CL functionality (see Section 2.5 on page 35).

In the following chapter these topics are discussed to identify highly relevant CL optimization possibilities and issues that arise in the process of introducing CL resilience optimization.

Chapter 2

Pre-analysis

The aim of the pre-analysis is to define possibilities and challenges in designing, implementing and deploying Cross-Layer resilience optimization. Based on the HIDENETS context the chapter will take a starting point in defining a car-to-car end-user service case in section 2.1. The case is used to define the wireless environment considered in terms of physical resources and mobility.
In section 2.2, existing work in the area of CL optimization and ad-hoc wireless network is investigated to identify relevant layer parameters and their impact on performance and resilience. Following, section 2.3 presents a discussion of relevant issues regarding CL optimization. Section 2.4 discusses general optimization methods which is extended in section 2.5 to consider architectures for realizing CL optimizations in COTS systems. Finally the problem statement in section 2.6 outlines specific problems which are considered in the following work.

2.1 Distributed car-data service case

In this section a service case is presented. The case is used in this chapter to define the problem domain of services operating in a wireless mobile environment. Specifically focus is on the ad-hoc network domain where large dependability issues exist due to the ever changing network operating conditions from mobility and the environment. Different cases have been considered for this work (see Appendix B) however only one has been chosen for detailed specification and analysis. In the selection of a service case, focus has particularly been on high dependability and real-time requirements. Thus the profile of the service must contain such aspects to enable an identification of the challenges of providing dependable services in an ad-hoc domain.
The outcome of this case analysis is a set of QoS requirements, a specification of the technical prerequisites for the operation of the service and a definition of the operating environment and its conditions. Finally an overall fault model is constructed to enable analysis of relevant resilience mechanisms needed in the optimization process.

2.1.1 Service description

Using wireless communication all cars in a stretch of motorway exchange information of their position, velocity, acceleration and usage of car controls i.e. steering, brakes and accelerator. This information is used to create a picture of the current traffic situation, which in a large scale can be used to detect and/or predict traffic congestion and accidents. In a local scale the information is shared between cars within some proximity of each other. This allows each car to discover potential hazards. The driver is warned so that an accident can be avoided and the safety systems in the car are prepared for a potential crash. In cases where the driver cannot react in a timely manner or fails to react to a warning a car may itself start braking or perform an evasive maneuvre. The latter action is denominated *assisted driving*; a concept which has already started to emerge in production cars [Presse-Information, 2006].
This service case is called *Distributed Car-data for Assisted Driving* (DCAD) and is a derivation from the service cases Floating Car Data, Platooning and Hazard warning of HIDENETS defined in [Svinnset et al., 2006a]. In this case specifically the assisted driving aspects define requirements that are critical to maintain making this derivation particularly interesting. Existing work in a service very similar to DCAD can be found in [Strassberger and Lasowski, 2006].

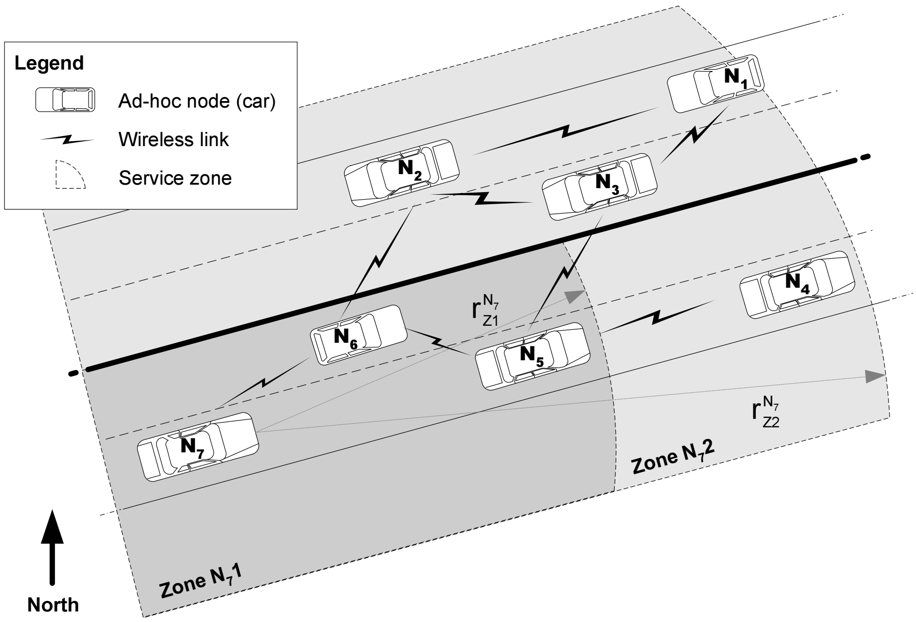

Figure 2.1: *The physical scenario of the "distributed car-data for assisted driving" service case. Zones are defined to describe different geographical areas in which information is relevant and is to be propagated. As an example the service zones 1 and 2 are depicted for node N_7.*

In the considered service case each car represents a mobile node. Note that the terms *node* and *car* are used interchangeably. The cars communicate in adhoc networks using layer 3 broadcast to distribute information to neighboring cars in one or more hops. The physical scenario is depicted in Figure 2.1. It consists of a motorway with four lanes; two in each direction separated by a heavy guardrail. A group of cars is heading in each direction. Some of the cars are in radio communication range of each other which is indicated by lightnings. Thus cars may communicate with each other regardless of their individual heading.

Geographically limited service provisioning

In general a service consists of the *end-user service* and the actual *user application* which makes use of the service. In this particular case each node acts as a service user as well as a service provider. Thus all cars have an instance of the end-user service and the user application running, while they respectively broadcast and receive information. This makes the communication type multipoint to multipoint. The radius of spreading the information may be limited geographically from the source and outwards to avoid irrelevant information to spread in the entire ad-hoc domain. Such a geographical limitation is given by a *zone*. Multiple zones exemplify how different levels of this service exist. Individually each zone defines a set of requirements for the pace at which information spreads and its spreading range. These requirements are set by the end-user service meaning that the notion of a zone is defined in the application layer. However, to geographically limit the spreading of messages the broadcast mechanisms in layer 3 must also be aware of the zones and node locations.

As an example Figure 2.1 depicts the DCAD service with two zones from the perspective of the node N_7. The depicted zones N_71 and N_72 have their origin in node N_7 and describe an area in relation to node N_7 defined by the radii $r_{Z1}^{N_7}$ and $r_{Z2}^{N_7}$. Similarly, each node N_n has its own notion of a zone N_n1 and N_n2 with an origin in itself and the radii $r_{Z1}^{N_n}$ and $r_{Z2}^{N_n}$ where $n = 1, .., 7, ..$ The end user-service in node N_7 broadcasts information about its own behavior to cars in vicinity to itself. This information is restricted to the area of zone N_71 where cars must be capable of reacting to the driving behavior of node N_7. For cars traveling several kilometers ahead in the same lane this information is irrelevant. Also cars in the opposite drivers lane will not be affected if N_7 slips out and crashes under the assumption the guardrail takes the impact. Consequently zone N_71 of node N_7 only covers the eastbound lanes. However, it must still be taken into consideration that a car in the opposite direction, e.g. westbound still may play an important role as a relaying station between two cars in the same eastbound lane. While information range is limited in zone N_71 there may still be several hops between sender and receiver node.
Other types of information may be relevant to cars further ahead and cars in the opposite drivers lane as given by *zone* N_72. For instance, eastbound node N_7 could have collected information relevant for the westbound node N_2 that the road is icy further ahead or that a fog bank covers the road.
In this work, for simplification, only zone N_n1 is considered where requirements are assumed to be equal for all information collected and distributed.

Reliable broadcast

Especially for driving assistance purposes the DCAD service is mission critical. If broadcast messages are lost accidents could potentially occur. This stresses the importance of high dependability. It is assumed that the DCAD service is always required and always available when a car enters an ad-hoc domain. Thus

dependability is defined from the reliability of the service alone.
The DCAD service relies on reliable broadcasting. Reliable broadcasting in ad-hoc networks is a highly relevant research topic as will be presented in section 3.1. For now broadcasting is merely defined as a layer 3 functionality where the domain of the broadcast is geographically limited as defined by the zone notion in the DCAD service. In section 3.1 the underlying mechanisms of the considered broadcast functionality are discussed in detail.

Service protocol stack
Based on the service description a preliminary COTS protocol stack can be introduced. It describes which assumptions are made for the functionalities in each layer. Potential layer parameters and optimization metrics are identified in section 2.2 from existing work.

L7 - DCAD service The DCAD consists of two individual processes: (I) The DCAD end-user service which collects information from the car systems and broadcasts it and (II) the service user application which collects broadcasted information from surrounding cars to create a map of their locations and movements. This information is used in assisted driving applications to control the car e.g. in emergency cases.

LM - HIDENETS middleware Standardized HIDENETS middleware is located in each node. It provides multiple internal services to aid dependable end-user services operate in ad-hoc and infrastructure networks.

L4 - UDP The DCAD service does not rely on end-to-end connections. Information is broadcast between nodes in a multipoint to multipoint manner. Thus the transport is based on UDP.

L3 - IP-routing and broadcast Broadcasting is executed in this layer. As the broadcast is geographically limited routing algorithms must consider this while maintaining an efficient forwarding of messages between nodes.

L2 - 802.11 MAC In this layer CSMA/CA and DCF is used for Media Access Control. For more information see [Tanenbaum, 2003].

L1 - 802.11 WLAN A standard COTS 802.11 WLAN network interface is used for wireless communication.

In Table 2.1 the Protocol Data Units (PDUs) specifications are given. From the application perspective broadcasting is related to the dissemination of messages. A message contains the application data and target all nodes within a specific zone. A frame is oppositely defined at the link layer where a transmission covers the dissemination from a node to other nodes. Typically multiple frames must be transmitted and re-transmitted in several hops to cover the entire range of the zone and thereby support the message dissemination.

2.1.2 Service requirements

The DCAD service is defined from its specific requirements to communication. In this section assumptions are made of how the service potentially could operate. As an outcome a set of service parameters are defined and their values

Layer	**PDU**
Application	Message
Network	Packet
Link	Frame

Table 2.1: *PDU specification.*

are set. These values also define what the service requires to function correctly. This subsequently leads to specific QoS requirements.
Each car is assumed to implement a closed loop control mechanism allowing them to adjust throttle, steering and brakes for assisted driving. This provides a basis to define the service parameters as presented in Table 2.2. The QoS requirements are explicitly given by the service parameters in the lower section of the table. Arguments for these requirements are presented in detail in Appendix B on page 148.
The requirements are defined for a single user application in node N_a receiving service from a single service provider in node N_b. If another node, N_c, also provide service to N_a the QoS is evaluated at N_a individually for each service providing node N_b, N_c, etc.
Whether a node N_a is within the zone of node N_b, and should receive its services, is defined by node N_a's lane direction and euclidean distance to N_b. This distance is evaluated in the moment N_a receives a message or, in case it is lost, at latest should have received a message from N_b.

2.1.3 Assumptions of operating conditions

The conditions under which the DCAD service typically operates must be established to analyze the challenges faced by the service to maintain reliability. These conditions are defined by the cars constituting the *nodes* in the ad-hoc domain(s) and their mobility in the surrounding *environment*. In this section considered conditions and their parameters are introduced. In the next section they are subsequently used to form an overall fault model describing threats toward the service. Thus a basis is created to later define which resilience mechanisms are required. Further the introduced elements are used in Chapter 5 to define a simulation environment enabling evaluation of CL optimization proposals.

Environment and mobility
The environment is introduced from Figure 2.1 depicting a stretch of motorway. The surroundings are rural and consist of flat open fields with sparse high vegetation. Consequently, reflections of wireless signals are limited which also limits the amount of *multipath-fading* and *shadowing* compared to an urban environment with buildings [Eude et al., 2005]. Potentially, the environment could also contain sources of noise that emit radio waves in the frequency band of the wireless network. For simplification these are not considered. However similar disturbances are created from mobile nodes that are not within communication range but still close enough to cause *interference*.

Mobility of cars has a large influence on the dynamics of ad-hoc networks.

DCAD service parameters		
Name:	*Description:*	*Value*
`Zone range`	The range at which the data payload is relevant I.e. zone $N_1 1$ in Figure 2.1.	300 m backwards, 300 m forwards
`Payload size`	32 bits position, 12 bits speed, 12 bits direction, 12 bits acceleration vector, 32 bit other status information (wheels, brakes, suspension), 32 bits sensor values, 104 bits application layer protocol overhead.	30 *bytes*
`Broadcast frequency`	Depending on its own velocity v_{car} each node initiates a broadcast containing the *data payload* according to this frequency.	$10 - 50\ Hz$
QoS service parameters (user application)		
`Throughput`	This parameter is derived from the payload size and broadcast frequency.	$1.500 B/s$
`Max latency`	The maximum latency allowed before a message with the *payload* has propagated to all nodes within the *range* of the zone.	160 ms
`Jitter`	99% of all messages must be within this range of jitter in relation to a mean delay measured over a 500 ms window.	$\pm 10 ms$
`Message BER`	The allowed BER in the application layer.	$< 10^{-4} BER$
`Max message loss`	Maximum message loss allowed. If messages are received failing the jitter and BER requirements they are counted as lost. NOTICE: Two consecutive messages may never be lost.	1%

Table 2.2: *Parameters for the end-user service and its requirements for operation. The QoS requirements are defined for a single user application in node N_x receiving service from a single service provider in node N_y.*

The mobility is affected by the road conditions for instance if there are traffic light systems, approach roads, speed limits etc. Also, the driver behavior defines how cars follow each other, overtake and act in relation to traffic rules. Clearly, these aspects are overall affected by the amount of traffic present in the traffic system. The resulting motion patters inherently define distances between nodes, their grouping and individual node velocity. This translates into the forming and breakage of links as well as changes in interference and reflections of wireless signals.

To evaluate solutions of CL optimizations in realistic conditions a central task would be to define a mobility model. A starting point should be taken in the introduced road stretch. Cars could be controlled by models of traffic behavior which could be stochastic, hydrodynamic or driver behavioral as presented in [Haerri et al., 2007]. In this section only the preliminary properties of the mobility aspects in the scenario are defined. The representation of the road, used for evaluation of CL optimization methods, is discussed and presented in section

5.1.
The stretch of motorway considered is 1 km and straight. A node travels in eastern or western direction in either the inner or outer lane. For now a single model parameter is introduced. It is the *traffic density* which defines how much traffic is present on the motorway. It is defined from a set of car arrival rates λ_{EBcar} and λ_{WBcar} into the East Bound (EB) and West Bound (WB) lanes of the road stretch. Defining an individual rate for each direction enables a control parameter of oncoming traffic.

Node properties
Each car is equipped with an on-board computer which hosts the DCAD service processes introduced earlier. For simplicity it is assumed that all nodes are equipped with the same radio, omni-directional antenna, processing and storage resources and properties. In reality this is not likely to be the case. Differences would cause more conditions to consider e.g. variation in antenna efficiency, available processing power for resilience mechanisms and relaying of messages etc. This should be considered in future work when deploying cross-layer solutions in real operation scenarios.
In contrast to many ad-hoc scenarios power consumption is not a predominant issue in the car-to-car scenario. Cars continually produce electricity. Thus optimization of power consumption is not considered a primary performance variable.
The DCAD service is not the only service competing for node and wireless resources. As specified in HIDENETS several end-user services and service users operate in the wireless nodes. These are conferencing, gaming and streaming video/radio just to mention a few [Svinnset et al., 2006a]. The result is cross traffic on the links as well as in internal network queues in the protocol stack of the individual nodes. Influences from other traffic must be evaluated from their impact on the DCAD service and its requirements.

2.1.4 Overall fault model

In this section the conditions have been used to identify which faults could appear in relation to the protocol stack. These faults are further relevant to consider which resilience mechanisms are needed to improve reliability of the DCAD service.
A hierarchy of faults is depicted in Figure 2.2. Next, the key aspects of the faults are discussed.
The arrows describe the causal relations between conditions and faults. All faults initiate from one or more conditions with obvious repercussions from the mobility and properties of wireless links. Eventually, intermediate faults may lead to a fault in the application layer depending on their severity and the strength of the resilience mechanisms to resist them. The application layer faults are in this case equivalent to a breakage of the QoS and consequently a service failure and loss of reliability.
In Figure 2.2 resilience mechanism are not directly considered. However some of these faults only exist given the presence of these mechanisms. Specifically this accounts to delays in the protocol stack. For instance *link contention* is controlled by the CSMA/CA mechanism. If a frame is lost during transmission

it is retransmitted until it successfully completes (or some retry limit is reached). This process will delay the frame transmission (causal relation marked by an '**R**') and eventually the packet. It is important to stress that *delay* as a fault in this work is defined as the added overhead to natural latency that is unavoidable in e.g. transmission. The severity of delay faults is in all cases defined from the application layer where the specific requirements for delay (and jitter) are set.
As a special case a causal relation given by a resilience mechanism is established from the *bit error* fault directly to the application layer. Typically the physical layer supports different channel modulation schemes where the trade-off is a lower channel data rate at the gain of minimizing bit errors. Thus bit errors can cause a reduced throughput in the application layer without being caused directly by packet loss.

Routing is a central aspect of communication in ad-hoc networks. Topology changes provide a challenge to ensure that messages are transmitted efficiently to the correct recipients. Thus the fault *routing error* is defined by a wrong decision in the routing mechanisms causing (1 - packet delay) messages to travel an inefficient path, (2 - datagram duplication) multiple identical messages to be received at the same node and (3 - datagram reordering) messages to arrive in the wrong sequence at the receiving node. The latter fault would in this application case lead to the oldest message being discarded.
In the environment two main faults are defined to potentially cause a link loss. Primarily mobility can cause two nodes to move out of range of each other. Another fault would be a node that sporadically disappears due to a software or hardware error. However to delimit the extend of the fault model, in this work software and hardware faults are not dealt with.
A central topic in ad-hoc environments is the trustworthiness of communication partners. It must be taken into consideration that other nodes could maliciously or accidentally broadcast and/or relay wrong information resulting in service failure [Avizienis et al., 2004]. Overall, such faults are vitally important to consider when applying resilience mechanisms. I.e. redundant messages in different paths could be necessary to ensure correctness of information. Again, to delimit the fault model these aspects are considered a part of future work when extending the introduced fault model.

To further specify how the presented faults and conditions can affect the operation of the DCAD service their pattern of occurrence and underlying models must be studied in detail. This is done in section 3.1 on page 45 where a subset of this model is established for further analysis.

2.2 Existing work in cross-layer optimization

In this section cross-layer optimization topics are discussed in relation to the environment of the presented case and existing work currently done in the area of cross-layer optimization in wireless and ad-hoc networks. The aim is to define possible CL optimization topics offering to strengthen resilience and performance in car-to-car ad-hoc networks. Based on the existing work, challenges in CL optimization are identified for a discussion in section 2.3.
This section is organized from the layers of the OSI reference model from the bottom and up. For each considered layer the starting point is taken in key *optimization variables* specific to this layer. The considered optimization variables

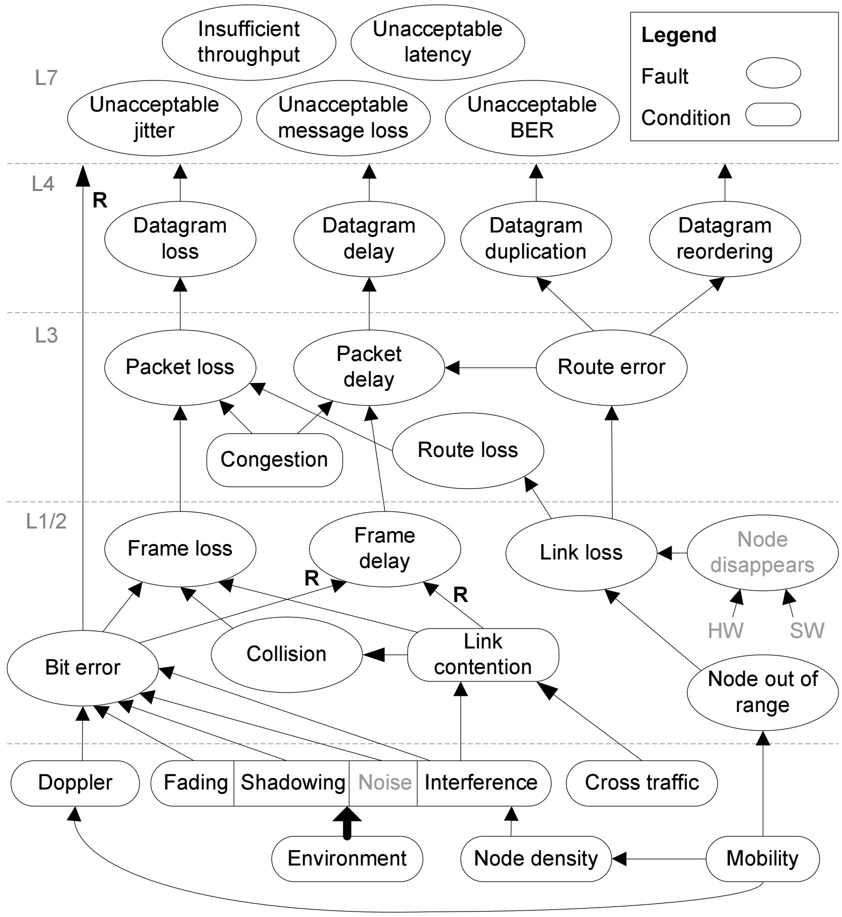

Figure 2.2: *Overall fault model in the protocol stack of the distributed car-data assisted driving service.* ***R*** *defines when resilience mechanisms are involved in the transition of one fault to another. Grayed faults and conditions define delimitations not considered further.*

common to the layer are emphasized initially. Each topic is further related to which *layer parameters* are used and which *conditions* are assumed. Both work with focus on resilience and performance optimizations is considered, as these topics are closely related.

2.2.1 Physical layer - transmission optimization

Optimization variables: **Energy consumption and Bit Error Rate (BER) (rate adaptation)**.
Considering initially the physical layer, a main optimization issue is energy consumption. As mentioned previously this is not a major issue in car-to-car environments. In general however it is a large research topic in the area of ad-hoc networks where mobile devices only rely on battery power.
An approach is to minimize energy consumption used in transmission. The work

in [Lettieri et al., 1999] considers energy efficiency in layer 2 resilience mechanisms ARQ and FEC in a wireless environment. Specifically fault handling is costly in terms of time and energy considering retransmissions, redundant bits that must be sent and additional processing power required. It is demonstrated how these resilience mechanisms can be adjusted to meet layer 7 application requirements of throughput and latency while maintaining a minimized energy cost. The authors argue how optimality of resilience (packet loss) and energy costs highly depend on the BER of the channel. This motivates a *dynamic* adjustment of the resilience mechanisms i.e. code rate of FEC, maximum amount of frame retransmissions and frame size. Similar work has been conducted in [Zhao et al., 2003] where a cross-layer approach is used to select the transmission power level and physical mode for IEEE 802.11a. The aim is to find an optimal ratio between throughput and energy consumption given an amount of mobile users and the Signal to Noise plus Interference Ratio (SNIR). By modeling the link-layer MAC schemes of CSMS/CA and RTS/CTS (see [Tanenbaum, 2003]), analytic models are derived and used to dynamically adapt power levels and physical mode.
Dynamic adjustment is a general CL issue that is discussed in Section 2.3.
A related layer 1 topic also considers transmission power levels. In a typical wireless environment nodes share the communication medium. Considering a single node it will be capable of communicating with nodes within a given range. However, it will interfere with other nodes at an even longer range, increasing link contention. In this case topology control mechanisms could be applied to adjust power levels or channels of each node separately [von Rickenbach et al., 2005]. This would allow an optimization of the access to the communication medium given knowledge from other layers in terms of communicating nodes, node locations and active routes. This approach raises another interesting CLO issue. Apart from considering node-local CL optimizations only, several nodes in an ad-hoc domain could collaborate to improve the overall optimization metrics and ensure fairness. This issue is also discussed in section 2.3.

Data rate adaptation

Different physical modes exist in 802.11 wireless networks and are defined by the used channel modulation and coding scheme [IEEE, 1999]. Each mode defines an individual data rate where higher data rates are increasingly sensitive to a degraded Signal to Noise Ratio (SNR). Layer 1 and 2 cross-layer approaches have been suggested to make an optimal selection of the data rate to reduce the BER. The existing strong bonds between these two layers make it advantageous and possible without introducing much complexity.
The authors of [Yuen et al., 2002] have extended the RTS/CTS frames (layer 2) to allow a receiving node to inform a transmitting node which data rate (layer 1) it should use based on an SNR measurement (layer 1) of each RTS frame. Clearly, the SNR measurement is most precisely estimated at the receiving node as this is where the transmitted signal is to be interpreted. With this information the data rate is continually adjusted to the transmission conditions. This causes a significant decrease in the amount of failed transmitted frames and consequently an increased throughput and decreased delay due to fewer retransmissions.

2.2.2 Network layer - Routing mechanisms

Optimization variables: **Packet throughput, delay and loss**.
Obviously a large topic in ad-hoc networks is routing of messages in the dynamic environment of mobile wireless nodes. As conditions continually change so must the routing mechanisms dynamically adjust how packets are distributed in the network. The challenge is to find the optimal route to increase the probability of messages reaching their destination in a timely manner. In this case optimality could potentially be defined from end-user service requirements toward performance and resilience. I.e. the routing paths could be chosen for each node to comply to multiple constraints of e.g. connection lifetime, packet loss, delay and throughput.
Existing established routing algorithms in ad-hoc networks principally rely on layer 3 routing metrics only. Examples are Ad-hoc On-demand Distance Vector (AODV) [Tanenbaum, 2003] and Dynamic Source Routing (DSR) [Johnson et al., 2007]. DSR is particularly popular in the research environments as it is simple and has proven to be fairly effective. This routing algorithm is based on a broadcast route lookup scheme where a shortest path routing metric is used to determine the route. However, the shortest route may not have the optimal conditions to ensure a required throughput or delay if it contains low bandwidth or highly utilized links [Yuen et al., 2002]. To add essential routing metrics cross-layer improvements are useful. Especially link information has been used to determine optimal routes. Besides rate adaptation, described in the previous section, the authors of [Yuen et al., 2002] also introduce routing metrics of link bandwidth, link congestion and link interference. The latter metric is obtained in the different wireless links measuring the delay from the first RTS for a frame until it has been successfully sent. A high delay indicates frame loss, multiple retransmission attempts and thereby contention and interference on the link. Adding this rate adaptation and link interference routing metric to an existing implementation of DSR, substantial improvements are achieved. In a specific test scenario the throughput is improved by 40% over normal DSR while the delay is decreased by 50%. Similar good results have been achieved in [Yang et al., 2005] with similar metrics.
In order to handle the continual topology changes, a control mechanism is needed e.g. to make decisions of route changes [Sun and Hughes, 2005]. In general such a decision entity is of relevance whenever dynamic adjustments of layer parameters are required to ensure optimal operation. Potential approaches to solve such decision problems are discussed in Section 2.4.

2.2.3 Transport layer - Transport protocols

Optimization variable: **TCP Throughput**
The transport layer contains substantial topics for cross-layer optimization. Specifically there is the well-known issue with TCP [Tanenbaum, 2003]. TCP was designed for reliable wired links and consequently always assumes congestion to be the cause of packet loss. The recovery reaction of TCP is to reduce the transmission rate to avoid congestion. In cases of wireless loss however, the correct action would be to retransmit the lost packet quickly and maintain or increase the transmission rate. Many solutions for this problem have been suggested. In particular a set of these solutions consider cross-layer options

where information about link losses are propagated to the transport layer. One of the solutions is called Explicit Loss Notification (ELN). In ELN the nodes on a path in a network are responsible for sending a flagged acknowledgment back to a TCP sender if a wireless loss has been observed [Barman et al., 2004]. This allows TCP to react accordingly. The concepts of ELN demonstrate how a cross-layer approach also could have a cross-node perspective. I.e. in this case the transport layer of the TCP sender is notified based on link layer observations in a remote node.
In [XIAO et al., 2005] the authors also use cross-layer optimization to make TCP aware of wireless properties. Specifically a TCP source may experience less contention within its own wireless range than a node in the route forwarding IP traffic toward the TCP sink. Consequently the TCP source will transmit more than the multiple wireless link can handle causing contention and eventually packet losses. The CL approach adjusts TCP maximum window size based on link layer retransmission counts to provide contention based flow control. In a random topology with 50 nodes improvements of 18-34% have been achieved with this approach compared to ordinary TCP.
The authors of [Barman et al., 2004] describe various approaches related to optimization of TCP performance in wireless networks considering layer parameters of transmission power (layer 1), FEC and ARQ (layer 2). In the work it is pointed out that cross-layer optimization does not necessarily require sharing of information between layers. E.g. link layer mechanisms could be TCP aware by trying to minimize packet loss based on models of known TCP behavior. From such models performance functions can be derived and used to optimize TCP performance and resilience by link layer parameters.
When considering UDP, as in this work, similar considerations must be made in relation to the application layer, e.g. when implementing retransmission schemes.

2.2.4 Application layer - Application optimization

Optimization variables: **Video quality and Communication overhead**.
The potential of cross-layer optimization schemes becomes very clear when considering the upper layers of the protocol stack. This is where end-user services reside and essentially also where the final requirements exist toward performance and resilience.
Much work has been conducted within optimization of video streams in wireless environments. In the work of [van der Schaar et al., 2003] a CL approach is used to provide robust video streams in IEEE 802.11 WLANs. Further in [Haratcherev et al., 2006] a synchronized rate adaptation (layer 1) and video codec (layer 7) adjustment approach is applied with good results. Another example is the work of [Kellerer et al., 2003] where multiple users are streaming video in a wireless environment. While the video streams may be of the same perceived quality they do not necessarily have equal requirements to bandwidth and packet loss at the same time. For instance a video with many changes in the picture over time would require more bandwidth and a lower packet loss than a video with small changes in the picture. The authors take advantage of this fact by estimating immediate video quality and feed this information into a decision entity. The entity uses this information to dynamically assign bandwidth to the mobile nodes in a multi-user transmission environment, which for instance could

be based on Time Division Multiple Access (TDMA)[Tanenbaum, 2003]. This again underlines mechanisms of dynamic adjustment and a perspective where fairness to all nodes in an ad-hoc domain are considered in the optimization process.

Cross-layer optimization in ad-hoc networks has also been mentioned in relation to realize new progress in overlay networks. In brief an overlay network is an application layer logical network topology hiding the underlaying network structure from applications. Overlay networks could potentially become a part of a middleware layer to reduce complexity of developing, deploying and using end-user services in ad-hoc networks. In [Borgia et al., 2005] a cross-layer adaptation of a popular overlay network called Pastry is tested experimentally. It is named CrossROAD and takes advantage of existing proactive routing protocols in layer 3. It reuses continually updated knowledge of actual node topology to construct the overlay network. In addition routing protocol update messages are piggybacked to contain information about available services in the network for service discovery. This represents an example of utilizing existing information and functionality in the protocol stack to minimize overhead and improve performance. A significant decrease in overhead from CrossROAD has been considered in comparison to the original Pastry without degrading performance in terms of reconfiguration delays and packet loss.

Considering the DCAD service similarly a potential layer parameter would be the *zone range* defining how far information needs to be spread. In special cases a reduction in the zone range could be imagined to reduce the overall amount of data in the network, thus improving the probability of QoS requirements being met. Clearly this would be considered from a CL perspective.

2.2.5 Concluding remarks

As shown it is possible to apply CL optimizations to performance and resilience metrics in the entire protocol stack. Worth noticing is that as optimization aims are focused on the upper layers, they become very specific to a considered application, here exemplified by quality of video streaming. Optimizations toward transport and routing are more general in relation to different applications based on IP networks. However, their optimal operation may still be defined by requirements set by different types of applications for e.g. throughput and delay.
As current work establishes, much gain can be achieved even by making simple CL optimizations as for example using new routing metrics in DSR. The useful results achieved in the different work presented underlines the large motivation to consider how performance and resilience can be improved in volatile ad-hoc networks. However significant challenges exist to enable CL optimizations in the protocol stack and maintain its strength of modular design and interoperability of different technologies. Risks are that CL optimizations increase the complexity significantly to maintain and develop the protocol stack when current layer interfaces are breached [Kawadia and Kumar, 2005]. This topic is discussed further in section 2.5.

As a summary of this section Figure 2.3 establishes an overview of the layer parameters that have been identified in existing work. The set of layer param-

eters is related to performance metrics. A dot marking defines that the specific layer parameter has been used in the optimization of the corresponding performance metric in one or more of the studied work cases. I.e. missing markings do not necessarily mean that the considered parameters and variables are completely independent; however their relations have not been considered.
To simplify this figure, performance metrics are presented generally in relation to the application layer with the exception of energy consumption. For instance latency is given by delay in ms of a datagram in the application layer. The identified optimization variables presented in this section are related causally to the application layer. E.g. packet delay leads partially to datagram delay (see Figure 2.2 on page 21). Similar assumptions can be made for throughput.
Jitter has been included with respect to the QoS requirements. In this figure it is represented with direct relation to the latency in cases where latency is bound to vary. Examples are varying retransmission tries or routing over varying paths with varying delay characteristics.
A special optimization variable, *user perceived QoS*, has also been included to capture aspects beyond the protocol stack. In this case toward the user. This is an interesting point that optimizations can also be targeted at user perception.

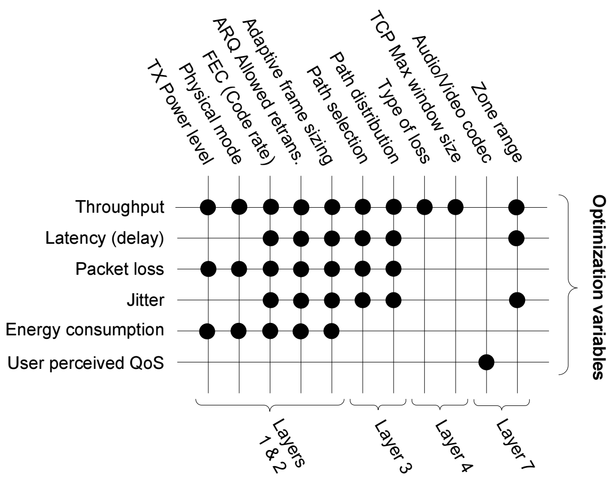

Figure 2.3: *The influence of layer parameters on optimization variables.*

From this figure and the considered work important points can be extracted:

- It is important to notice that layer parameters may impose a tradeoff on optimization variables. Example: as the coding rate of L2 FEC is increased the packet loss decreases at the cost of an increased delay[Sun and Hughes, 2005].
- Several different layer parameters can be adjusted in parallel to ensure the ability to achieve the desired performance and resilience.
- QoS requirements may be given for one or more of the optimization variables.
- Cross-layer optimization relates to both observations and optimization variables across the layers.

- Most current work focuses on performance optimization and improvements and not directly on reliability.

The presented layer parameter and optimization variable sets in Figure 2.3 are by no means complete. Depending on other protocols, work not considered here and future ideas this list could be extended significantly. Further, the task remains to introduce which parameters and optimization variables are considered in this work. This is done in Section 3.2.

2.3 Challenges in Cross-Layer resilience optimization

In this section important Cross-Layer optimization challenges, that have been identified in the current work, are introduced and discussed.

Static and dynamic optimization
A CL optimization can be realized as either static or dynamic optimization. Static optimization is a matter of tuning layer parameters at design time for a given protocol stack and/or application, in order to obtain an overall optimization given expected conditions. An example of a static optimization could be a joint optimization of ARQ timeout values at link and application layer (UDP) to increase throughput given a common wireless scenario. However, conditions may be varying heavily at run-time thus motivating dynamic optimization where layer parameters are adjusted continually to adapt the protocol mechanisms accordingly. Enabling this adaptation involves functionality to make a qualified decision of the optimal layer parameters as discussed previously. This is further considered in the next section. In the context of adaptation there is an important challenge of adapting to rapidly changing channel states. In cases where obtaining observations and the adaptation algorithms are slower than the changes in the channel state dynamic approaches may result in very poor performance [van Der Schaar and Sai, 2005][Goldsmith and Chua, 1997]. Thus the dynamics of the channel conditions impose requirements toward how fast decision techniques must be. In addition, a proactive approach that attempts to predict future channel conditions could prove useful.
It must also be considered that a majority of observations are made directly or indirectly from network traffic. For example Round-Trip Time (RTT) measurements are extracted from transmitted packets and their acknowledgments. In addition ELN messages, as described earlier, are generated by nodes in the network and sent to the node of concern. Due to the nature of wireless networks and shared network resources this inherently means that observations may be missing, delayed, noisy, uncalibrated and/or ambiguous [Steinder and Sethi, 2004] [Nickelsen and Grønbæk, 2006]. Further some observations may be hidden in relation to communication nodes and consequently difficult directly to obtain by measurements. Examples are Signal to Noise Ratio which in [Haratcherev et al., 2006] is estimated from Signal Strength Indication measurements and in [Yuen et al., 2002] where channel contention is estimated by measurements of delay in sending a frame on the wireless link (including retransmission attempts).

Model-based estimation and methods for coping with unreliable and hidden observations are discussed in section 2.4.

CL Optimization conflicts

If several CL optimizations are used, it is important to consider how the optimizations influence each other. The optimizations could potentially be conflicting and counteracting each other. The optimizations should not only be viewed independently but also from a holistic view to understand their interactions. Likewise, setting some layer parameters may improve one optimization variable but decrease the performance of another. As mentioned in the example previously increasing the network layer FEC code rate would result in lower packet loss, but at the same time increase the packet delay.

Precautions should also be taken if several optimizations adjust the same parameter via feedback loops. Here, timescale separation should be used to avoid conflicting adjustments and obtain stability [Kawadia and Kumar, 2005].

Redundant resilience mechanisms

As emphasized earlier a traditional layered view focuses on single-layer optimization. Thus fault resilience mechanisms are also operating individually on each layer. This means that some redundant resilience mechanisms are implemented in different layers. For example, ARQ could be available in both the link layer and transport layer. Clearly link layer ARQ is only responsible for a single hop while transport protocols must maintain the entire end-to-end connection. However, there may be a dependency which may affect performance. For example if the time-out value of TCP is less than the delay at transmitting a frame. This would cause an expensive packet loss in cases where possibly another inexpensive frame retransmission could have done the job. In other cases large differences in link-layer and transport layer timeouts could result in a timely overhead in detection of packet loss. CL optimization could efficiently target such problems.

Node local or global optimization

The scope of possible CL optimizations extends from adjusting parameters within a single node based on locally available observations, to adjusting parameters in and/or obtaining *cross-node* observations from remote nodes, i.e. a global optimization. In order to realize a global optimization, a holistic view of the concerned ad-hoc domain is needed. The global optimization could be directed from a central entity in the ad-hoc domain, e.g. a node that has been elected as a supernode. This node would then coordinate the settings of the ordinary nodes. Contrary to this is a distributed approach, where all nodes are equal and hold a view of the ad-hoc domain and agree on the parameter settings of each node.

Finally, when considering the use of an end-to-end service in the ad-hoc domain, the optimization could be performed in the endpoints only, or also in intermediate nodes.

Having identified the challenges in realizing a CL optimization, the following section 2.4 discusses optimization methods, model-based approaches that can be applied to make decisions in dynamic optimization approaches and means for dealing with faulty observations. Subsequently a discussion of architectural considerations in CL design is made in section 2.5.

2.4 CL optimization approaches

With the overall aim of this work being the development of reliability and performance CL optimization, this section will identify the necessary steps in this process and investigate how such optimization can be achieved. A general view on CL optimization is taken in Figure 2.4.

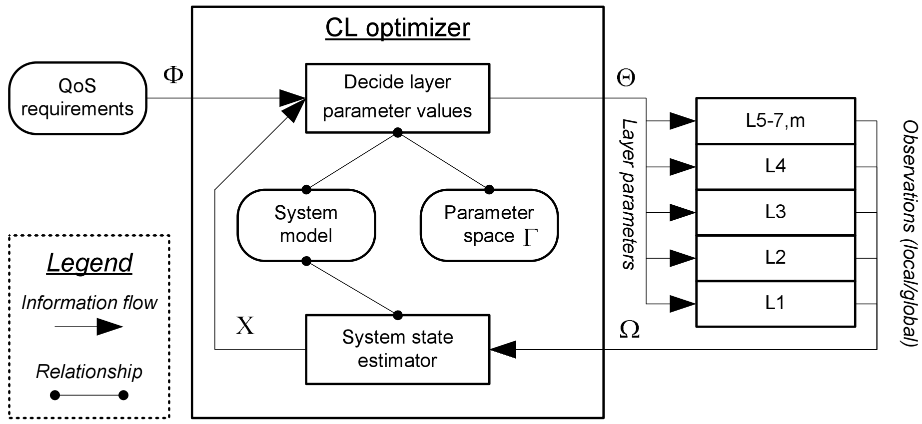

Figure 2.4: *CL optimization decision.*

The central part of this figure is the *CL optimizer* whose job it is to decide the optimal layer parameter values. This decision is based on two variable factors: the *QoS requirements* (Φ) from the end-user service and the state of the system, which is obtained from *observations* (Ω) in the protocol stack.
The system state (X) is given by the variables that affect the behavior of the system, i.e. current layer parameter values, conditions and use of the protocol stack. The set of all possible system states is the state space of the system. However, some regions of the state space may not be directly observable from observations. Furthermore, observations may be unreliable. Therefore the *System state estimator* is used to preprocess the observations appropriately and estimate the actual system state. To be able to make this estimate, the system state estimator needs knowledge about the behavior of the system, through a *system model*.
In order to *decide layer parameter values* that are optimal for the current system state, the CL optimizer also needs a system model and knowledge about the possible layer parameter values, i.e. the *parameter space* (Γ). The CL optimizer chooses the optimal layer parameter values (Θ) from the parameter space, that is $\Theta \in \Gamma$.
The distinction between the system state estimate and the layer parameter value decision is not always clear, since both could be an integrated part of the system model.

The full definition of the used variables is available in Table 2.3.

Layer parameter vectors	$\Theta = \{\theta_1, \theta_2, ...\theta_7\}$
Observation vectors	$\Omega = \{\omega_1, \omega_2, ...\omega_7\}$
QoS requirements vector	$\Phi = \{\phi_1, \phi_2, ...\phi_k\}$
Layer parameters for layer i	$\theta_i = \{\alpha_1^i, \alpha_2^i, ..., \alpha_n^i\}$
Observation variables for layer j	$\omega_j = \{\beta_1^j, \beta_2^j, ..., \beta_m^j\}$
System state	X
Parameter space	Γ

Table 2.3: *Variable definitions.*

2.4.1 Optimization problem

Generally the problem of deciding the optimal layer parameter values can be formulated as an *optimization problem*. Such an optimization problem has the objective of optimizing a function *f*, denoted the *objective function* [Kreyszig, 1999, pp. 990-1008]. Formally, an unconstrained optimization problem can be expressed as in eq. (2.1).

$$\bar{x}_{opt} = \underset{\bar{x} \in X}{\operatorname{argmax}} f(\bar{x}) \quad \text{where} \quad \bar{x} = \{x_1, ..., x_n\} \tag{2.1}$$

The objective function is based on the system model and expresses the concerned optimization variables. Optimization is a matter of either maximizing or minimizing the objective function, depending on the optimization variables. If the objective function expresses a performance aspect of the system it should probably be maximized, whereas an objective function that expresses an error rate should be minimized.
The objective function typically depends on several *control variables* $x_1, ...x_n$, that can be controlled to adjust the behavior of the system. In many cases the choice of control variable values is not completely free but is subject to some constraints, e.g. upper or lower limits.
The optimal solution for the objective function is the choice of control variable values that result in the maximum or minimum value of the objective function.

Determining the optimal solution
Many different approaches exist to determine the set of control variables that maximize the objective function. For a simple system or parts of a system it may be possible to derive a closed form expression that describes the system behavior. In that case the optimal solution can be calculated directly using an analytical expression. Typical examples from networking systems are the $M/M/1$ and $M/G/1$ queuing system models, described in [Cassandras and Lafortune, 1999], where closed form solutions exist for metrics such as mean service time.
In case a closed-form solution is not available, it is necessary to search for the optimal solution. If the objective function and constraints have certain properties, methods that are more efficient than a brute force search, can be applied to determine the optimal solution of parameter values. If for example the objective function is linear and the constraints can be expressed as linear equations, the *simplex method*[Kreyszig, 1999] can be applied to systematically search a very much reduced subset of feasible solutions. In recent years *convex opti-*

mization has gained renewed interest, since methods for obtaining the optimal solution that are nearly as efficient as for linear optimization, have been developed [Boyd and Vandenberghe, 2004]. Convex optimization can be applied to a wider range of problems than linear optimization. Besides linearity and convexity, other properties allow for efficient search for solutions. More information on this topic can be found in e.g. [Rao, 1996].

Multiple objectives

Typical optimization problems often have multiple objectives. This results in more than one objective function to be optimized. Eq. (2.2) shows a formal expression of such a multiobjective optimization problem. Having several objective functions to optimize introduces the need to specify the mutual importance between the objective functions. Further information on methods for optimizing multiple objectives is available in [Andersson, 2000], where the author has produced a survey of such multiobjective optimization methods.

$$\bar{x}_{opt} = \underset{\bar{x}\in X}{\operatorname{argmax}} \begin{bmatrix} f_1(\bar{x}) \\ \ldots \\ f_m(\bar{x}) \end{bmatrix} \quad \text{where} \quad \bar{x} = \{x_1, \ldots, x_n\} \tag{2.2}$$

Relating this general method to the specific topic of optimizing reliability and performance of the system in Figure 2.4 via a CL approach, the overall objective function should consist of objective functions for reliability and performance of the system. The adjustable layer parameters (Θ) are control variables. In addition to the layer parameters, also the system state (X) and QoS requirements (Φ) should be input to the objective function. However, these two variables are not controllable, but are relevant for evaluation of the objective function since they potentially affect the behavior of the system. Eq. (2.3) shows the general objective function to obtain the set of optimal layer parameters.

$$\Theta_{opt} = \underset{\Theta\in\Gamma}{\operatorname{argmax}} f(X, \Phi, \Theta) \tag{2.3}$$

In [Barman et al., 2004] the authors perform a multiobjective optimization, where the objectives are to increase TCP throughput and minimize resource and energy usage. The overall objective function they use is basically the relation $f = \frac{TCP\ throughput}{resource\ +\ energy\ cost}$. The control variables adjust the radio transmit power and ARQ retransmission count and FEC code rate in TCP.

2.4.2 System model

A prerequisite for the system to be able to decide the optimal layer parameters, is a model of how the system behaves. This model is then used to construct an objective function as described in the preceding text. This means that the system model is actually contained in the objective function.

Model simplification

In complex stochastic systems the behavior of the system is often impossible to describe completely accurately. Therefore simplifications are made to allow for less complex system models to be used. The level of complexity depends on the required accuracy, which again depends on the purpose of the model.

Here, the system model should characterize the system or the relevant parts of the system well enough to allow for qualified decisions about the optimality of different parameter combinations to be made.
In addition to making simplifications, the individual layers of layered system architectures such as a typical five layer internet protocol stack can be subdivided into smaller and less complex models. In [van der Schaar et al., 2003] and [Barman et al., 2004] analytical sub-models are combined to achieve expressions of the complete protocol stacks. Many different types of models can be used to capture the behavior of the system. The model type should be chosen on the basis of the properties of the system and on what aspect of the system behavior the model should describe.

Choice of model type and decision method
The choice of model type is closely connected to the choice of decision method as shown in Figure 2.4. A selection of decision methods will be presented shortly in section 2.4.3. Whether one chooses the model type or the decision method first, could depend on many factors. If certain properties are evident to include in the model, the model is chosen first. If low computational complexity is most important, the decision method is of decisive importance, and should influence the choice of model.
As mentioned in the beginning of section 2.4, some regions of the system state space may not be directly observable. Furthermore, observations may be missing, noisy, uncalibrated, ambiguous or unreliable. The chosen system model and decision method should therefore be able to cope with such uncertainties, if relevant.

The following nonexhaustive list presents examples of model types that could be used for modeling the considered system. The list is inspired by and partially based on [Lollini et al., 2005].

Markov chain A Markov chain is a stochastic model that describes the system state at successive times. The model structure determines how state transitions happen in the system. The system can make a transition to another state or stay in the same state. The Markov property means that the current state only depends on the last state. The sojourn time in a given state is exponentially distributed for a continuous time Markov chain and geometrically distributed for a discrete time Markov chain.
Generally, system models that include the wireless channel often include a stochastic part to take the uncertainties caused by interference and mobility into account. For example in [Khan et al., 2006] and in [Lettieri et al., 1999] a two-state Markov chain model is used to represent the radio link layer.

Bayesian Network (BN) This probabilistic model expresses causal relations between variables that describe the state of the system. The network is built as a directed graph where the nodes in the graph represent state variables and vertices specify the causal relations between variables. Each variable has a number of possible states. The model is initialized by specifying system states in terms of á priori conditional probabilities between the states of the system variables. BNs are excellent for modeling systems where unobservable states exist, since not every variable in the model need

to be observable.
With a focus on fault localization, the authors in [Steinder and Sethi, 2004] show that BNs can be used to perform fault localization in complex communication systems while using dynamic, ambiguous, uncertain, or incorrect information about the system structure and state.

Diffential equations Mechanical and flow systems are often modeled using differential equations. These equations describe the input-output relations of the system. Since networking systems have some conceptual similarities with flow systems, this type of model might be useful, however possibly complex for communication systems.

Fuzzy logic If a system has nonlinearites and uncertainties, a fuzzy system description can be used to describe the system behavior since it is defined from observations rather than system structure. Fuzzy logic uses membership functions to map the system input to fuzzy system states.
In [Xia and Liang, 2005] the authors use a 27-state fuzzy model based on 3 antecedents with 3 possible states each, as the basis of a CL optimization of delay, throughput and network lifetime in ad-hoc mobile networks.

Experimental data For a system whose model is difficult to derive analytically, or systems that are too complex to model analytically, experimental data may constitute the model. The data may have been collected through experiments performed on an actual test-bed or from simulations of complex system models. Examples of experimental test beds are available in [Borgia et al., 2005] and [Lundgren et al., 2002]. The downside to this type of model is that it may be very time-consuming or even practically impossible to obtain coverage of all important system states.

2.4.3 Decision methods

For the system to be able to choose optimal layer parameters based on the system model, i.e. determine argmax of the objective function, a decision method is needed. As mentioned in the previous section 2.4.2 the system model type and decision method are often closely related, which is also clear from the listing below.

Online or offline optimization

The choice of values of the layer parameters can be either a static off-line assignment or a dynamic on-line adjustment. The static optimization has the advantage that the time it takes to determine the optimal layer parameter values is not critical. However, off-line optimizations are only suitable when the parameter choice does not depend on dynamics such as the mobility of a node and changing channel conditions. In such cases, a dynamic optimization is preferred to be able to continually adapt to the varying conditions. For the dynamic optimization the computational complexity associated with choosing the parameter values is therefore critical.

Self-learning

Given the dynamic nature of mobile ad-hoc networks, it might be difficult to design a model and decision approach that works optimally in every possible setting. Adaptive or self-learning approaches could be considered as a solution to cope with such issues. A self-learning optimization entails that in addition to layer parameters, also the model parameters are adjusted online to better reflect the actual system behavior. The relevance of a self-learning approach may be considered secondarily.

The following nonexhaustive list presents relevant decision methods. The listed methods should be viewed in connection with the list of model types in section 2.4.2.

BN inference A BN model is specified in terms of causal relations and á priori probabilities. Using this as a starting point, inference in the BN is a matter of using available observations to instantiate the appropriate variables. The outcome is the posterior probability distributions in the residual variables that can be used to estimate the system state. Since multiple observations are used to estimate the system state, the BN may be able to tolerate unreliable observations. Even if observations are missing, the inference approach ensures that a solution is always provided, though probably less accurate. The ability to handle unreliable observations depends on the concerned system model and the variety of available observations.

Markov Decision Process (MDP) MDP is an extension of Markov chains. The difference is the addition of actions and rewards (or cost). The purpose of MDPs is to predict the cost of choosing certain actions over time, which allows for the optimal policy to be found. Further, MDPs are closely related to BNs [Pearl, 2000]. MDPs are suitable for solving the class of problems in BN called planning problems. Hidden system states may be handled using Partially Observable Markov Decision Processes (POMDP). The authors in [Yu and Krishnamurthy, 2005] show that optimal connection admission control problem in CDMA networks can be approximated with an average cost MDP, which further allows for a dramatic reduction in the CPU time required for determining the optimal solution.

Hidden Markov Model (HMM) Like MDP, HMM is also an extension of Markov chains. As the name indicates the HMM is able to cope with hidden or unobservable system states. The HMM can be used in three ways with regards to hidden states as described in [Rabiner, 1989]. The first is to evaluate the probability of a sequence of observations given a specific HMM. The second is to determine the best sequence of model state transitions given a sequence of observations. The last is to adjust the parameters so as to best account for a given a sequence of observations. In order to solve the HMM, the Viterbi and forward-backward algorithms are used. Actually the HMM is a subclass of BNs. The authors in [Smyth, 1997] describe that the Viterbi and forward-backward algorithms are in fact equivalent to the algorithms developed independently for BNs.

State space control A state space controller is designed from a system model based on linear differential equations as presented in [Franklin, 1993]. The differential equations are transformed to obtain the controller structure. The state space controller can be extended with an observer, which is system model that makes controller able to cope with hidden states.

Fuzzy control On the basis of the fuzzy states that the fuzzy logic model defines, the fuzzy controller decides which action to take using a set of rules. Finally the result of the decision is mapped into a control signal, e.g. layer parameter choices.

Convex optimization Uses numerical *interior-point methods* to search for the optimal solution of a problem [Boyd and Vandenberghe, 2004]. The method is a general optimization method as described in section 2.4.1.

Proactivity

As a concluding remark on methods for deciding optimal layer parameters, the relevance of proactivity should be emphasized. In [Feng and Reeves, 2004] a proactive approach to mobile IP handover based on motion prediction is proposed. Both handover latency and packet loss rate are dramatically reduced with the proposed scheme. Seen from a more general perspective it is anticipated that the introduction of proactivity can enhance other aspects than handover. More specifically, the BN and HMM models presented above could be used to forecast likely behavior of the system. This would allow faults such as congestion or link loss to be prevented instead of tolerated.

This section has detailed the possible elements of a CL optimization and presented model types and decision methods. This constitutes a basis for deciding upon a certain optimization method in Chapter 4.

In the following section the architectural considerations concerning CL optimization are discussed.

2.5 Cross-Layer architecture

A CL enabled architecture should allow CL observations and CL parameters adjustments in relation to a dynamic optimization in a traditional protocol stack. This rises the classical issue of good structured and well-arranged design against a heavily optimized design. The latter typically also implies increased complexity and a design that is potentially difficult to maintain. The concepts of the layered OSI reference model form the basis for the most widely available networking systems today with the TCP/IP reference model as the exemplary example. Thus it has in its current form proved highly efficient to amongst other ensure:

- Interoperability across different protocols and vendors
- Maintainability to offer progress and improvements in communication technology
- Possibility of standardized interfaces and functionality to ensure market penetration of communication products

These principles and advantages of a structured design are at the risk at being jeopardized when introducing cross-layer optimizations [Kawadia and Kumar, 2005]. Thus careful considerations should be made when introducing a cross layer design. In this section general approaches from literature are described and discussed in brief. These design approaches are considered when the CL resilience optimizations are to be designed and implemented in this work.

The work in [Srivastava and Motani, 2005] is a survey of CL design and implementation proposals in the literature. These proposals are summarized in the following.

2.5.1 Cross-Layer design approaches

Current CL designs may be categorized into the following four approaches.

Defining new interfaces CL information may be transferred between layers in the protocol stack by extending the existing interfaces. An example of *upward* communication conforms to some types of Explicit Loss Notification (ELN), which involves the transmission of link loss messages to the TCP protocol [Barman et al., 2004]. In the *downward* direction the CL information could consist of QoS requirements, which allows lower layers to adapt. Finally, some designs use interactive *back-and-forth* communication, typically to strengthen cooperation between MAC and PHY layers.
This approach is straight forward but increases the complexity of interfaces between layers decreasing the level of information hiding and simplicity. Further specific interfaces would be required depending on individual optimizations. This could potentially lead to an unbounded amount of new interfaces exploding with different types and versions of protocols and communication technologies. Thus this approach has clear implications.
As a special case CL optimization information could also be piggybacked in PDUs and sent between layers and nodes. E.g. ELN has been implemented as a flag in IP packets [Barman et al., 2004]. Clearly the implementation of TCP must support this notification and optimization. If this is not the case such a flag would just be ignored.

Merging layers To the extend of the investigation in [Srivastava and Motani, 2005] no specific CL design proposals have been found that explicitly creates a *superlayer*. However, they argue that the collaborative design between the PHY and MAC layers tends to blur the boundary between these two adjacent layers. Similar arguments could be applied to the Internet layer (IP Network) and UDP/TCP transport protocols.
Again such approaches increase design and implementation complexity while dependencies could arise between protocol functionalities. E.g. TCP could become dependent of a specific IP implementation.

Design coupling without new interfaces Optimization of one layer with respect to a certain implementation of other layer.
This possibility has already been discussed in the analysis of existing work. Specifically the authors of [Barman et al., 2004] define optimization of mechanisms in the link layer based on implicit model information of the behavior of TCP. This approach avoids the implementation of new interfaces. However if assumptions do not match, i.e. if the used TCP type differs from the one optimization is initially based upon, performance could be degraded instead.

Vertical calibration Basically, the performance seen at the level of the application is a function of the parameters at all the layers below it. Based on

application level metrics and requirements layer parameters in the entire protocol stack can be tuned either statically or dynamically.
As discussed earlier the static approach does not require any new interfaces being added. However, it may not be efficient for different application cases and conditions. On the contrary the dynamic approach enables adjustment for both application requirements and varying conditions at the cost of requiring interfaces to access and use layer parameters as well as observations.

Parameter abstraction
Some generality could be introduced to make CL optimization an optional part of the protocol stack. An approach that is being used in several cross-layer optimization suggestions, e.g. in [Kellerer et al., 2003], [Khan et al., 2006] and described elaborately in [Choi et al., 2006], is to use abstracted layer parameters. The motivation is that layer specific or technology specific parameters may be incomprehensible or of limited use to other layers and the optimizer. Therefore a set of abstracted parameters that describe the generic functionality of each layer is used by the optimizer. For example, in [Khan et al., 2006] the following abstracted parameters are used in the radio link layer: transmission data rate, transmission packet error rate, data packet size and channel coherence time. A concrete implementation of the radio link layer is then responsible for converting between abstracted parameters and implementation specific parameters.
This vendor/technology independent abstraction is useful with respect to the COTS requirement in HIDENETS.

2.5.2 Cross-Layer implementation methods

Another aspect of the CL architecture is how the CL interactions could potentially be implemented. The following three proposals are suggested in literature [Srivastava and Motani, 2005].

Direct communication between layers The layers communication directly via function calls to set and get, parameters and observations.

Shared database In this case the layers do not communicate directly, but are able to access the needed information a shared database.

New abstractions A more drastic approach would be to break away from the traditional layered model and create entirely new abstractions of functional components. For example this could be to organize protocols in heaps instead of layers and allow direct interaction between all components.

These considerations do not entail a single advantagous approach to apply CL optimizations in COTS protocol stacks. As discussed, many challenges exist to ensure that suggested solutions do not reach a level where the benefits of optimizations are eclipsed by the added costs of handling the increased complexity of the network design. In this work the aim will not be to define and develop a general approach for CL resilience optimization. To do this many types of optimizations should be considered covering different layers as defined in Figure 2.3 on page 26. However, the architectural discussion is considered highly relevant and is continued later in this work.

2.6 Problem statement

In this section the important results of the pre-analysis are emphasized to define the main problem of this work and identify the objectives in applying Cross-Layer resilience optimization to a relevant service scenario in an ad-hoc network.

2.6.1 Problem description

Pervasive computing encompasses the possibilities to provide electronic end-user services in all types of scenarios. Common to many scenarios is that a networking infrastructure does not exist while both end-user service providers and user applications are located in mobile nodes. Together these nodes form ad-hoc networks where communication links are wireless and communication topologies are reconfigured continually and autonomously. To realize these possibilities, the most sought approaches are to use existing hardware and software COTS technologies. This reduces development costs and allows for simple integration with existing infrastructure based communication systems. However, basing future wireless ad-hoc networks on existing technology imposes new challenges. Specifically, wireless networks are inherently volatile and changing conditions caused by mobility and varying service demands render service provisioning in such environments fragile. Current networking software architectures and protocols have not been constructed to handle these issues which consequently leads to poor performance.

Introducing a cross-layer perspective

One approach to address the issues of unreliable service provisioning in wireless ad-hoc networks is to improve the resilience and performance of services by introducing a Cross-Layer perspective. In a standard protocol stack optimizations are made in a single layer, however, with a CL perspective optimizations may be made across layer boundaries. Thus, protocols and communication mechanisms can be adapted to conditions in other layers. This could happen either *dynamically* during operation, or *statically* in the design of a given service and protocol stack deployment. In recent studies, CL approaches have proven useful by offering significant performance improvements. However, compromising layer boundaries must be done with great caution. The layered structure of the OSI reference model architecture forms the basis of the most widely available networking systems today. This layered architecture is highly recognized for enabling interoperability between different protocols and between products from different vendors, as well as ensuring maintainability and standardization. These desirable properties could easily diminish with the introduction of more interfaces and dependencies between the layers that entail increasing complexity of the protocol stack. This must be considered when introducing CL designs.
In extension to the architectural discussions suggested, CL optimization approaches must be related to the COTS systems in which they would be implemented. Besides maintaining existing protocol functionality, access must be provided to adjust layer parameters and establish observations across the different layers.

Mission critical end-user services

The CL perspective is considered an important instrument when requirements are to ensure Highly Dependable services in a wireless ad-hoc networking system. The background of this work is the HIDENETS project where HD services are considered in car-to-car scenarios. A high level of mobility combined with complex services, establish significant challenges to ensure QoS requirements. Thus, CL optimization techniques must be studied to increase service performance and resilience.

To allow this study, a specific car-to-car scenario is considered employing the Distributed Car-data Assisted Driving (DCAD) service case. All cars broadcast information with real-time constraints about their own movements to other cars within a geographically bounded area. This allows assisted driving service applications to take control of e.g. brakes and steering in critical situations to avoid accidents. A service failure could in worst case lead to an accident, thus, the DCAD service is mission critical.

The broadcasting behavior of the DCAD is a large challenge in wireless networks. The broadcast car-data must cover multiple hops, meaning that it potentially will be re-transmitted multiple times throughout the ad-hoc network. As all cars act as data sources, this could lead to much contention in the wireless medium. The requirements on services being reliable entail that the broadcast mechanism must also be reliable. Consequently, lost frames could be re-transmitted, however, this would cause further delays and challenge the real-time requirements. In addition, this service is one of many services competing for access to the wireless medium and node resources. Thus, ensuring service continuity is a highly complex problem. The CL perspective, however, enables a coordinated effort between layers and even globally across nodes to maximize the probability of services being dependable. This is done by optimizing the parameter settings of resilience mechanisms in different layers in relation to metrics of performance and reliability and ultimately dependability.

Optimization models

To enable CL optimizations, system models encompassing the communication processes must be constructed. These models should make it possible to define an objective which can be used to identify optimal settings for the layer parameters. Different model types have been considered ranging from simple measurement based models over differential equations to probabilistic models. One or more of these models could be relevant for modeling different parts of the system. For instance, Markov Chains have proven useful to model wireless channel conditions. In addition, link lifetime characteristics have been modeled with Fuzzy Logic, etc. The choice of model relies on which system properties that are central for the chosen approach to optimizing resilience mechanisms.

Adaptable optimizations

In the car-to-car ad-hoc scenarios mobility is a dominant factor. As a result the conditions for wireless communication and service provisioning are continually changing. This results in variabilities of e.g. quality of links, number of hops between nodes and the number of nodes competing for access to the wireless medium. Thus, adaptable CL schemes are sought which can optimize operation for performance and resilience to ensure the required level of QoS and service reliability. In this relation system models also play a central role. To cope with many different conditions the models can be used to continually calculate optimal settings of the layer parameters. This is denominated the *decision process* which may be considered as an extension to the existing system models. The significant challenge in establishing the decision process is that it must be capable of finding optimal solutions accurately while also being computationally feasible. As input to the decision process observations are collected from network traffic, meaning that they are potentially missing, delayed, noisy, uncalibrated and/or ambiguous. These properties additionally complicate the decision process, which besides being accurate must promptly adapt to potentially rapidly changing channel conditions without lagging behind or degrading performance.

2.6.2 Main problem

As an approach to consider Cross-Layer resilience optimization in the HIDENETS context the specific DCAD service case is analyzed further and solutions are developed to address the problems described in the problem description. Based on this defined problem domain the following main problem is given:

How can cross-layer resilience optimization be applied to improve node-global reliability of geographically bounded and mission critical broadcasting in a car-to-car ad-hoc domain?

This problem is considered from the DCAD service where specific requirements for throughput, delay and message loss define when the service is reliable.

2.6.3 Objectives

Definition of optimization problem(s)

- **Existing broadcast in ad-hoc networks**
  Specify current methods to enable broadcasting in ad-hoc networks. This must establish which reliability is currently achievable and specifically which problems exist ensuring reliability. A delimited fault model is specified for use in the remainder of this work.
  The existing methods must also provide a foundation for evaluating potential improvements of the suggested CL approach.
- **Identify resilience mechanisms**
  Identify which resilience mechanisms have an influence on performance and reliability in the considered end-user service. The resilience mechanisms may either be present in the considered protocol stack or other resilience mechanisms may be introduced to enable the desired resilience functionality.
- **Choose layer parameters and define resilience metrics**
  Primarily based on relevant resilience mechanisms considered layer parameters are chosen and specified. Resilience and reliability metrics are defined for optimization and qualitative evaluation.
- **Optimization problems**
  Define one or more optimization problems to consider based on the delimited fault model and identified optimization variables.

Development of CL optimization methods

- **Define system models**
  One or more systems models are developed to describe relations between layer parameters, optimization variables and observations. Consider that unobservable system states should be identified. The system model(s) must form a basis to construct objective function(s) for the optimization problems.
- **Dynamic optimization method**
  The starting point for development of the system models is to deliver dynamic adjustment of layer parameters under varying operational conditions to continually ensure optimal reliability.
  The aim is to consider global optimization where the overall reliability is improved for all nodes in an ad-hoc domain.
  The applied optimization methods must be evaluated to consider the amount of processing resources required to perform the dynamic optimization.
- **Architecture design**
  An architecture is defined for the proposed cross-layer optimization method. In relation to a COTS system it must be considered how observations can be collected and discussed how access to layer parameters and resilience mechanisms can be achieved. This topic must be related to the HIDENETS context for a general discussion on which middle-layer functionalities should be available.

Evaluation

- **Model implementation**
  The model used for the optimization approach must be implemented to allow a simulation-based evaluation of its capabilities to derive optimal layer parameter settings for varying conditions.
- **Simulation model**
  A simulation model representing the case application environment is developed and implemented in ns-2. It should include conditions imposed by wireless aspects.
- **Evaluation approach**
  Test cases for the application case are developed and implemented. Simulations are executed with the normal and the optimized protocol stacks to allow for a comparative evaluation of the proposed optimizations.

2.6.4 Delimitations

The following delimitations are made in this work:

Environment of operation

- **Urban/rural environments**
  Only a single stretch of road in a rural environment is considered. Thus, more advanced influences on wireless traffic from obstacles in

urban environments and multiple crossing lanes are considered a part of future work.

- **Mobility models**
  To simplify the extend of the mobility models required to study conditions from mobility the following delimitations are made:
  - **The stretch of road is straight, and only two lanes going in the same direction are considered.**
  - **No abnormalities in the traffic flow are considered, i.e. no accidents and no evasive maneuvering.**

Faults and conditions

- **Software and hardware faults**
  Software and hardware faults are not considered. I.e. nodes do not disappear/crash as a result hereof.
- **External sources of noise**
  Other electronic devices like microwave ovens and communication systems like Bluetooth create noise that interferes with existing wireless networks. This can degrade performance. However, in this work only interference from other nodes is considered and no additional noise sources are introduced in the environment.
- **Malicious faults**
  Besides faults generated from conditions of the environment, congestion and contention malicious faults could also be introduced by systems and individuals who try to sabotage service continuity. Being highly relevant these aspects are considered as future extensions to this work.

Aims of optimization

- **Energy**
  Each car is equipped with a generator. This makes power consumption a minor issue primarily concerned on improving the mileage. Consequently energy consumption is not considered an important optimization metric in this work.
- **End-to-end transport**
  According to scenario description do not consider end-to-end transport for optimization. However, cross-traffic in simulation environment may be.

2.6.5 Prerequisites

Wireless network 802.11b The IEEE 802.11b wireless networking standard is considered for communication in the ad-hoc domain. This standard well known throughput the research community enabling analytical knowledge about its behavior and while being available in simulation environments. Future work should consider potentially 802.11n based on hi-speed MIMO technology or 801.11p which is aimed at communication in a vehicular environment. Both of these standards are not completed. 802.16 (WiMAX) is also a relevant candidate.

Node resources Each car is equipped with a similar computer system containing a modern x86 processor, sufficient memory and solid state storage.

Addressing All nodes in an ad-hoc domain have unique IP addresses.

Chapter 3

Resilience analysis

Broadcasting in wireless ad-hoc networks poses significant challenges due to the unreliable wireless medium, a significant risk of contention and mobility which continually changes the operating conditions. In section 3.1 reliable broadcasting is analyzed to identify its functional components, properties and threats which can breach its reliability. This also comprises a specification of the actual faults which can affect the operation of the broadcast processes. The outcome hereof is subsequently used in section 3.2 to define which resilience mechanisms could be applied in the protocol stack to enable the required QoS. Based on the resilience mechanisms potential CL resilience optimization problems are defined to be addressed in this and future work.

3.1 Ad-hoc broadcasting and fault models

In relation to the introduced DCAD service, keywords to investigate in relation to broadcasting in Mobile Ad-hoc Networks (MANETs) are *reliability* and *efficiency*. These terms define two primary classes of challenges to consider. Reliability requires that means of fault tolerance are introduced to increase the probability of messages reaching their destination. Efficient broadcasting aims at minimizing resources consumed by broadcasts to also enable other services to co-exist. In addition, efficiency may become a prerequisite for reliable broadcasting, as insufficient resources increase the risk of faults in the wireless ad-hoc network. Both aspects of broadcasting in MANETs are considered in this section.

Currently broadcasting in ad-hoc networks is being actively researched. Thus, in the literature many approaches are suggested and compared performance wise. Taking a starting point in those, substantial problems and possible solutions targeting reliable and efficient broadcasting are identified. Initially, assumptions for the DCAD service are defined to specify how it uses the broadcast mechanisms and which issues and possibilities arise as a consequence hereof.

3.1.1 Assumptions of the DCAD service

As defined in the service description in section 2.1 on page 13 the DCAD service is based on layer 3 broadcasting. In addition these assumptions are made:

Multipoint-to-multipoint communication - Each node broadcasts car-data to all other nodes within a specified geographical area defined by a zone around each car.

Multi-hop broadcast - The source node of a message may not have all destination nodes within transmission range. Consequently, messages needs to be relayed by retransmissions in multiple hops to cover the zone.

Geocasting, limited broadcast - Since the broadcasting of car-data is geographically limited, the broadcast mechanism belongs to the category of *geocasting* routing protocols where nodes in a particular area are addressed [Maihöfer, 2004]. In this case the routing problems are limited to determining which nodes are within the geographical area that the broadcast zone spans, and thus, when a message should not be relayed further. The geographical area is here defined relatively to the source node, so that the node is always located in the center of the area.

Periodic updates - Each node broadcasts car-data with a fixed interval T corresponding to the transmission frequency as introduced earlier. T is the same for all nodes in considered service scenarios.

Addressing - Each node is assigned a layer 3 IP address when entering an ad-hoc network. Basically the client is assigned an address by the network or it makes one up and asks the network if it can join with this address. Separate Ad-hoc network domains may also merge [Mohsin and Prakash, 2002]. In this work it is assumed that all nodes have been assigned an individual layer 3 and clearly also layer 2 address in all considered service scenarios.

Reliable broadcast - The notion of *reliable broadcast* is used in this work. However it must be emphasized that a reliable service in this case is not defined as complete absence of packet loss, but instead with a small packet loss as specified in Table 2.2 on page 18.

3.1.2 Basic problems in broadcasting in MANETs

Numerous problems must be addressed to enable reliable and efficient broadcasting in MANETs. In the following some of these major problems are described. Solutions to overcome these problem will be considered later when designing the CL optimization approach.

Hidden and exposed nodes

Hidden and *exposed* nodes constitute two basic problems in wireless communication. In an 802.11 based ad-hoc network the nodes would typically be configured to use Distributed Coordination Function (DCF), meaning that they manage Media Access Control (MAC) in a distributed manner with no central coordination. This enables the following scenarios.
When a node 'A' wants to transmit to node 'B' it senses if the channel is clear (CSMA mechanism) and starts transmitting if clear. 'C' is within communication range of 'B' to which it wishes to transmit, but out of range of 'A'. 'C' does not sense that 'A' is occupying the medium, thus 'C' starts transmitting. Now 'B' receives transmissions from both 'A' and 'C', meaning that the transmissions

result in collision. This is the *hidden node* problem, whereas the *exposed node* problem deals with the reverse situation. In this case 'B' wants to transmit to 'A'. However, 'C' is now sensing the channel is occupied. As a result it cannot start communication with node 'D' which is outside the range of 'A', and unintendedly communication is blocked. More details on these issues can be found in [Tanenbaum, 2003].
In the broadcasting scenarios these effects can have a great impact on efficiency. Hidden nodes may start communicating causing collisions and need for retransmission. Exposed nodes would also be blocked and potentially hinder the multi-hop relaying of messages, increasing the overall message propagation delay.

Medium access
Elaborating on the issues of hidden and exposed nodes an essential issue of efficient broadcasting is to make the best possible use of wireless resources. This means that multiple nodes in best case should not try to access the medium at the same time causing costly frame losses and back-off time. As the MAC is decentralized this is a difficult problem to solve. 802.11b uses an RTS/CTS scheme where nodes ask each other for permission to transmit, however this may be inefficient in terms of overhead and exposed nodes.

Layer 2 and 3 broadcasting
In IP/Ethernet based infrastructure networks broadcasting capabilities exist in both the link and network layers. Link layer broadcasts address nodes in the physical network targeting a broadcast MAC address. In the network layer broadcasts address the logical network to reach a specific subnet. These broadcasts may be routed. In the destination network the link layer is responsible for targeting all nodes with the broadcasted message. The same relations between these two layers could be realized in MANETs, however notions of subnets may not necessarily exist. In the considered application case broadcasting could be managed in either one or both of the layers. From a general perspective this work considers the latter case where layers 2 and 3 have different roles in the broadcasting process.
Layer 3: This layer handles the logical addressing of nodes and routing of messages. This translates to handle mechanisms of the geographically bounded broadcast and relay messages within this area.
Layer 2: Management of wireless resources is an important task to ensure efficient broadcasts as exemplified by the hidden and exposed node problems. Thus MAC is required to optimize the usage of the wireless medium.

Redundancy
When addressing problems of reliability and limited transmission ranges, new problems arise. The problems can be considered from an example of a basic broadcasting scheme. The scheme offers reliable broadcast by introducing layer 2 ARQ where destination nodes reply with positive acknowledgments to the source of broadcasts. If one or more destination nodes fail to receive a message it is retransmitted. This broadcasting scheme is based on flooding to relay messages. I.e. nodes who have received a message, broadcast it once again to potentially target nodes not covered by the initial broadcast. Besides other

obvious problems considered shortly, this causes a large amount of redundancy in the ad-hoc domain. Nodes who have received a message successfully will receive the same message multiple times causing inefficient usage of node and networking resources. Thus redundancy should be minimized. However, it must be considered that redundancy clearly also could be intendedly used to correct faults. A possibility is in case of malicious faults. Intendedly altered messages from one node could be identified and dropped given the same unaltered message from multiple other nodes would indicate this. However, this aspect is delimited in this work and not considered further.

Flooding
Considering again the basic broadcasting scheme, as introduced in the previous section, the flooding features do ensure that a message can be forwarded in multiple hops to cover the entire ad-hoc domain. However, this could be very inefficient. There is no guarantee that an intermediate node, relaying a message, reaches new nodes. Also, as previously mentioned, redundancy could be high. Clearly, nodes would also need to register whether a received message has been relayed earlier to avoid loops and subsequently instability due to a *broadcast storm*.

Acknowledgment implosion
An approach to achieve reliable broadcast is to introduce ARQ schemes. The most forward approach is for all nodes to acknowledge a message to the source of a broadcast. This causes more messages in the wireless channel which will become increasingly contended. As messages are lost, even more messages and acknowledgments are transmitted which again increases contention and the probability of messages being lost. In worst case scenarios this acknowledgment implosion can clog the wireless channels entirely. Especially around the source node this is a problem. Multiple nodes would send it a positive acknowledgment all at approximately the same time; unless measures have been made.

3.1.3 Broadcast challenges

From the introduced problems it is possible to specify specific challenges in MANET broadcasting that must be faced to ensure reliable and efficient broadcasting. These challenges are targeted at the reliable broadcasting methods employed which also encompasses the resilience mechanisms and introduced CL optimizations.

Selection of forwarding nodes
When reducing redundancy the goal is to increase the efficiency of broadcasting. Improvements over flooding can be introduced where only a limited amount of nodes are allowed to forward messages from the initiator of the broadcast. The challenge is to find and select the fewest amount of nodes to forward a message while ensuring coverage of all nodes. This can be defined as the problem of finding the minimum Connected Dominating Set (CDS). The ad-hoc network can be considered as a graph $G = (V, E)$ where V is a set of vertices and E a set of edges. This is depicted in Figure 3.1 on the next page. A dominating set of G is a subset V' which describes a set of vertices to which all nodes not in V' are

connected by at least one vertex. In a CDS the vertices in V' are all connected. Finding the minimum CDS would lead to a solution where the fewest amount of forwarding nodes needed to reach all nodes are found. However, finding this solution is NP-complete and approximations must be made instead [Liu et al., 2007]. Solving and maintaining a CDS is an expensive operation in ad-hoc networks. The instance, at which the CDS is created, must have a complete picture of the topology while ensuring that all nodes are aware of their role as either a forwarding node or passive node adjacent to a forwarding node. This requires much control information to be propagated in the network which degrades the advantage of finding a near optimal MCDS [Mastrogiovanni et al., 2006]. This is especially true in under mobile conditions where topology continually changes. Much recent work is made in finding a small CDS based on distributed decisions in the individual nodes and minimum requirements for topology knowledge (see [Mastrogiovanni et al., 2006][Williams and Camp, 2002]). It should be emphasized that minimized CDS is not unconditionally advantageous. As mentioned earlier redundant messages may be utilized to improve reliability. I.e. in [Lou and Wu, 2003] the authors create a CDS where each adjacent node ideally has two forwarding neighbors. Also, a CDS may be formed considering link quality between adjacent nodes and forwarding nodes.

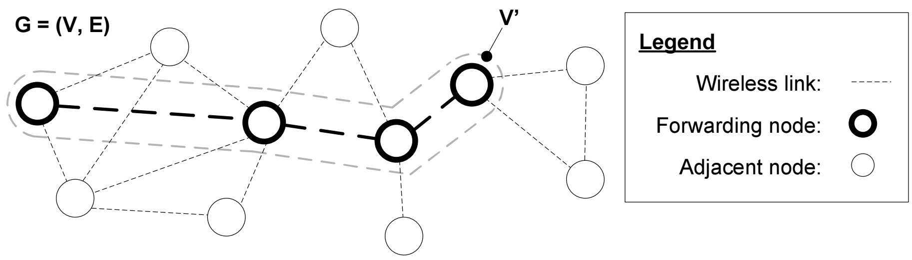

Figure 3.1: *The definition of an ad-hoc network as a graph of forwarding and adjacent nodes in a CDS.*

Minimize and manage link usage

Another common challenge in MANET broadcasting is to decrease the usage of the wireless link resources and ensure that proper MAC is applied.

Already by introducing limited forwarding many redundant transmissions can be avoided. In addition focus is on minimizing the amount and size of dedicated control messages, i.e. "hello" messages and possible control information piggybacked in data messages. In reliable broadcasting acknowledgment reduction to avoid acknowledgment implosion also becomes a highly relevant topic. A basic improvement of positive ACKs is to make them explicit where the sender in the data message explicitly states who should respond. In this manner only nodes who failed to get the first transmission will need to respond to retransmissions [Xie et al., 2005]. An alternative to positive ACKs is to introduce negative ACKs where receivers keep track of sequence numbers and request missing messages to be retransmitted [Lou and Wu, 2003]. To work properly this approach requires that messages are sent continually and in all cases a receiver cannot verify a message loss until the next successful reception of a message. The result is an increase in the delay. Other approaches only rely on ARQ between forward-

ing nodes, e.g. as in [Lou and Wu, 2003] where adjacent nodes are guaranteed to receive at least two broadcasts of the same message. Acknowledgments could also be implicit where a source node overhears transmissions from forwarding nodes to establish that they have forwarded it successfully.
The selection of a proper ARQ scheme is clearly a balance between added reliability and costs of link usage and delay.

The shared wireless medium with issues of hidden and exposed nodes provides a clear motivation for management of medium access to optimize utilization and minimize colliding transmissions. Normal uni-cast transmissions e.g. in 802.11b use the RTS/CTS scheme to ensure that nodes in a wireless domain all know when other nodes have been given permission to transmit. These properties have also been applied to broadcasting, where surrounding nodes are silenced by an RTS/CTS exchange with a random neighboring node before a broadcast. Unfortunately only the area around the source and selected neighboring node is secured as the hidden and exposed node problems could arise at other neighboring nodes. Approaches have also been demonstrated to exchange RTS/CTS with multiple neighbors at a significant cost of efficiency to improve reliability [Xie et al., 2005].
Besides aiming at reducing ACKs some ARQ schemes also try to accommodate that multiple ACKs may be sent at the same time to the source node. As a simple example in [Chen and Huang, 2004] response messages are each assigned an order to enable a coordinated acknowledgment process. In other approaches neighboring nodes are scheduled in a Round-Robin manner to control the acknowledgment process [Xie et al., 2005].
A good MAC approach could be important to achieve the required reliability. However, this must be carefully considered in relation to the loss of efficiency. In some cases, especially under low contention, the best approach could be to only use basic channel sensing to avoid collisions.

Increasing probability of successful message delivery

To meet reliability requirements the broadcasting approach must aim at keeping message delivery high to all destination modes. Layer 2 ARQ has already been introduced. In addition FEC schemes could beneficially be introduced in layer 2 to minimize ARQ usage and consequently reduce delay at the cost of a larger frame size and increased transmission latency.
Another interesting approach is to optimize transmission power and consequently transmission range. The idea is to reduce the area of interference from a node transmission and consequently reduce network contention, collisions and noise which can lead to message losses. Possibly more hops may be required to spread messages to the entire zone. However the advantage could in all cases be a reduction of retransmissions and the resulting overhead.

Decrease message end-2-end delay

In particular the DCAD service has strict requirements for delays (and jitter) in the propagation of messages. These requirements may be inherently difficult to comply to due to expected high contention; especially when considering the multipoint-to-multipoint communication and other services operating in parallel. In addition resilience mechanisms like ARQ and FEC increase message delays. This presumably makes low *message delay* a particularly interesting challenge in this work.

3.1.4 Existing MANET broadcasting schemes

In the preceding sections problems and specific challenges in MANET broadcasting have been introduced. Much work with different approaches has been conducted targeting one or more of the presented challenges. It would be too comprehensive to introduce them all in this section. Interested readers are referred to [Mastrogiovanni et al., 2006][Williams and Camp, 2002]. Instead, two existing solutions are described which are used as a starting point for the CL resilience optimization approach. They consist of a *simple* and an *enhanced* broadcast scheme. The simple scheme defines a baseline for CL optimization which can gradually be made more complex to encompass current solutions given by the enhanced scheme. This enables a consideration of and discussion on how the different schemes may be optimized. Both the simple and enhanced broadcasting schemes can be extended to include the geographically bounded broadcasting zone.

Simple broadcast - a flooding scheme
While being highly inefficient flooding in some cases provide good resilience due to the high level of redundant information. Typically flooding schemes are simple to implement and define few functional elements to avoid a broadcast storm. In this scheme each node forwards the first copy of a received message by issuing another broadcast. This continues until all reachable nodes have received and forwarded the message [Williams and Camp, 2002].
This scheme is initially considered without ARQ functionality.

Enhanced broadcast - a CDS based scheme
Among the existing schemes choosing a clear favorite is difficult. New solutions and their comparisons to existing are often made with varying focus from minimizing node forwarding count to increase message delivery ratio in stationary or highly mobile environments. In this work the aim has been to identify a tested approach which is basic to implement and deploy while offering comparative performance.
This complies to the Ad-Hoc broadcast Protocol (AHBP) as described in [Williams and Camp, 2002]. AHBP aims at creating a minimum CDS based on a distributed approach where nodes exchange information to create 2-hop topology knowledge. The resulting broadcast scheme can be categorized as a *non-localized neighbor knowledge* scheme. *Non-localized* refers to that forwarding nodes are selected by neighboring nodes as opposed to *localized* where each node makes its own decision whether to forward a message. In [Williams and Camp, 2002] AHBP shows overall the best results in comparison with other neighbor knowledge, area based and probabilistic schemes. In the tested cases with high density AHBP manages to approximate the MCDS within 10%. Similarly, in more recent work the AHBP scheme shows its competitiveness [Jun et al., 2006]. The most predominant weakness of AHBP is its requirement to continually maintain 2-hop topology knowledge. This can affect its performance in high mobility environments.

Further details on how AHBP constructs the CDS and performs in general may be found in [Williams and Camp, 2002] and [Peng and Lu, 2002].

MAC and ARQ functionality

A good MAC mechanism can improve the reliability of broadcasting by minimizing problems of collisions, hidden and exposed nodes [Xie et al., 2005], typically at the cost of overhead for CTS/RTS like methods. The performance of the MAC method is dependent on the broadcasting scheme. Thus it should be considered in relation to the applied scheme. As the aim of this work is to consider CL optimizations little effort is spend on selecting and analyzing a broad variety of MAC methods. Instead a simple one is applied in this work. It is assumed that the CSMA/CA of 802.11 is present and consequently that a node can sense if the medium is idle. Thus the largest problem are collisions due to hidden nodes and nodes transmitting at approximately at the same time.

To avoid collisions CSMA/CA uses a backoff algorithm. I.e. when a node wants to transmit it senses the state of the channel. If the channel changes state from *busy* to *idle* it draws a random backoff time between 0 and CW_{min} (Minimum contention window) and starts waiting for the channel to be available for a total period of the backoff time. After the backoff time has expired it transmits. Now, if a collision is detected the transmitting node increases the contention window, draws another random backoff time and waits another round before attempting to retransmit. However in basic ad-hoc mode no ARQ is present and collisions cannot be detected. As a result only the minimum contention window is relevant to consider.

Collisions where nodes transmit at the same time are likely to occur specifically in flooding where multiple nodes successfully receive a message at time t_0 and schedule to forward it all at approximately the same time t_1. If the medium is *idle* at t_1 the forwarding nodes are according to the 802.11 standard not required to initiate a random backoff to avoid collisions (see section 9.2.5.1 in [IEEE, 1999] and [Schmidt-Eisenlohr et al., 2006]). To minimize this problem a synthetic jitter can be added to the scheduling of forwarding messages. In reality a contribution to such jitter may exist due to varying processing delay in the different nodes. However, this may not be sufficient to avoid collisions. Thus, a sample t_{jitter} is drawn from a random variable $\mathbf{X}$ which is uniformly distributed in the interval from 0 to Tfw_{max}. This can significantly reduce collisions and improve particularly flooding schemes [Williams and Camp, 2002][Peng and Lu, 2000]. Improved schemes may be considered in future work.

Adding link and/or application layer ARQ may be crucial to achieve the desired reliability. In future studies different ARQ methods may easily be added to the broadcasting scheme. Briefly, optional ARQ methods are described subsequently:

Explicit acknowledgments - This basic ARQ method is sender initiated and requires an acknowledgment from all nodes. The sender of a transmission defines in the data message which nodes should reply. Thus on retransmissions only nodes to which transmission failed will need to respond. If receiving nodes have not responded within some time-interval a retransmission is initiated.

Reduced acknowledgments - This method reduces overhead from acknowledgments by only requiring a subset of receiving nodes to reply. E.g. in the broadcast scheme of [Lou and Wu, 2003] only forwarding nodes need to reply. This scheme is practically an extension to the explicit acknowledgment scheme where only a subset of nodes are explicitly ordered to acknowledge messages.

Implicit acknowledgments - Assuming that nodes in a CDS are connected by bidirectional links using implicit acknowledgments a source node may take advantage of forwarded transmissions from neighboring nodes. Thus as the source node overhears a retransmission it is interpreted as an acknowledgment.

The assumption has been made that the considered broadcasting mechanism is a layer 3 service. Further it has been discussed that efficient broadcasting also includes some involvement of layer 2 mechanisms for Media Access Control. In addition it is specified that broadcasting is geographically limited which requires node location information. In the car-2-car environment such information is readily available from a middleware positioning service. However, this raises a discussion with two perspectives: (1) Which routing metrics are expected to be available in layer 3 in future ad-hoc network protocol implementations. Location information may not be one of them. (2) Whether or not reliable broadcasting/geocasting is a service that inherently belongs to the middleware layer where amongst other location information is easily available. These aspects are still open issues in the research communities; including HIDENETS. To limit the extend of this discussion in this work, the assumption of the broadcasting mechanism as a layer 3 service is maintained. Thus, it is considered a service that may coexist with unicast routing mechanisms like DSR that relies on an efficient (and reliable) broadcast service. The availability of node location information as a routing metric is assumed.

The further studies and CL optimizations considered in relation to the broadcasting scheme used in this work are presented in section 4.2 while the implementation for evaluation and comparison is described in Chapter 5.

3.1.5 Faults in broadcasting

In section 2.1 on page 13 an overall fault model is defined. It describes which faults may lead to a failure of the DCAD service. In this section a subset of these faults is extracted to define a limited, however, essential set of faults. They will contribute to clarify in the subsequent section which resilience mechanisms can be considered across the layers to maximize the probability of a successful service in ad-hoc broadcasting.

The predominant faults in the considered ad-hoc scenario is as emphasized a result of the conditions from *mobility* and the *wireless environment*. In addition multiple nodes are *contending* over the available resources providing an additional challenge to ensure service continuity. The following analysis and resilience optimization goals are specifically aimed at these basic problems. From Figure 2.2 on page 21 the mentioned conditions have an impact on the three basic faults: frame loss, frame delay and link loss. These faults all contribute

significantly through error propagation to the causation of faults in the application layer. Thus strengthening resilience against these faults is expected to improve reliability of broadcasting.
Congestion may also cause message loss in cases where packets to be forwarded and packets generated in the application layer exceed the transmission rate of the outgoing traffic from a node. Both the rate of forwarded packets and the transmission is dependent on the amount of contention. Thus congestion is heavily related to contention in the considered wireless scenario, however, the relation may not be trivial. The primary focus in this work is on preventing the faults caused by contention where secondarily the resulting congestion effects are monitored.

As emphasized previously reliability is evaluated from the fulfillment of the QOS requirements for the end-user service, which in this case is represented by the DCAD end-user service. Requirements are set for the message latency and acceptable message loss as well as jitter and distribution of message losses. In this work the primary focus is latency and message loss where control of jitter and the message loss distribution is considered a challenge in future work.

Faults	**Conditions**	**Fault categories**
Bit error $\rightarrow$ Frame loss and Frame delay	Fading, Shadowing	Channel
Collisions $\rightarrow$ Frame loss and Frame delay	Contention from cross traffic (DCAD)	Medium access
Link loss	Mobility	Routing

Table 3.1: *Faults in broadcasting.*

Based on these considerations an overview of the faults and conditions included in the remainder of this work are defined in Table 3.1. The faults and conditions have been sorted in three fault categories that relate to in which layer these faults are handled. Specific definitions and models of these faults are introduced in Chapter 4 in relation to the modeling approach considered.
In the next section resilience mechanisms are identified which are expected to be readily available in the COTS service protocol stack as introduced in section 2.1 on page 13.

3.2 Resilience strategies and mechanisms

From the knowledge of which faults may jeopardize service continuity it is finally possible to discuss which resilience mechanisms could be applied in relation to efficient and reliable broadcasting. As defined previously a *resilience mechanism* offers the means to protect the end-user service from faults. Either by reducing their probability of occurrence or handling them properly when they occur. In this section the notion of a resilience mechanism is considered broadly. It both includes direct error correcting mechanisms like FEC and indirect mechanisms like transmission power control where varying power levels may lead to different amounts of faults.

The final outcome of this section is to specify key resilience mechanisms to consider for optimization and consequently which layer parameters they define. In addition resilience metrics are specified for the given resilience mechanisms.

3.2.1 Resilience in the physical layer

Achieving resilient wireless channels is essentially a question of minimizing transmission bit errors and/or achieving a connection between a given pair of nodes. The basic resilience metric for the physical layer is *BER*.

Rate adaptation
As initially described in section 2.2 on page 20 presenting *existing work in cross-layer optimization* this may be done by choosing a suitable 802.11 *physical mode* and adjusting the transmission power level. In the considered 802.11b standard the physical mode defines four different rates 1, 2, 5.5 and 11Mbit using three different modulation schemes. 5.5 and 11Mbit use the same modulation scheme but different bit/symbol ratios. As mentioned previously different schemes allows an adjustment to the given SNR measured at a receiver. As SNR decreases, due to effects of e.g. path loss and multipath fading, switching to a lower rate decreases the BER.

Transmission power control
Transmission power control also offers a mean to improve resilience in the channel. In a simple path loss environment increasing the transmission power also increases the SNR in relation to a specific receiver node. Further the transmission power is directly related to which nodes can be reached and resultingly which links exist. This may be essential to ensure connectivity in a given ad-hoc domain.

Channel control
A final physical layer resilience mechanism suggested in this work is channel control. The 802.11 standard [IEEE, 1999] defines 14 different communication channels defined by different frequency bands. Several channels allow multiple wireless networks to co-exist. However as the channel frequencies overlap efficiently only three channels operate without potentially interfering each other. Channel control may be applied in the ad-hoc domain to separate communication and collision domains. E.g. cars driving in opposite directed lanes may be using different channels when not communicating with each other.

3.2.2 Resilience in the link layer

In the link layer the primary concern is to correct faults that have occurred in the channel and to some extent try to prevent them from happening i.e. by managing medium access to prevent collisions.

FEC
FEC introduces redundant information in the transmission to enable a correction of bit errors in the channel and increase the probability of a successful frame reception. The 802.11b standard inherently does not include FEC coding

like for instance 802.11a does. Presumably the lack of FEC in 802.11b is more a cost related than technical consideration made at the time the 802.11b standard was launched as a cost-efficient wireless solution. However, this limitation does not prevent the consideration of FEC as a potential resilience mechanism in the considered protocol stack. The layer parameter of FEC is as mentioned earlier the code rate describing the relation between payload data and redundant bits.

ARQ

ARQ methods in relation to broadcasting have been discussed thoroughly in the previous sections. Different schemes will have different properties in relation to induced delay, overhead and reliability. Thus a layer parameter could be which ARQ mechanism is actually active based on the end-user service requirements and the operating conditions. Another layer parameter would be how many attempts the ARQ mechanism would make to transmit a frame successfully before leaving fault correction to upper layers.

Frame size control

Frame size control enables the link layer to adapt the sizes of frames to the BER probabilities experienced in the wireless medium. At a high BER smaller frames have a higher probability of successful reception at the cost of increased overhead for headers. The efficiency of this approach is clearly also dependent on the original size of the payload to include in a frame. The DCAD end-user service induces a fairly low payload size ($\approx 30\ B$). Thus effects of frame size control may be limited in the studiet case.

MAC

It has been argued how efficient MAC methods may significantly increase reliability of broadcasting. However, the approach in this work is to consider the existing CSMA/CA mechanism in 802.11. In section 3.1 it was discussed how 802.11 does not engage a random back-off when a received packet is to be forwarded and the channel is idle. A simple transmission time jitter was introduced to avoid this problem. This mechanism introduces a single layer parameter Tfw_{max} defining the maximum added delay of a transmission. Increasing Tfw_{max} may in these cases reduce the probability of collision at the cost of increased jitter and delay of the individual packets.
In the same discussion it was emphasized that only the minimum contention window CW_{min} was relevant to consider. The motivation to adjust Tfw_{max} also applies to CW_{min}.

3.2.3 Resilience in the broadcasting mechanisms

Resilience optimization of broadcasting naturally follows the discussion in ensuring a reliable and efficient broadcasting method. Clearly available resilience mechanisms depend on the broadcasting scheme. In this section two examples are provided in relation to the simple and enhanced broadcasting schemes introduced in this work. The important resilience metrics when optimizing the schemes are *message loss* and *message delay* in relation to broadcast destination nodes.

Packet bundling
The DCAD end-user service relies on multipoint to multipoint communication. This leads to many concurrent sources of a broadcast and consequently increased contention. A mechanism to potentially decrease contention should do buffering of packets to be forwarded and include these in larger packets. This could reduce the amount of transmissions at the cost of increased message delays and the risk of loosing more information in a single packet loss. Layer variables control maximum buffer delay and the bundled packet size.

Forward node selection
In CDS based broadcasting schemes the selection of forwarding nodes has a large significance on both efficiency and reliability of the broadcasting scheme. In the selection of forwarding nodes it is possible to vary the density of forwarding nodes. CL metrics like BER of individual links may also be introduced to select forwarding nodes that decrease the probability of message losses. In essence these options offer an instrument to optimize the relation between efficiency and reliability from a cross-layer perspective.

3.2.4 Resilience mechanisms in the application layer

Clearly the end-user service itself could also include resilience mechanisms that would highly relevant to include in the cross-layer optimization approach. An example is the DCAD end-user service. In the detailed service description of the service (in Appendix B on page 148) an observer-based control mechanism is described to use the broadcast location and motion information from surrounding cars to assist the driving. The basic idea behind observer-based control is that observations are filtered through a state space model. In this case the model describes the motion of the surrounding cars. With new observations the model states are updated. However the model is also predicting future states. A parameter L (observer gain) defines a weight between the observations and the predictions to perform state updates. In cases where observations are faulty or missing, the gain may be weighed at trusting the predictions more than observations. This could ensure service continuity in cases where the communication becomes unreliable.

3.2.5 Layer parameters and resilience metrics

The identified resilience mechanisms, layer parameters and their corresponding resilience metrics are gathered in Table 3.2. In the next section a subsets of these mechanisms is defined to establish the resilience optimization problems that are subject for further analysis and development.

3.3 Optimization problems

The resilience metrics introduced in Table 3.2 offer many interesting options to attempt optimization of resilience in relation to the broadcasting service enabling the DCAD end-user service. To limit the complexity of studying the cross-layer optimization approach of this work, only a subset of the mechanisms are considered. The selection criteria have been to focus on resilience mechanisms close to the unreliable channels. This is where faults should be handled and prevented before they propagate to the upper layers in the protocol stack. Thus the resilience mechanisms of ***transmission power control***, ***rate adaptation*** and *FEC* form the basis of an optimization problem. In future work it would be obvious to add other mechanisms to address limitations of this initial approach. Previously both a simple and an advanced broadcasting scheme has been introduced. For the optimization problem a starting point is made in the simple flooding broadcasting scheme. This scheme may not necessarily lead to the least complex models for optimization of the two. Clearly in relation to efficiency the CDS-based scheme is advantageous. However, the assumptions of operation for flooding are simple and equal for all nodes which expectedly simplifies result analysis. In addition basic flooding algorithms are readily used in current ad-hoc network protocols like DSR. While considering flooding in the scope of this work clearly cross-layer optimization in relation to the most recent developments in efficient and reliable broadcasting protocols would also be of relevance in future work.

Parameters, variables and metrics

In section 2.4 on page 29 the terminology framework used in this work has been introduced (see table 2.3). With the delimitation of the considered resilience mechanisms the cross-layer optimization approach can formally be defined in this framework.
From the selected resilience mechanisms, layer parameters exist in layers 1 and 2 to give the following layer parameters and their valid values:

Layer	**Resilience mechanisms**	**Layer parameters**	**Resilience metrics**
Application	Observer based control	Observations/ model balance	Prediction accuracy
Network (broadcast)	Packet bundling	• Packet size • Max buffer delay	• Message loss rate • Message delay • Packet loss rate • Forwarding node ratio
	Routing	• FW node density • Control packet TR. • FW node selection	
D-link	FEC	Code rate	• Frame loss rate • Frame delay • Contention
	ARQ	• Type • Retry limit	
	Frame size control	Frame size	
	MAC	• CW_{min} • Tfw_{max}	
Physical	Rate adaptation	PHY mode	BER
	Transmission power control	Power Tx./Range Tx.	
	Channel control	Channel	

Table 3.2: *Resilience mechanisms in the protocol stack, their offered layer parameters and resilience metrics.*

Tx power (α_1^1)	PHY mode (α_2^1)	FEC Code rate (α_1^2)
$-10\ dBm \ldots 30\ dBm$	1, 2, 5.5, 11 $Mbit$	$\frac{1}{1} \ldots \frac{1}{2}$

Further, the following layer parameter vectors are established:

$$\theta_1 = \{\alpha_1^1, \alpha_2^1\},\ \theta_2 = \{\alpha_1^2\} \tag{3.1}$$

$$\Theta = \{\theta_1, \theta_2\} \tag{3.2}$$

The layer parameters are defined in Appendix A.1 on page 146.

The overall resilience and performance of the broadcasting service is as stated in section 3.1 evaluated from the probability of a node failing to receive a message and the end-2-end delay. Requirements for these metrics are given in the QoS requirements vector: $\Phi = \{\phi_1, \phi_2\}$ with $\phi_1 = P_{mf}$ and $\phi_2 = D_{e2e}$.

Effects of adjusting layer parameters

Based on the previous analysis an overview of the effects from adjusting the introduced layer parameters is given.

Considering initially an increase in transmission power. This action will *increase the transmission range* and improve the *SNR* at nodes within transmission range. In practice this leads to a larger range allowing *more nodes to be reached* in a single transmission. As a result a *reduction in the amount of hops* is achieved decreasing delays of a broadcast message. The increased *SNR* and redundancy from more transmitting nodes should *decrease the probability of frame losses* due to link conditions and *increase the probability of nodes successfully receiving a message*. However, this last consideration is not ambiguous. As transmission range is increased so is the collision range. In addition the *increased contention* in the wireless links leads to longer delays for nodes to access the medium. Consequently the resulting message loss and end-to-end broadcast delay may increase.

Similar considerations exist for the physical mode. Increasing the rate leads to *shorter transmission times* and thereby reduced contention in relation to the amount of data being transferred. Yet, higher rates rely on modulation schemes more sensitive to the SNR which again could decrease the efficient transmission range and increase frame losses from bit errors. Again, the effects on message losses and delays are ambiguous.

FEC can correct bit errors and as a result decrease frame loss. However, the increased frame length from redundant bits increase transmission time and the probability of collisions.

Defining the objective functions

Finding the optimal values of Θ_{opt} requires the definition of the objective function. In this work the objectives are to increase the probability of successful message reception within the zone while minimizing the end-to-end delay. According to Equations 2.2 and 2.3 the multi-objective optimization function can be defined as:

$$\Theta_{opt} = \underset{\Theta \in \Gamma}{\operatorname{argmin}} \begin{bmatrix} f_1(X, \Phi, \Theta) \\ f_2(X, \Phi, \Theta) \end{bmatrix} \quad \text{where} \quad \Theta = \{\theta_1, \theta_2\} \tag{3.3}$$

$f_1 = P_{mf}$ and $f_2 = D_{e2e}$. The properties of these metrics defines the optimization objective to be a minimization of the two.

Conditions and end-user service parameters

The optimization of equation 3.3 depends on both conditions and the end-user service parameters. As discussed previously a condition may change during operation. Only one condition is considered subsequently. This is the *density* of nodes/cars in the environment. As the density increases contention increases as well. This requires a new estimate of Θ_{opt} to enable an adaption to these conditions. Leaving the discussion of a practical implementation of the optimization until section 4.1 at this point the condition is defined as an observation:

$$\omega_1 \;\; = \{\beta_1^1\} = density \; [nodes/m] \tag{3.4}$$

Besides varying conditions varying service parameters also affect equation 3.3. As described in the DCAD service description the *broadcast frequency* and possibly *zone range* may vary depending on the car velocity. In addition the DCAD service may have different requirements to the payload size depending on its offered functionality. An overview of the service parameters is found in Table 2.2 on page 18.

With the optimization problem defined the challenges remain to attain the system models to define the functions f_1 and f_2. Further, it must be considered how the optimization of equation 3.3 can be realized to improve reliability and efficiency in the broadcast service. These topics are the main focus of Chapter 4.

Chapter 4

Models for optimization analysis

This chapter considers the cross-layer resilience optimization from the modeling perspective. Initially a discussion on the approach to resilience optimization is made in relation to functional requirements of a resilience optimization mechanism. This discussion also covers a brief consideration on how such a mechanism could be implemented. These aspects are extended to define an approach to find optimal parameters. This includes a general perspective on the type of models sought in this work.
Next step is to define an overall model constituted by multiple sub-models. Finally the sub-models are developed and integrated into the overall model.

4.1 Conceptual design

In section 2.4 on page 29 general approaches to establishment of CL optimization have been introduced. In this section these perspectives are considered specifically in relation to the objective function given by Equation 3.3 on the facing page. This entails two perspectives. Initially it must be considered which functional requirements exist to the optimization mechanism and how it could be expected to operate. Next it must be considered specifically how optimal parameters for the objective function can be found. These should finally be incorporated to improve resilience in relation to the flooding based broadcast service and the DCAD end-user service making use of it.

4.1.1 Desired functional properties

A strong argument in relation to introducing CLRO is the potential to dynamically adjust the functionality and resilience mechanisms of the protocol stack to handle varying conditions. This is as previously discussed particularly interesting in a highly dynamic environment given by mobile nodes operating in a wireless environment. The following considerations are made with the overall goal to establish a mechanism that may be included in a wireless node to provide an adaptive and dynamic approach in the optimization process. Following the functional requirements for such an operation is analyzed to provide a background for the subsequent optimization design.

The starting point is a deployment of the simple flooding based broadcast scheme with geocast properties. In its basic form it relies on a best effort approach where no knowledge about the operating conditions are required. This approach is in many cases successful for this broadcast mechanism as it achieves good coverage, however, at the cost of an excessive amount of overhead. These are the conditions under which the CLRO approach is applied.
To potentially improve the reliability metric P_{mf} and the performance metric D_{e2e} an optimization attempt is made to adjust transmission power/range, transmission rate and FEC code rate. The preconditions to identify the optimal parameters are observations about *node density* and information about the current configuration of the end-user service. The latter defines specifically the *broadcast frequency* and the transmitted message size. In addition QoS requirements specify how the optimization mechanism should weigh the optimization toward end-to-end delay and message reception probability.
The node density observation required is clearly not available in a standard protocol stack. However in this work an additional HIDENETS middleware layer is assumed to provide additional available services (see Section 2.1). One of these services is the *proximity map* [Casimiro et al., 2006] which gathers information about the location of neighboring nodes and the movements. This could basically be a sub-service used by the DCAD end-user service, however detailed considerations on how this could be realized are not essential. However it allows a reasonable assumption that the car position information gathered by the multipoint-to-multipoint broadcast operations may actually be useful to the resilience optimization mechanism itself. Again this raises numerous issues some of which have been mentioned earlier. For instance the efficiency of the optimization will rely on itself. I.e. as performance and reliability of broadcasting decreases, the observations to optimize operation may be degraded. In addition observations may generally just be missing, noisy, uncalibrated, ambiguous or unreliable and even malicious. This complicates these considerations additionally. These problems are not of primary concern in the remainder of the work. Thus as a simplification a reasonable assumption is made that reliable approximations of node density are available.

4.1.2 Architectural design

As clarified in Section 2.5 on page 35 the architectural design is a key aspect to consider when introducing a cross-layer approach. Good CL optimization possibilities may be showed. However, they may be of little use if they are impractical to include in a real deployment of COTS based nodes from multiple vendors. In this work a specific CLRO optimization is investigated. In reality it could potentially be one of many deployed to support multiple end-user services running concurrently across multiple nodes. Thus architectural discussions do not only consider how CLRO could be realized in terms of good design principles, but also how different optimizations would interact and effect each other. These are both major issues that are not in the scope of this work to solve. However to establish the optimization approach and discuss its capabilities it is relevant to do this in an architectural context. As a result an assumption of an architecture is established.

At this point some assumptions of cross-layer observations are already made. The introduced layer 3 based broadcast mechanism makes use of it own position information obtained in the middleware layer. Further it relates this information with broadcast source position information to determine if packets should be forwarded. This information may readily be available in optional fields of packet headers sent as well as received [Tanenbaum, 2003]. In this case the information is piggybacked. This requires only little alteration of existing IP implementations where this information is used. In addition compatibility is maintained to other IP implementations where such information would simply be ignored.

This example establishes an important point in relation to the COTS protocol stack introduced. If possible a breach of the protocol definitions should be avoided. In practice nodes with and without optimization should be capable of co-existence. I.e. a CLRO optimization mechanism is in the foundation of this work considered a feature that may optionally exist in a node.

As mentioned previously it is expected that multiple CL resilience optimizations would operate concurrently. Thus the architectural discussion should from that particular perspective aim at a general approach to offer different CLRO optimizations. A possibility could be that the CLRO mechanism is a part of the HIDENETS middleware layer. It would make use of knowledge about specific end-user service QoS requirements and usage to activate and execute relevant optimizations dynamically. This corresponds well to the definition of the general view on CL optimization provided in Figure 2.4 on page 29. It could be considered as the functional outline of an optimization middleware service. This also corresponds well to the *vertical calibration* approach described in the survey in [Srivastava and Motani, 2005].

A practical question in the taken perspective is if the desired observations can be obtained from different protocol implementations and if they offer interfaces to establish desired layer parameters. In the considered case this is not a problem as the density is an observation from another middleware service. In addition this work does not include a real implementation of the optimization approach. However this is an important point to consider in future work when moving from simulation based analysis to real implementation cases.

4.1.3 Scope of optimization

In Section 1.2 on page 9 the concepts of node local and node global optimizations have been introduced. In a node local perspective the node only optimizes its own operation to improve its service provisioning or service usage. In the node global perspective optimizations are considered to optimize all nodes. The DCAD end-user service is in principle global as all nodes both provide and make use of it. Thus the consideration made in this work is also global. I.e. the aim is to ensure that most possible nodes will receive a correct service.

The global optimization problem could be defined in various ways. An option would be to consider mean values of performance and resilience metrics over all nodes within a zone. However, this makes it difficult to evaluate whether the individual nodes actually experience the required QoS. To enable this perspective an approach is taken to base the optimization on the QoS experienced by the individual nodes. This allows an approach where optimization of the QoS at the presumed *worst case* node would ensure that all other nodes experience an optimized service as well. Intuitively this is a node within, but close to the edge of a zone. It is situated as far from the source node as possible. In addition it has the fewest number of neighbor nodes to retransmit messages, and thereby increase message reception probabilities. This is exemplified in Figure 4.1. The properties of this approach is later discussed in relation to simulation and model results.

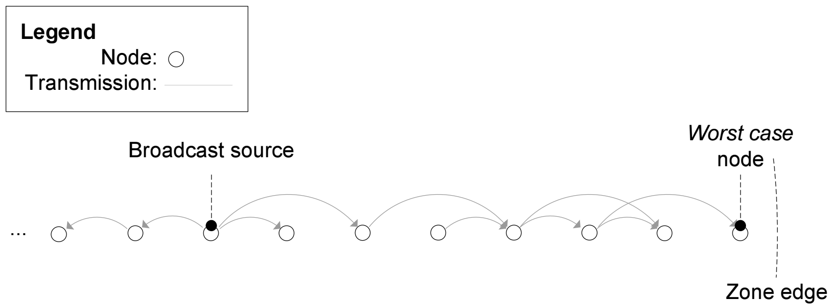

Figure 4.1: *The worst case node is assumed to get the worst QoS. The optimization aims at improving QoS for this node to increase the QoS experienced within a zone.*

4.1.4 Optimization methodology

At this point the objective function has been defined and the desired functional properties of CLRO introduced. Next step is to consider the model methodology used to enable decisions of layer parameter settings based on car density observations (see Figure 2.4).

In the pre-analysis different approaches have been presented to solve CL optimization problems in terms of modeling and decision making. An interesting approach is to consider methods like Markov Decision Processes where the modeling design and decision approach are combined. This could enable models tuned for the purpose of making optimal decisions. This would be highly relevant to create models that only contain relevant aspects and consequently are efficient to solve. However, to fully understand what these *relevant aspects* are an elaborate modeling approach could be introduced as the first step in this process. This should clarify how the different mechanisms of transmission, MAC and flooding broadcast can be modeled and just as important integrated. Based on this motivation in this work a starting point is taken in elaborate models covering the different system components. These models are then considered in the context of making CLRO of the flooding broadcast mechanism.

In the following section an overall model design is defined. Sub-models are created to split up the problem. As a result different modeling approaches may be used to describe the given sub-model. The aim of this approach is to define an overall model enabling the derivation of f_1 and f_2 to include in the objective function in eq. (3.3).

Optimal decisions
Decisioning based on this model approach takes a basic starting point in conducting the optimization of the objective function. The aim is to find the best setting of the layer parameters in Θ. As closed form solutions for f_1 and f_2 may be difficult to obtain, search based methods must be applied. An approach to handle the multi-objective optimization problem is to define a cost metric that weighs metrics of e2e-delay and message loss probability in relation to QoS requirements.

A part of the motivation for this work is to enable a dynamic optimization approach. This defines requirements for the models and the decision method in relation to performance and efficiency. In the following the suggested models are analyzed from a static perspective to compare their accuracy to simulation based models. However their applicability is additionally discussed in the perspective of dynamic optimization.

4.2 Model design

As mentioned the approach to model the components of the optimization problem is to split it into several sub-models that are tied together by an overall model design. In this section this overall model design is specified.
Defining multiple sub-models enables an individual development of each sub-model. Primarily this may reduce the complexity of developing the model. In addition the definition of sub-models can partially be based on layers in the protocol stack. This allows a starting point in already studied aspects like channel models.

The studied CL resilience optimization approach covers layers 1 and 2. Consequently the main focus of the overall model is to capture (1) behavior of transmissions in a wireless environment and (2) the functionality of layer 2 resilience mechanisms to prevent and correct faults in the wireless environment. The DCAD end-user service defines the perspective for the overall model. This implies a model of the broadcast mechanism and requirements to which model parameters and metrics are to be included in the model.

The outline of the overall model is given in Figure 4.2. The sharp boxes define the sub-models whereas the rounded boxes describe aggregations of model outputs to provide the desired model internal and external metrics. It is important to emphasize that the sub-models do not explicitly model the layers of the protocol stack. This will become clear in the subsequent presentation of the model components.

4.2.1 Model components and interfaces

Following the sub-models and their interfaces are introduced. A complete list of the interfaces may be found in Appendix C.1 on page 152.

PHY sub-model
The PHY model represents both the physical channel and the physical layer of 802.11b. Overall this sub-model is responsible for providing metrics that can be

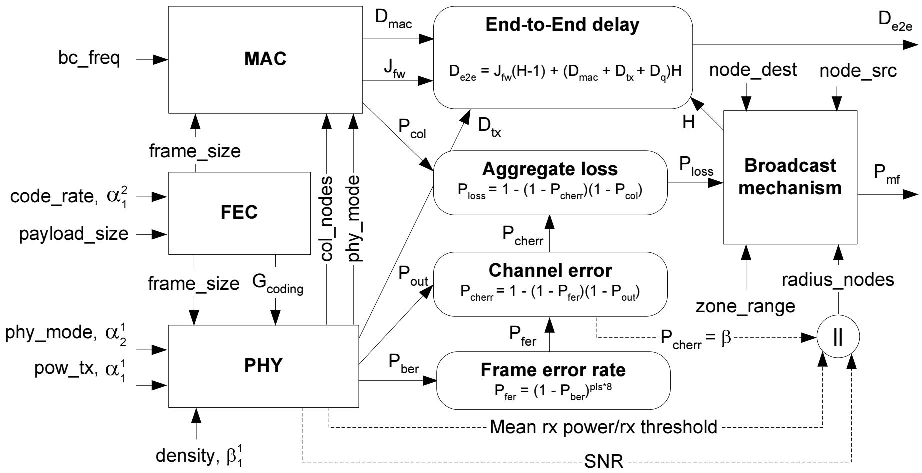

Figure 4.2: *Components of the overall optimization model.*

used to calculate frame losses given errors that have appeared in the channel. The PHY model includes channel modulation schemes making α_2^1 an input to this model. Clearly also the transmission power α_1^1 is a part of this model. The model outputs two metrics in relation to the calculation of frame losses: (1) P_{ber} is the bit error probability and (2) P_{out} the 'outage probability' which defines the probability that a message is discarded due to an insufficient signal strength in the transmission. From P_{ber}, the frame error probability P_{fer} is easily derived. Finally the probability of a frame loss due to a channel error P_{cherr} can be calculated from the two. Notice that a frame cannot be lost due to a bit error if the message has already been dropped in relation to P_{out}. As additional input the PHY model gets the actual frame size to be transmitted on the medium. This frame size depends on the amount of redundancy added by FEC. In PHY the frame size and knowledge about the data rate is used to calculate D_{tx} which is the transmission time of the frame.

The PHY model also includes the observation of *density*. The density is calculated from the amount of nodes within a zone in the unit of $[nodes/m]$. Based on a given transmission power and models of signal attenuation the PHY model can provide a measure for the mean amount of nodes that can be reached by a transmission. This information is used by the MAC model to estimate collision probabilities. The broadcast mechanism model needs the information to establish how many nodes are reached in a broadcast. As the collision range and transmission range are two different metrics two different interfaces are provided. Notice that the interface `radius_nodes` is defined in three different ways. This indicates that the definition of the transmission range could be based on different assumptions. For instance the transmission range may be defined by some probability limit of P_{cherr}. However, it could also be defined in relation to a certain level of received signal power. Besides defining the derivation of `radius_nodes` this definition also affects the derivation of P_{loss}. This aspect is discussed further in the PHY model design.

Finally, the PHY model provides the MAC sub-model with information about which PHY mode is selected. The last input parameter is G_{coding} which is presented in relation to the FEC sub-model in the next paragraph.

FEC sub-model
While being a layer 2 service FEC is closely related to the PHY model. It has been defined explicitly in the overall model to emphasize its influences in the model. The FEC sub-model uses the end-user service payload size to calculate the actual frame size given the amount of redundancy added. The amount of redundancy is controlled by the code rate α_1^2. In the FEC model this leads to a certain coding gain G_{coding} which in the PHY model has a direct effect on the resulting bit error probability. This aspect is also discussed further in the PHY model design.

MAC sub-model
The overall purpose of the MAC model is to enable a calculation of P_{col}, the probability of collisions. As mentioned it is based on a collision radius provided by the PHY model. Additionally the MAC model uses information of frame sizes and the rate (PHY mode) to calculate for how long time a frame occupies the medium. From the MAC model a mean MAC delay D_{mac} can be calculated which depends on the level of contention on the medium. In addition the MAC model also includes the synthetic jitter introduced in section 3.1 to decrease collision probabilities.

Aggregate loss
For simplicity it is assumed that the frame losses due to collisions are independent of frame losses due to channel errors. This finally enables a derivation of the *aggregate loss* probability P_{loss}. This is the overall mean probability that frame is lost in a transmission between two nodes. In the remainder of this work either P_{loss} or its counterpart $P_s = 1 - P_{loss}$ is used.

Broadcast mechanism sub-model
The broadcast mechanism sub-model is responsible for the derivation of P_{mf}. P_{mf} is the mean probability that the destination node `node_dest` will fail to receive a broadcast message from `node_src`. It is based on the mean radius of nodes that will receive a frame with the probability $1 - P_{loss}$. Additionally the zone range defines how far a message is propagated. Finally H is the mean amount of hops the node `node_src` needs to travel from the source node. This clearly depends on `radius_nodes`.

End-to-end delay
The end-to-end delay is defined in terms of the number of hops H, and the delays that occur at each hop. The forwarding jitter J_{fw} is a small random delay that is added when a node forwards a broadcast in order to avoid collisions. Since this jitter is only applied when forwarding and not at the source broadcast, it is multiplied by $H - 1$.
For every transmission, a node is delayed while trying to access the medium. This MAC access delay D_{mac} depends on the contention window size, number of

contending nodes, and level of contention. If a node receives further messages to forward while trying to gain medium access, the incoming messages are stored temporarily in a queue. If the level of contention is sufficiently high, a node may not be able to forward all received messages, and this queue delay D_q will increase until the buffer overflows. Finally, when a node has gained access to the medium, the transmission takes a certain time to complete. This transmission delay D_{tx}, depends on the PHY mode and the frame size, which may vary with the level of FEC.

The end-to-end delay is considered in relation to the model verification in Chapter 6.

Assumptions of car positions

As delimited in the problem statement, the considered stretch of road has two lanes with cars traveling in the same direction. The two lanes are considered as merged into a single lane. Since the lanes are close to each other, this assumption is not expected to have a big influence in practice.

As an aid in the model design a set of general assumptions are defined about the positioning of cars in the environment. This leads to two definitions:

Equidistant Here the cars are distributed along a straight line, with a constant distance between subsequent cars. The equidistant car positioning provides a deterministic base for model evaluation, since the number of neighboring nodes is constant.

Poisson Instead of distributing the cars with equal distance, the cars are distributed according to a Poisson process, thus resulting in exponentially distributed distances between cars. This enables the assumption of independent car positions, which is a helpful simplification in the modeling process.

Model assumptions and their effects on optimization

This model does not consider individual setting of layer parameters for each node. This is expected to simplify the model considerably, however it also defines some limits in the optimization. Specifically optimal settings of transmission power may be given by setting each node individually in relation to the density of a small part of the zone. Future extensions of the overall model should also enable this possibility. Possibly simplifications could be made by defining the node global optimization problem into a subset of smaller node groups where optimality is ensured.

4.2.2 Model properties

A CL optimization could take into consideration the feedback that exists in ad-hoc networks. An example is the case where an increasing broadcast rate or increasing number of nodes in the network lead to higher degree of contention, which causes frame losses due to collision and interference, which in turn leads to less forwarded messages. The result is the initial broadcast rate has not been achieved. Such a feedback effect is currently not taken into consideration in this work.

The aspect of handling unreliable observations is an important topic to consider when implementing a CL optimization. However, the focus of this work is on the conceptual functionality of CL optimization in MANETs, and unreliable observations is therefore a task to be tackled in future work.

4.3 The physical channel submodel

In MANETs the physical environment plays an important role in relation to the performance of the communication, since physical phenomena deteriorate the conditions of the communication and may lead to link breakage and frame loss. The purpose of this model is to estimate the experienced frame loss probability due to phenomena existing in the wireless channel, P_{cherr} and determine the number of nodes within transmission and collision ranges for the broadcast mechanism and MAC model.

There are two main aspects that determine the amount of frame losses in the physical channel: radio wave propagation environment and the techniques used for modulating and coding the transmission. The radio wave propagation environment defines the non-controllable conditions, whereas the modulation and coding of the transmission in the form of the PHY modes of the wireless adapter and FEC are controllable parameters, which may be used to adapt to the physical environment. This section concerns the modeling of propagation, modulation and FEC functionalities in relation to the defined service case. Since 802.11b is used for wireless communication in this work, the following analysis will consider modeling of the PHY functionalities in this context.

The output of this submodel is the average number of neighbor nodes, the average number of collision nodes and P_{cherr}. P_{cherr} is an average over the channel error probability within the transmission distance. The calculation of P_{cherr} is based on radio propagation and channel models and takes the used modulation scheme into account.

4.3.1 Radio propagation

As radio waves propagate from the transmitter to the receiver they can be said to suffer from three separate phenomena: *Path loss*, *shadowing* and *multi-path fading* [Goldsmith, 2005]. Generally, a model of the radio propagation can be said to describe the relationship between the transmitted and received power.

The path-loss that is the direct consequence of the distance between the transmitter and receiver is typically denoted *large scale fading*. Shadowing is caused by large obstructions such as buildings. The effect of shadowing is a local fading of the received signal when the receiver is in the shadow of the obstruction. Shadowing is also referred to as large or medium scale fading effect. The last phenomena, multi-path fading, is typically seen as rapid variations in the received power due to rays being reflected, diffracted and scattered, and is thus denoted small-scale or fast fading.

Many different radio propagation models exist that describe the relation between the transmitted and received signals exist. The best choice of model depends on the environment that is being modeled and the intended use of the model. In relation to the service case defined in section 2.1, the considered propagation model should take into consideration that the considered environment is rural with a direct Line of Sight (LoS) between the receiver and transmitter. Further, both nodes are mobile and there are no large obstructions in the environment that cause shadowing.

This means that typical models such as the log-normal and Rayleigh are not considered, since they are typically used for modeling shadowing and multi-path situations where no line of sight is present, respectively.

The purpose of the model is to obtain a realistic representation of the radio channel in the service case. Since this case is fictitious, no empirical measurements are available to parametrize the models. Thus, the models are parametrized according to the definitions in the service case and expected properties of the environment. Table 4.1 presents the used model parameters. The following presents the propagation models used in this work. The descriptions are based on [Goldsmith, 2005] and [Prasad, 1998].

Friis free-space

A basic propagation model is Friis' free-space model that assumes a clear line of sight and no multi-path or shadowing effects between the transmitter and receiver. The received power P_r at distance d is given by

$$P_r(d) = \frac{P_t G_t G_r \lambda^2}{(4\pi)^2 d^2 L} \tag{4.1}$$

Where P_t is the transmitting power, G_t and G_r the transmitter and receiver antenna gains, λ is the wavelength, and L is the system loss factor, which is usually 1.

Two-ray

In situations such as in the considered service case where a direct LoS ray and an additional ground reflection is present, the two-ray model approximates the path loss via the following expression:

$$P_r(d) = \frac{P_t G_t G_r h_t h_r}{d^4 L} \tag{4.2}$$

where the additional parameters h_t and h_r are the antenna heights. This model has been shown to give a better prediction of the path loss at long distances than the free space model. For an actual radio setup, at short distances, the result of the constructive and destructive interference between the direct ray and the ray that is reflected on the ground is actually represented better by the free space model. A combined model that includes both the free-space and two-ray models is therefore typically used. The free-space model is used up to a given cross-over distance and the two-ray model at distances greater than this cross-over distance.

When d is large compared to $h_t + h_r$, the cross-over distance is given by

$$d_c = \frac{4\pi h_t h_r}{\lambda} \tag{4.3}$$

In [Paolo Barsocchi and Potortì, 2006] the authors compare measurements from an urban environment based on 802.11b to the prediction of this combined free-space and two-ray model. From these measurements they find that when using vertical antenna polarization for 802.11b equipment the received power level at $16m$ drops to a level which equals the level at $160m$. They develop an extended two-ray model which is able to predict these dips. However in the following, the simple combined free-space and two-ray model will be used for simplicity, but it is worth noticing that the performance in an actual system may deviate significantly from this simple model. The extended two-ray model could be a possible extension to obtain a more accurate representation of the path loss than with the simple combined free-space and two-ray model.

Ricean

In the considered service case, the nodes are cars where a direct LoS between the transmitter and receiver node exists. A dominant radio ray follows this path to the receiver. In addition to this direct LoS ray, radio rays are also subjected to reflection, diffraction and scattering due to other cars, guard rails, the road pavement, etc. The characteristics of these multi-path radio rays are changing continually due to varying conditions such as mobility. The Ricean model is a statistical model that describes the small-scale fast-varying power level of the received signal when subjected to multi-path.

The Ricean model relies on the Ricean K-factor, which is presented shortly, and the local-mean power, which is the average power level at the receiver P_r calculated from path-loss models.

The instantaneous power p of the received signal subjected to multi-path fading may be described by its pdf:

$$f_p(p|P_r) = \frac{(K+1)p}{P_r} exp\left(-\frac{(K+1)p^2}{2P_r} - K\right) I_0\left(2\sqrt{\frac{K(K+1)p}{P_r}}\right) \quad (4.4)$$

Where $I_0(\cdot)$ is the zero-order modified Bessel function of the first kind. The Ricean distribution is characterized by the Ricean K-factor given by

$$K = \frac{s^2}{2\sigma^2} \quad (4.5)$$

Where $\frac{1}{2}s^2$ is the signal power of the dominant LoS component, whereas σ^2 is the local-mean power of the multi-path components. For $K = 0$ there is no dominant ray and the expression in (4.4) simplifies to the Rayleigh distribution, where all multi-path rays are considered equally strong and the instantaneous power is given by an exponential distribution.

The actual fading of a signal is a time-dependent process that depends on the wavelength of the transmitted signal and the velocity of mobile nodes. Thus, besides the power of the faded signal, other metrics that are important when considering fading models is the fade and non-fade durations and the level crossing rate. However, for a starting point these metrics are not considered specifically in this work.

Model parameters

Table 4.1 presents the parameters used in the above models. The parameter values have been chosen based on the considered service case. In [Davis and Linnartz, 1994] measurements of the Ricean K-value has been made in vehicle-to-vehicle situations, and typical values were found to be in the range of 5.0 to 11.0. A

h_t	1.5 m	Height of transmitter antenna.
h_r	1.5 m	Height of receiver antenna.
G_t	1	Transmitter gain.
G_r	1	Receiver gain.
L	1	System loss factor (1 equals no loss).
λ	$\frac{3 \cdot 10^8 m/s}{2.472 \cdot 10^9 Hz} = 0.121m$	Wavelength for channel 13.
K	6	Ricean K-factor.

Table 4.1: *PHY model parameters.*

Ricean K-value of 6 is therefore assumed to be representative for the type of environment considered in this work.

As described above, different physical phenomena deteriorate the quality of wireless transmissions. In addition to these, also the interference caused by collocated transmitting nodes adds to the noise in the channel and degrades the performance. In relation to the optimization objective, this could be a relevant factor to take into consideration for the transmission power control. However, initially this effect is not considered for the PHY model, but it may be added later to improve the accuracy of the model.

4.3.2 802.11b Performance

In order to cope with varying conditions, different modulation schemes and coding rates, so-called PHY modes, are used in wireless networking equipment. The PHY modes have different properties with respect to resilience against channel errors and as a result hereof different performance. The effect that varying conditions has for a transmission given a specific PHY mode, is typically quantified in terms of the probability of bit errors, the Bit Error Rate (BER), and the probability that the received power level drops below a certain threshold, the *outage probability* as described in [Goldsmith, 2005]. Both of these depend on the channel conditions and the modulation and coding used for transmitting.

The 802.11b operates in the 2.4 GHz area and provides 4 PHY modes with data rates of 1, 2, 5.5 and 11 $Mbit/s$ that use different modulation and coding schemes. Due to these differences, the performance also differs between the PHY modes. In the following it is assumed that the considered wireless adapter is an ORiNOCO 11b card based on the Intersil Prism chipset.

The ORiNOCO 11b wireless adapter is based on the Prism chipset, which uses the Intersil 3861B DSSS baseband processor [PRISM, 2001]. This processor has different modes of operation that differ in the reception of weak signals. The threshold that defines this, here denoted the *receive threshold*, may be set to an absolute power level or a relative level above the noise floor. The noise level is measured continually when no carrier sense is detected and the noise floor is thus dynamically adapted. The processor does not attempt to decode transmissions that are below the selected threshold. This property is useful for the high rate modes, where the processor may have to discard weak transmissions very often due to bit errors. By filtering out these weak transmissions, the processor is free to receive a later but overlapping and stronger transmission instead. In this work it is assumed that the *receive threshold* is defined in terms of an absolute power level threshold. The data sheet [PRISM, 2001] suggests that this threshold is set to a value between $-70\ dBm$ and $-80\ dBm$. The receive threshold is therefore set to an absolute power level of $-75\ dBm$. In addition to the receive threshold, also a carrier sense threshold is used to determine when the channel is sensed idle. The purpose of this parameter is to be able to limit the level of interference between transmissions. Here a value of $-85dBm$ is assumed for the carrier sense threshold.

In [Xiuchao, 2004], which considers radio propagation in an indoor environment, receive and carrier sense thresholds have been defined to $-95\ dBm$ and $-104\ dBm$, respectively. In the model verification in section 6.1 on page 111 results based on these parameters are compared to the values used in this work that are defined above.

To be able to calculate the SNR, which is needed in the following, it is necessary to define the considered noise floors. Since the noise floor is adapted dynamically for the Prism chipset, the noise floors are defined from the receiver sensitivity. In the specifications [Proxim, 2003], the receiver sensitivities are given as shown in Table 4.2. The receiver sensitivity is a property of the wireless adapter that expresses at how low a power level the adapter is able to decode a certain fraction of received frames successfully. For the exact definition refer to the 802.11b standard in [IEEE, 2003]. In [Xiuchao, 2004] it is argued that the noise floors for the different PHY modes are 10 dB below the receiver thresholds. This is also assumed for this work. These noise floors are used in order to calculate the Signal to Noise Ratio in the following.

PHY mode	DBPSK	DQPSK	CCK 5.5	CCK 11
Receiver threshold	-94dBm	-91dBm	-89dBm	-85dBm
Noise floor	-104dBm	-101dBm	-99dBm	-95dBm

Table 4.2: *Used parameters for the ORiNOCO 11b wireless adapter.*

802.11b PHY modes

The following summarizes the functionality of the different PHY modes in 802.11b. For further details refer to [IEEE, 2003] and [PRISM, 2001].

All 4 PHY modes use Direct Sequence Spread Spectrum (DSSS) modulation. The concept of DSSS is that the information-bearing binary signal is multiplied by a pseudo-random binary coding signal, which ensures that the modulated signal is spread and uses a wider band. This coding signal consist of a number of chips that can be either real $(+1, -1)$ for 1 $Mbit/s$ or complex $(+1, +1j, -1, -1j)$ for 2 $Mbit/s$. Both the 1 $Mbit/s$ and 2 $Mbit/s$ PHY modes in 802.11b use an 11 chips Barker sequence to encode each symbol. In 1 $Mbit/s$ mode 1 bit is transferred in each symbol by using Differential Binary Phase-Shift Keying (DBPSK), while the 2 $Mbit/s$ mode transfers 2 bits in each complex symbol by using Differential Quadrature Phase-Shift Keying (DQPSK). The 1 $Mbit/s$ and 2 $Mbit/s$ modes both operate at a chip-rate of 11 $Mchips/s$ and thus, a symbol-rate of 1 $Msymbols/s$.

The 5.5 $Mbit/s$ and 11 $Mbit/s$ PHY modes of 802.11b use 8 chip long Complimentary Code Keying (CCK) codes instead of Barker codes. In 5.5 $Mbit/s$ mode the data signal is treated in groups of 4 bits. Here 2 bits are used to select the spreading function from 4 available CCK codes, and the last 2 bits are used to modulate that complex symbol using DQPSK. The 11 $Mbit/s$ mode treats

8 bits at a time, and uses 6 of those bits to select a spreading function from 64 available CCK codes. Like for the 5.5 $Mbit/s$ mode the last 2 bits are used to modulate that complex symbol using DQPSK. The 5.5 $Mbit/s$ and 11 $Mbit/s$ modes both operate at a chip-rate of 11 $Mchips/s$. But with only 8 bits per symbol, the symbol-rate is 1.375 $Msymbols/s$ in order to achieve 11 $Mchips/s$.

Generally, Phase-Shift Keying (PSK) modulates an Radio Frequency (RF) carrier by shifting the phase according to a data signal. Differential Phase-Shift Keying (DPSK) works relatively, and changes the phase based on the data signal rather than setting the phase as in standard PSK. This helps to overcome certain challenges and allows for a simpler receiver design. The two variants of DPSK used in 802.11b are DBPSK and DQPSK. They use 2 and 4 phases, respectively, to modulate the RF carrier. With 4 phases DQPSK allows a two times higher data rate than DBPSK, however at the cost of higher noise sensitivity.

4.3.3 Signal to Noise Ratio definition

Since different variants of the Signal to Noise Ratio (SNR) is used extensively in the following, the following defines these. A more elaborate definition of the SNR variants is given in Appendix C.3 on page 155. The SNR is generally defined as the ratio between the received signal energy and the received noise energy:

$$SNR = \frac{E_c}{N_0} = \frac{E_s G}{N_0} = \frac{E_b H}{N_0} \tag{4.6}$$

Here, E_c corresponds to the received power resulting from path-loss, shadowing and multi-path effects, while the noise power is defined by thermal noise or electrical noise in the receiver equipment. G is the number of chips per symbol while H is the number of chips per bit. The values of G and H depend on the used 802.11b PHY mode. For 1 $Mbit/s$ DBPSK $H = G = 11$ since only one bit is sent in each symbol with a length of 11 chips. 2 $Mbit/s$ DQPSK transmits 2 bits in each 11 chips symbol, thus $H = 5.5$ while G is unchanged at $G = 11$. For the CCK based PHY modes the symbol length is 8 chips, and thus $G = 8$. 5.5 $Mbit/s$ CCK transmits 4 bits per symbol resulting in $H = 2$, while 11 $Mbit/s$ CCK transmits 8 bits per symbol, resulting in $H = 1$.

4.3.4 Outage probability

As mentioned previously, the probability of an outage is an important aspect to consider in mobile environments where multi-path fading is present, since the power level may occasionally drop below the required minimum, thus leading to frame loss. The outage probability is given by [Goldsmith, 2005]:

$$P_{out} = P(\gamma_s > \gamma_0) = \int_0^{\gamma_0} p_{\gamma_s}(\gamma) d\gamma \tag{4.7}$$

where γ_0 is typically a required SNR level, and γ_s is the SNR per symbol, and $p_{\gamma_s}(\gamma)$ is the pdf of the instantaneous power of the fading distribution, given the local mean signal power to noise ratio per symbol γ_s.

In this work the receive threshold of the DSSS processor defines the minimum required power level. Therefore, γ_s is used to represent the "signal power to receive threshold ratio" and γ_0 is set to 1. $p_{\gamma_s}(\gamma)$ is the pdf of the instantaneous power of the Ricean distribution given in eq. (4.4). P_{out} thus gives the probability that the received signal power level drops below the required minimum threshold of $-75dBm$.

4.3.5 BER of 802.11b in AWGN channel

As described previously, the different PHY modes used in 802.11b have different characteristics with regards to resilience against bit errors. The effect of this is that the range where the amount of frame loss is acceptable, is much shorter for the PHY mode with the highest bit-rate 11 $Mbit/s$ than for the PHY mode with the lowest bit-rate 1 $Mbit/s$. The BER for a given PHY mode is typically specified as a function of the SNR per bit E_b/N_0 in an Additive White Gaussian Noise (AWGN) channel. In the AWGN channel, bit errors are assumed to occur independently, and the Frame Error Rate (FER) may be readily calculated based on the number of successful bits required to transmit a frame. This will be considered in greater details later.

In the following the expressions for obtaining the BER for the different PHY modes of 802.11b in an AWGN channel are described, based on [Proakis, 1995] and [Fainberg, 2001]. It should be emphasized that these expressions are only valid for AWGN channels. The calculation of the BER for Ricean fading channels is considered subsequently.

DBPSK

For 1 $Mbit/s$ operation 802.11b uses DBPSK modulation. The BER of a DBPSK modulation in an AWGN channel is given by

$$P_e\left(\frac{E_b}{N_0}\right) = 0.5exp\left(-\frac{E_b}{N_0}\right) \tag{4.8}$$

Where $\frac{E_b}{N_0}$ is the signal to noise ratio per bit. Since DBPSK in 802.11b uses 11 chips per bit, $\frac{E_b}{N_0} = 11\frac{E_c}{N_0}$.

DQPSK

In the 2 $Mbit/s$ PHY mode, DQPSK is used.

$$P_e = Q_1(a,b) - 0.5I_0(ab)exp(-0.5(a^2+b^2)) \tag{4.9}$$

where $Q_1(a,b)$ is the Marcum Q-function and $I_0(ab)$ is the zero-order modified Bessel function of the first kind. The parameters a and b are defined as

$$a = \sqrt{2\frac{E_b}{N_0}(1-\sqrt{0.5})} \text{ and } b = \sqrt{2\frac{E_b}{N_0}(1+\sqrt{0.5})} \tag{4.10}$$

As DQPSK uses 5.5 chips per bit, $\frac{E_b}{N_0} = 5.5\frac{E_c}{N_0}$.

CCK

For the 5.5 $Mbit/s$ and 11 $Mbit/s$ modes CCK is used. The BER for both modes are given by

$$P_e = 1 - \int_{-X}^{\infty}\left(\frac{1}{\sqrt{2\pi}}\int_{-(v+X)}^{v+X} exp(-\frac{y^2}{2})dy\right)^{\frac{M}{2}-1} \cdot exp(-\frac{v^2}{2})dv \tag{4.11}$$

where $X = \sqrt{2\frac{2E_b}{N_0}}$. For the 5.5 $Mbit/s$ mode $M = 4$ and for the 11 $Mbit/s$ mode $M = 8$.

4.3.6 Average BER of 802.11 in Ricean channel

In a fading environment the SNR of the received signal varies due to the randomness caused by shadowing and multi-path fading. Thus, the BER is not a deterministic function of the SNR as is the case for AWGN channels. The BER in a fading channel is therefore expressed as an average over the fading channel. The average BER is given from [Goldsmith, 2005]:

$$P_{e|\gamma_b} = \int_0^\infty P_e(x) \cdot f(x|\gamma_b) dx \tag{4.12}$$

Where $P_e(x)$ is determined by the modulation scheme and $f(x|\gamma_b)$ is the pdf of the instantaneous power for the fading channel. For the Ricean channel considered in this work the pdf of the instantaneous power is given from eq. (4.4).

Since the model developed in this work is considered for static as well as dynamic optimization, the computational load needed to calculate the model needs to be considered. In this context, a definition of the BER in terms of an integral as the one defined in eq. (4.12) is not very efficient to calculate. For a more efficient calculation of the BER, closed-form expressions are desirable. In the following, focus is on being able to efficiently calculate the BER of the used PHY modes in a Ricean fading channel.

Average BER of DBPSK in Ricean channel

The expression for the average BER in Ricean channel is obtained by inserting eq. (4.4) and eq. (4.8) into eq. (4.12), which gives

$$P_e = \int_0^\infty \frac{1}{2} exp(-x) \cdot \frac{(1+K)}{\frac{E_b}{N_0}} exp\left(\frac{-(1+K)x}{\frac{E_b}{N_0}} - K\right) I0\left(2\sqrt{\frac{(K^2+K)x}{\frac{E_b}{N_0}}}\right) dx \tag{4.13}$$

This expression may be reduced to

$$P_e = \frac{M}{2} exp(K(M-1)) \tag{4.14}$$

Where

$$M = \frac{(1+K)}{(1+K) + \frac{E_b}{N_0}} \tag{4.15}$$

Appendix C.3.2 on page 156 contains the specific details in the derivation of this expression. Eq. (4.14) has been verified numerically against eq. 4.13.

The average BER for DBPSK in a Ricean channel may now be calculated efficiently using the expression in eq. (4.13). The plot in Figure 4.3 shows curves for the BER of AWGN and Ricean channels. This figure shows the importance of averaging the BER over the Ricean pdf. For the AWGN channel the BER decays exponentially as the SNR increases, while in the Ricean channel the BER only decreases linearly due to the burstiness of errors in the channel. For the Ricean channel a much higher SNR is required to obtain an acceptable BER. Effectively this means that a higher transmission power is needed.

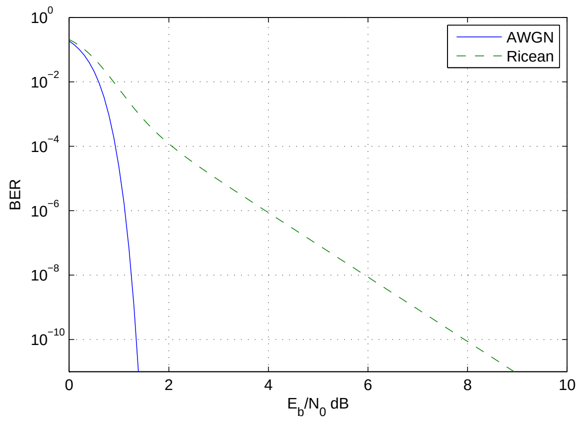

Figure 4.3: *Comparison of BER in AWGN and Ricean channels*

Average BER of DQPSK in Ricean channel

With regards to the average BER of the DQPSK modulation in a Ricean channel, the approach used above for the DBPSK modulation cannot be used. The expression for BER in an AWGN channel in eq. (4.9) is defined by the Marcum Q-function, which, as discussed in [Kam and Zhong, 1998], must be evaluated numerically. This prevents the derivation of a closed form expression for the BER in a Ricean channel. However, among others, the authors in [Ferrari and Corazza, 2004] have achieved a tight approximation to the upper bound for the expression of the BER of DQPSK in an AWGN channel. This approximation may be used in a numerical evaluation of the expression in eq. (4.12). However, it is still necessary to approximate a solution for the semi-infinite integral when evaluating numerically. The authors in [Kam and Zhong, 1998] present what they claim is an approximate upper bound to the average BER of DQPSK in Ricean channel. This expression allows an efficient computation of the BER in a Ricean channel. This approximation has been used to compute the FER, which is described in in the following, and compared against the results of the exact but intractable expression based on eq. (4.12) and eq. (4.9). The difference between the results based on the approximation and the results based on the exact numerical calculation has been found to be insignificant in practice.

Average BER of CCK in Ricean Channel

Like for the DQPSK above, the expression of the BER for CCK modulation in an AWGN channel in eq. (4.11), may not be easily averaged over the Ricean channel. A numerical evaluation is also inefficient, rendering this approach intractable in relation to dynamic optimization. In [Goldsmith, 2005] a so-called Moment Generating Function approach is presented as a simplified method for obtaining the average bit error probability in fading channels. This approach is however not investigated further in this work, but could be a possible way of obtaining a more efficient model.

As a result hereof, only the expressions for the probability of bit errors in

DBPSK and DQPSK are available, and the physical channel submodel therefore does not support the CCK modes. Figure 4.4 shows the bit error probability plotted versus the distance from the transmitter. The combined free space and two-ray model has been used as path-loss model, and a transmission power of 100 mW is used.

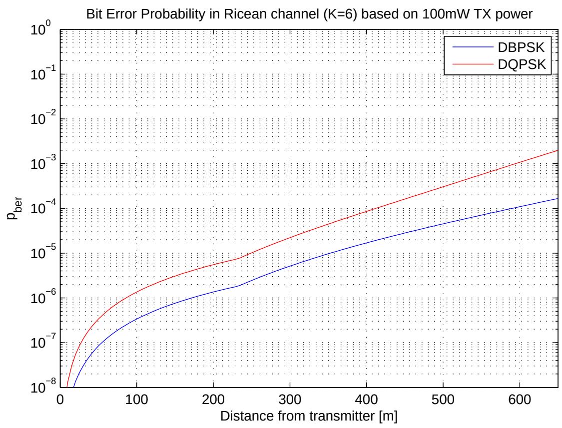

Figure 4.4: *Comparison of BER for DBPSK and DQPSK in Ricean channel.*

The graphs show that the DQPSK PHY mode has 5-10 times more bit errors than the DBPSK, which results in a higher frame error rate. For the CCK modes the bit error rate is expected to be even higher, and frame errors even more probable.

4.3.7 FEC model

The purpose of implementing FEC is to obtain a higher degree of resilience toward channel errors, such as bit errors and outages. In relation to the optimization objective in this work, FEC allows the bit error probability to be lowered without increasing transmission power, which would lead to an increase in the collision range. However, FEC has the effect that the transmission of each frame lasts longer time, and the probability of collisions within the transmission range is therefore increased. Similarly, the best case maximum achievable goodput will decrease due to the increased transmission length per frame.

In an AWGN channel where bit errors are assumed independent, a simple redundancy FEC mechanism that transmits subsequent copies of each symbol would be sufficient to combat bit errors. However, in a Ricean channel, errors may occur in bursts, which requires the use of interleaving to obtain independent bit errors, as discussed in [Goldsmith, 2005]. For the interleaving to be able to make bit errors independent, it is necessary that the duration of fades does not exceed the spreading length of the interleaving. Since broadcast messages are not transmitted in fragments, the length of the transmitted message limits the possible spreading length. Since the channel is considered a fast-fading Ricean fading channel, it is assumed that fade durations are sufficiently short to allow efficient interleaving.

Since the messages transmitted have a fixed length, the following assumes that FEC capabilities are obtained with the use of a block code such as Reed-Solomon [Proakis, 1995]. Such codes are characterized by the length of a codeword N in bits, the number of information bits in a codeword K. The codeword has $N-K$ parity check bits. The redundancy ratio may therefore be defined as the ratio between the parity check bits and the length of the uncoded block, $x = \frac{N-K}{K}$. The minimum distance between codewords is $D_{min} = N - K + 1$, which is the minimum number of bits that differ between two codewords. This means that the maximum number of correctable errors is given by $t = \lfloor 0.5(D_{min} - 1) \rfloor$, or equivalently $t = \lfloor 0.5(N - K) \rfloor$. The code rate for an FEC code is the ratio between information bits and the codeword length $\frac{K}{N}$.

The effect of the FEC coding may be expressed as a coding gain. This is the reduction in the required signal to noise ratio per bit $Eb/N0$ to achieve a specified BER when adding FEC. For low values of received SNR, the coding gain may be negative, and the FEC thus has a negative influence on the BER [Goldsmith, 2005].

The authors in [Barman et al., 2004] assume that the coding gain $G_{coding} = \frac{K}{N} d_{min}$ of a Reed-Solomon code may be multiplied by the SNR of the received signal to obtain the bit error rate of the coded transmission. In the following, the same assumption is made for simplicity. Further it is assumed that a single code block is transmitted in each frame. Thus, the coding gain

$$G_{coding} = \frac{K}{N} D_{min} = \frac{Kx + 1}{x + 1} \tag{4.16}$$

is multiplied onto the SNR level of the received signal to account for the effect of FEC.

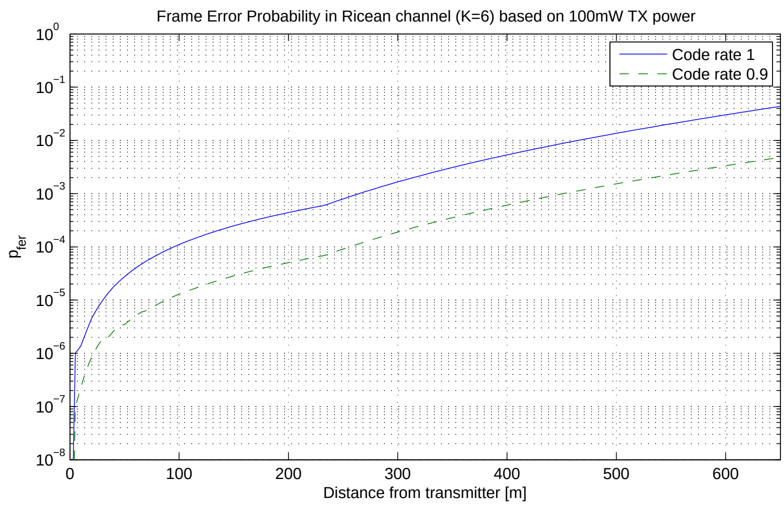

Figure 4.5: *Frame error probability without FEC coding and with FEC code rate 0.9.*

Figure 4.5 shows an example of the calculated frame error probability versus distance for no FEC and for a code rate of 0.9. By applying FEC, the frame error probability has been lowered, which reduces the amount of frames lost due to bit errors. This may help to increase the transmission distance when using high-rate PHY modes that are particularly sensitive to bit errors.

4.3.8 Calculating frame size

Since the size of the payload in the transmitted frames is constant, the frame size only depends on the amount of redundancy added via FEC. The number of added redundancy bits $N - K$, should therefore be added to the frame size, in order to account for the extra overhead, due to FEC. Table 4.3 shows the parameters that define the size of the transmitted frame.

Parameter	Size
Service payload	30 *bytes*
UDP header	8 *bytes*
IP header	20 *bytes*
MAC header	34 *bytes*
FEC redundancy	$N - K$ *bits*

Table 4.3: *Frame size definition*

In addition to these parameters, the physical layer in 802.11 uses a Physical Layer Convergence Protocol (PLCP) header and preamble. When using 802.11b, the physical layer has short preamble option for the PHY modes above 1 $Mbit/s$. Enabling this option means that the PLCP preamble is shorter and that the PLCP header is transmitted at 2 $Mbit/s$, as shown in Figure 4.6.

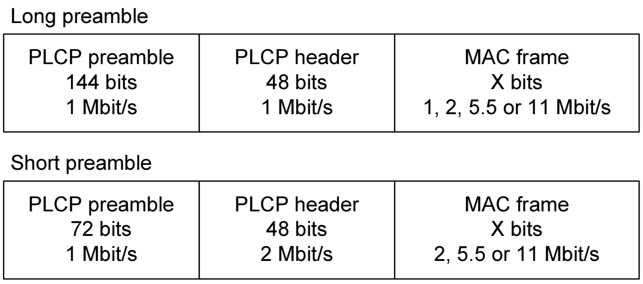

Figure 4.6: *802.11b preamble options from [IEEE, 2003].*

4.3.9 Calculating channel frame error probability

In order to calculate the channel frame error probability, the combined effect of bit errors, which lead to frame errors, and outages is considered. Initially the frame error probability due to bit errors is calculated.

Frame error probability

Under the assumption that interleaving is used to obtain independent bit errors in the Ricean channel, the frame error probability for a frame transmitted using the long preamble option may be obtained as

$$P_{fer} = 1 - (1 - P_{e_{DBPSK}})^{n_{pre}+n_{hdr}} \cdot (1 - P_{e_{Mod}})^{n_{frame}} \tag{4.17}$$

where $P_{e_{DBPSK}}$ is the BER of DBPSK modulation and $P_{e_{Mod}}$ is the BER of the chosen PHY mode, both in a Ricean channel. n_{pre}, n_{hdr} and n_{MAC} are the number of bits in the PLCP preamble, header and MAC frame, respectively.

For the short preamble the PLCP preamble and header must be considered separately

$$P_{fer} = 1 - (1 - P_{e_{DBPSK}})^{n_{pre}} \cdot (1 - P_{e_{DQPSK}})^{n_{hdr}} \cdot (1 - P_{e_{Mod}})^{n_{frame}} \quad (4.18)$$

Combined frame error and outage probability
The probability that a frame is not received successfully depends on the frame error probability as calculated above, but also on the probability that an outage occurs. A frame will be received successfully in cases where an outage does not occur and the frame is not discarded due to a too high frame error probability. Thus, the combined channel error is given as

$$P_{cherr} = 1 - (1 - P_{fer}) \cdot (1 - P_{out}) \quad (4.19)$$

4.3.10 Physical topology

In the overall model the MAC and the broadcast mechanism submodels need information about the amount of nodes within reach. The MAC model needs the average number of nodes within collision range and the broadcast mechanism needs the average number of nodes within transmission range and an average probability of successful transmission within this range, P_{loss}. That is, each node is considered having the same probability of successful transmission within the transmission range.

The number of collision nodes and nodes within transmission range are calculated as the expected number of nodes within each of these ranges given the node density.

For the collision range, the carrier sense threshold is used, since this defines the hearing region of nodes, and affects the number of hidden nodes.

The transmission distance is not trivial to define, since it may be defined as a threshold in terms of many different metrics. The following definitions of the threshold β for defining the transmission range have been considered.

$\mathbf{P_{cherr}}$ threshold: Defining the transmission range as the distance where the avg. channel error probability P_{cherr} drops to the threshold β, would result in the contribution from the PHY model to the aggregate loss P_{loss} would become very dependent in this threshold setting.

Mean power/receive threshold ratio: Another option could be to define the threshold β in terms of the mean received power to receive threshold ratio. This would effectively make the contribution of the outage probability to P_{loss} fixed. However, the P_{cherr} would vary according to the selected PHY mode, since a different noise floor is defined for each PHY mode. Here the FEC code rate would vary the average P_{cherr} contribution to P_{loss}.

SNR threshold: The transmission range threshold β could also be defined in terms of the SNR ratio of the received signal before any FEC coding gain is added. Adjusting the FEC code rate would vary the achieved channel error probability P_{cherr}, which would also be different for each PHY mode.

SNR threshold incl. FEC: Finally, transmission range threshold β be defined as a SNR ratio of the received signal including the FEC coding gain.

The effect would be similar to the option SNR threshold option described above, but instead of varying the channel error probbility P_{cherr}, adjusting the FEC code rate would vary the transmission range.

Each of these options have different effects on the output of the overall model. Initially the first option, which defines β in terms of a fixed channel error probability P_{cherr} is used, since this option guarantees a minimum channel quality for all nodes within the transmission range. The probability of collision P_{col} will affect the aggregate loss probability depending on the amount of collisions due to traffic load and hidden nodes. The other options for β would yield different results, and the definition of this parameter would therefore be an interesting parameter to vary in order to adjust performance of the model.

The physical channel submodel includes both large scale and multi-path fading aspects in modeling the radio propagation in the service case. The model is expected to represent the challenges present in an actual environment as that of the service case, well. The performance of the wireless adapter is considered in relation to the used modulation scheme in a Ricean channel, which accounts for the burstiness of errors in the fading channel. The model currently only supports 1 and 2 $Mbit/s$ PHY modes. Finally the model does not consider interference from collocated nodes.
Since the physical channel submodel captures many details of the conditions in the service case, this requires that the simulation environment in which the DCAD service is considered, accounts for a similar level of detail.

4.4 The medium access submodel

The purpose of this model is to provide a probability of frame loss due to collision. Initially, a brief summary of the 802.11 MAC functionality in the context of the DCAD service considered in this work is given.

4.4.1 802.11 MAC

As mentioned in section 3.1, the hidden node problem may lead to collisions in wireless networks. Since no RTS/CTS scheme is available to provide virtual carrier sensing when broadcasting, the effect of hidden nodes is assumed to be of high importance.

The IEEE 802.11 is a slotted CSMA/CA protocol, which typically uses a binary exponential back-off procedure to reduce the number of collisions. However, since the RTS/CTS or acknowledgment schemes are not active when broadcasting, the back-off procedure is limited to using only a fixed size contention window.
When a node wants to transmit, it senses the channel to determine whether the channel is idle. Next, the node monitors if the channel state remains idle for a DIFS period. If the channel is still idle, the node instantiates a back-off counter with a random number from a uniform distribution in the interval $[0; CW_{min}]$, where the CW_{min} is the initial size of the contention window. At the end of every following idle slot that the node observes, the node decrements its back-off counter. If the medium becomes busy, the node does not decrement the back-off counter but freezes. The node waits until an idle period of duration DIFS has

been observed, after which it resumes decrementing the back-off counter. When the back-off counter reaches 0, the node transmits in the subsequent slot.

Another important aspect of the 802.11 MAC protocol also mentioned in section 3.1 is that when broadcasting, a node need not back-off before forwarding the message if the medium is sensed idle after the broadcast has been received. If multiple nodes have received the same broadcast, they are likely to rebroadcast within the same time slot. Thus, as argued in [Viswanath and Obraczka, 2006] and [Williams and Camp, 2002] a small jitter may be introduced, which delays the scheduling of packets from the network layer to the MAC protocol by a random amount of time. This random spreading effectively reduces the number of collisions, and enables carrier sensing for the rebroadcasts.

4.4.2 Related work

The submodel of the MAC functionality takes a starting point in existing work concerning modeling of the 802.11 MAC functionality and hidden node aspects and aims at evaluating the applicability of the considered models.

A well-referenced piece of work concerning modeling the 802.11 MAC functionality is that of [Bianchi, 2000]. This work presents a Markov Chain based model for accurately calculating the performance of the 802.11 DCF mechanism in saturated conditions. Key assumptions in this work are perfect channel conditions, equal loads, and that all nodes are within hearing range, i.e. there are no hidden nodes. This model accurately takes into account the details of the exponential back-off procedure, and the details of both the RTS/CTS and basic access mechanisms.

The work by [Takagi and Kleinrock, 1984] takes hidden nodes into account in the evaluation of optimal transmission radii for CSMA and ALOHA protocols. This work has been extended by [Wu and Varshney, 1999] who introduce a Markov Chain to calculate the performance of non-persistent CSMA and BTMA protocols, while considering the effect of hidden nodes.

Since the effect of hidden nodes is expected to have a significant influence on the amount of collisions, a starting point for considering this problem is taken in the model by [Wu and Varshney, 1999], while the 802.11 model by [Bianchi, 2000] will be considered subsequently.

4.4.3 CSMA MAC model

The following presents a CSMA MAC model, which is based on the CSMA model in [Wu and Varshney, 1999]. The back-off mechanism in this model is simplified compared to the model by [Bianchi, 2000], since it approximates a slotted non-persistent CSMA. In a non-persistent CSMA protocol, a node that wishes to transmit performs carrier-sensing in order to determine whether the channel is idle. If the channel is idle, the node transmits immediately. If the channel is busy, the node backs off for a random period of time. After this back-off has elapsed, the node repeats the carrier sensing and repeats the back-off procedure if necessary. This back-off procedure differs from the back-off procedure in 802.11 where the back-off counter is frozen and resumed, and this may be a source to inaccuracies in the obtained results.

The model by [Wu and Varshney, 1999] assumes that nodes are distributed according to a two-dimensional Poisson process. To maintain the independence assumption given by the two-dimensional Poisson process, which is necessary for this model to reduce the complexity of the problem, while adhering to the fact that nodes are positioned on a road stretch, the nodes are represented as distributed along a single line according to a Poisson distribution. That is, the inter-node distances are exponentially distributed with average spacing λ and the node positions may be assumed independent.

Assumptions
The model is based on the following assumptions:

Node placement An infinite number of nodes are distributed along a straight line according to a Poisson process with density λ. This placement of nodes represents the motorway defined in the service case. The probability of finding i nodes in a region A is:

$$P[i \text{ nodes in } A] = \frac{exp(-\lambda A)(\lambda A)^i}{i!} \tag{4.20}$$

Frame length All frames for all nodes have equal length T. In the service case, all cars are assumed to transmit messages with a payload of 30 bytes. The frame size is therefore only dependent of the FEC code rate, and is assumed to be equal for all nodes.

Slots The model considers a slotted CSMA protocol where the time line is divided into slots with unit a. This assumption matches the 802.11 protocol, which is also based on slots.

Transmission time Nodes may transmit in the beginning of each slot and a transmission lasts for $\tau = T/a$ slots.

Transmission process When silent, nodes are always trying to transmit with a fixed and equal probability p according to a Bernoulli process. Transmission attempts in idle slots are assumed to be independent. This assumption is justified by the random jitter that is added when a broadcast is forwarded.

Uniform traffic Sender and receiver nodes are chosen uniformly so that all nodes have equal loads. This is a reasonable assumption, since all nodes act as forwarding nodes.

Transmission independence The probability of transmission between any pair of nodes is equal. This is a reasonable assumption, since all nodes act as forwarding nodes and are expected to forward approximately the same amount of broadcasts.

Transmission and collision distance For each node, an equal radius R describes the reach of the node, which is determined by the transmission power used. The node may transmit to or hear nodes within R. Channel conditions are assumed perfect, i.e. there is no path loss, shadowing or multi-path fading. These effects are considered separately in the PHY model.

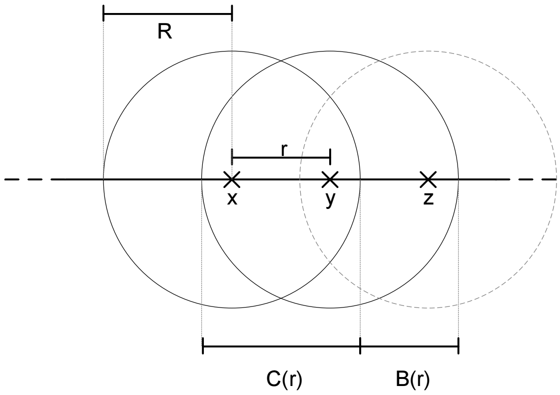

Figure 4.7: *Nodes have a transmission distance R. The common hearing region for nodes x and y that are a distance of r apart, is $C(r)$. The node z is in the hidden node region $B(r)$ of node y.*

Definitions

The considered scenario is depicted in Figure 4.7. The transmitter node denoted x and the receiver node denoted y are placed a distance of r apart on a line, which makes up the road defined in the service case. All nodes have the same transmission power radius R. The hearing region of a node x is given by $N(x)$ and is defined by a distance of R to each side. N is the average number of nodes within the transmission radius, i.e. $N = 2\lambda R$. Within a radius R nodes are able to make a transmission, which may lead to a collision. In the figure, node z represent a hidden node to node x. If node z initiates a transmission that overlaps the transmission from node x, node y experiences a collision. The region in which hidden nodes may exist is denoted $C(r)$ in the figure. This region depends in the distance r between x and y and is given by

$$C(r) = 2R - r \tag{4.21}$$

Specifically, a transmission from a hidden node will collide with the transmission from node x if the hidden node initiates a transmission within the vulnerable period of node x. This period is depicted in Figure 4.8. Since node x is unable to sense whether a hidden node is transmitting, the transmission from a hidden node will collide with the transmission of node x if it falls within the vulnerable period. Here it should be noted that an idle slot is needed between transmissions to account for the propagation delay.

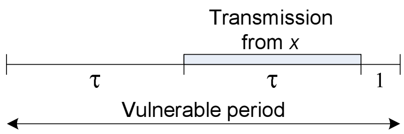

Figure 4.8: *Vulnerable period of transmission from node x.*

In addition to collisions caused by hidden nodes, collisions may also occur if two nodes within hearing range initiate transmissions simultaneously, i.e. within the same slot. The region in which this type of collisions occur is given by $B(r)$ in Figure 4.7. $B(r)$ is given by

$$B(r) = r \tag{4.22}$$

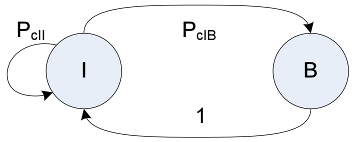

Figure 4.9: *The channel has an idle state I and a busy state B.*

Markov Chain model

The following modeling is based by the approach taken by [Wu and Varshney, 1999], however with the modified assumptions regarding node positions mentioned above. The complete derivation is given in Appendix C.4 on page 157, in the following an overview is given.

The wireless channel around a node may be modeled as a Markov Chain as depicted in Figure 4.9. The duration of the busy state B is the time it takes to transmit a frame $D_B = T$, and the duration of the idle state I is a slot time $D_I = a$. As specified in the list of assumptions, a silent node tries to transmit in each slot with probability p according to a Bernoulli process. Since a node may only transmit if it senses the channel idle, the achieved transmission probability p' is lower than the ready probability p. The achieved transmission probability may be expressed as

$$p' = p \cdot P_{cI} \tag{4.23}$$

where P_{cI} is the limiting probability that the channel is idle. This probability depends on the amount of transmitting nodes within hearing range, and therefore yields

$$p' = p \cdot P_{cI} = \frac{p \cdot a}{a + T(1 - P_{cII})} = \frac{p \cdot a}{a + T(1 - exp(-p'\lambda N))} \tag{4.24}$$

In order to evaluate the performance obtained by a single node, a node is modeled as a three-state Markov Chain as depicted in 4.10. A node may be in the idle state I while it does not transmit. The probability that it does not transmit is $P_{II} = 1 - p'$. The duration of the idle state is $D_I = a$. When a transmission occurs, the node may experience a successful transmission and enter state S or a colliding transmission and enter state C. The duration of a transmission is $D_S = D_C = T$.

The aim of this analysis is to calculate the probability that a receiver node experiences a collision given that a transmission has occurred. To determine this probability the limiting probabilities of the states S and C, P_S and P_C are needed. These may be determined from the transition probability P_{IS}.

The probability that a successful transmission occurs depends on the preconditions described initially. The transition probability $P_{IS}(r)$ may be expressed

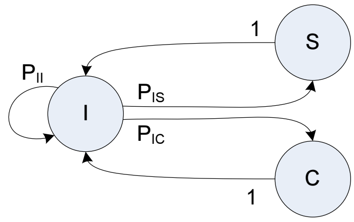

Figure 4.10: *The three states of a node; idle I, successful transmission S and collision C.*

as

$$\begin{aligned} P_{IS}(r) = & P[x \text{ transmits}] \\ & \cdot P[y \text{ does not transmit in the same slot}] \\ & \cdot P[\text{nodes in } C(r) \text{ do not transmit in the same slot}|r] \\ & \cdot P[\text{nodes in } B(r) \text{ do not transmit in vulnerable period}|r] \end{aligned} \tag{4.25}$$

From the Markov chain in Figure 4.10, the following may be obtained

$$P_{IS} = \int_0^R \frac{1}{R} p' \cdot (1-p') \cdot exp(-p'\lambda(2R-r))^1 \cdot exp(-p'\lambda r)^{2\tau+1} dr \tag{4.26}$$

In [Wu and Varshney, 1999] the corresponding expression for P_{IS} is used to derive the throughput of the channel. However, in this work the desired output is the probability that a transmission leads to a collision, that is, the probability that a collision occurs, given that a transmission has occurred. Since the duration of a successful transmission equals the duration of a collided transmission, this collision probability may be obtained from the limiting probabilities of the node being in the successful and collision states, as described in the following.

The steady state probability of the state C may be specified in terms of the transition into state C. Further, from the transitions from state I it is known that $1 = P_{IC} + P_{IS} + P_{II}$. Thus,

$$P(C) = P(I) \cdot P_{IS} = \frac{1}{1+p'}(p' - P_{IS}) \tag{4.27}$$

The limiting probability of state C may be expressed as

$$P_C = \frac{D_C P(C)}{D_C(1-P(I)) + D_I P(I)} = \frac{p' - P_{IS}}{p' + \frac{a}{T}} \tag{4.28}$$

and the limiting probability of state S is

$$P_S = \frac{D_S P(S)}{D_S(1-P(I)) + D_I P(I)} = \frac{P_{IS}}{p' + \frac{a}{T}} \tag{4.29}$$

The probability that a transmission results in a collision may be expressed as

$$P_{col} = \frac{P_C}{P_S + P_C} = \frac{p' - P_{IS}}{p'} \tag{4.30}$$

Model parametrization
The table 4.4.3 presents the parameters used in the CSMA model. The parameter a and $DIFS$ represent the slot length and length of a DIFS period in IEEE 802.11 [IEEE, 1999]. The duration of a transmission T is defined in terms of the frame time provided by the FEC model and the duration of a DIFS period.

Parameter	Value
a	$20\mu s$
T	$t_{frame} + DIFS$
DIFS	$50\mu s$
R	Carrier sense range

Table 4.4: *Parameters used in CSMA MAC model.*

Transmission radius R in the model is defined to be the carrier sense range of a node, since this defines the range in which collisions may occur.

The ready probability p determines the rate at which a node tries to transmit. This probability is clearly related to the broadcast frequency defined in the service case. This relation is however not obvious since the achieved transmission probability p', and thus throughput depends on the utilization of the channel through the relation in eq. (4.23). As a starting point the ready probability will be defined as

$$p = (f_{broadcast} + f_{rebroadcast}) \cdot a \tag{4.31}$$

This definition relates the ready probability linearly to the broadcast and rebroadcast frequency of a single node, under the assumption that these are scheduled independently according to a Bernoulli process with probability p.

4.4.4 802.11 MAC model

As discussed initially, the CSMA MAC model described above considers hidden nodes but assumes a nonpersistent CSMA scheme which differs from the 802.11 CSMA/CA. The model by [Bianchi, 2000] accurately models the details of the 802.11 CSMA/CA scheme, however this model assumes a saturated network of equally loaded nodes, where no hidden nodes exist. In the following these models are compared to determine what effect the differences in assumptions means, and which future improvements could be made. Initially the parts of the 802.11 model that are relevant to obtain the collision probability summarized.

To obtain the probability of a transmission being unsuccessful, it is necessary to determine the probability that a station transmits in a randomly chosen time slot, τ. When considering the 802.11 protocol with binary exponential back-off enabled, it is necessary to solve a nonlinear system of two unknowns. Since the binary exponential back-off procedure uses a fixed size contention window when broadcasting, the probability τ that a node transmits in a randomly chosen slot only depends on the size of the contention window, as

$$\tau = \frac{2}{W+1} \tag{4.32}$$

When a transmission occurs in a network the transmission can be either successful or result in a collision. A key assumption in this model is that collision

occurs with equal and constant probability in each transmission attempt, since a fixed contention window size is used in this work, this assumption may hold well.
When not considering hidden nodes, a transmission is successful if exactly one node transmits in a slot. The probability that a transmission occurring on the channel is successful, is thus given by the probability that exactly one node transmits in the channel conditioned on that at least one transmission occurs [Bianchi, 2000]

$$P_s = \frac{n\tau(1-\tau)^{n-1}}{1-(1-\tau)^n} \tag{4.33}$$

where n is the number of contending stations. Since the only other possible outcome of a transmission is a collision, the probability that a transmission results in a collision is

$$P_{col} = 1 - P_s \tag{4.34}$$

4.4.5 Comparison of models

In the following the collision probability in eq. (4.34) is compared to the results obtained from the previously described CSMA MAC model. Since the 802.11 model considers a saturated network without hidden nodes, the results from here should be considered an upper bound and should be compared to the results from the CSMA model calculated explicitly for $r = 0$, that is when the effect of hidden nodes is not present.

In Figure 4.11 the predicted collision probabilities from the models have been plotted for different numbers of nodes. The contention window size that has been used here is the default minimum of 31 slots. The non-persistent CSMA MAC model has been plotted for an average over r, which results in the effect of hidden nodes to be included, and for $r = 0$, where no hidden nodes are present. Finally the probability of a colliding transmission P_c in eq. (4.34) may be compared to the result obtained from the non-persistent CSMA model for $r = 0$ to be compared. Figure 4.11 shows the collision probability for the range of ready probabilities from 0 to 1, where $p = 1$ represents a fully contended network.

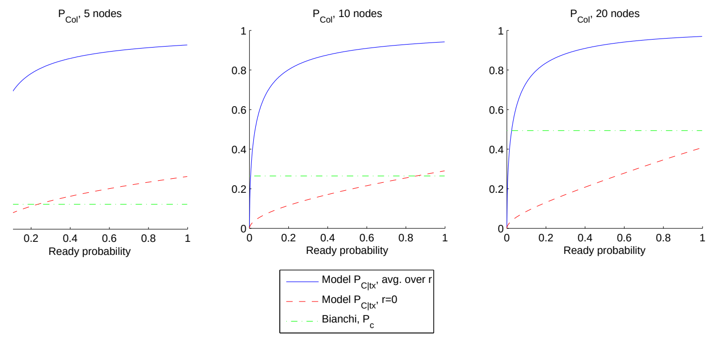

Figure 4.11: *Comparison of collision probability P_{col} for varying ready probability p for MAC model that considers hidden nodes (avg. over r), MAC model without hidden nodes ($r = 0$) and finally P_c calculated from 802.11 model.*

When fully contended, the 802.11 model has a limited number of collisions that depends on the number of contending nodes. This is due to the backoff procedure in 802.11, where a node cannot transmit before it has backed off according to the previously described scheme, and the fact that collisions may only occur when more than one node transmits in the same slot. The same dependency of the number of contending nodes is also seen for the non-persistent model when disregarding the effect of hidden nodes by setting $r = 0$, even though the variation is not as pronounced as for the 802.11 MAC.
When taking hidden nodes into consideration, the non-persistent model predicts that transmissions in a fully contented network will almost solely result in collisions. This behavior is plausible, when considering that hidden nodes are not able to sense the ongoing transmission, and back-off. Instead the collision probability rather depends on the broadcast rate.

4.4.6 Discussion

This section has presented two different approaches for modeling the performance with respect to collisions of the MAC functionality of the 802.11 protocol. The CSMA model that is based on the work by [Wu and Varshney, 1999] has been shown to predict an increased probability of collision when taking into consideration the effect of hidden nodes in comparison to results of the CSMA model without hidden nodes and the 802.11 model by [Bianchi, 2000], which also does not consider hidden nodes. However, the CSMA MAC model is less reactive to varying number of nodes in the network. When adding the hidden nodes aspect to the CSMA MAC model, the broadcast rate becomes the dominating factor with respect to collisions.

The presented non-persistent CSMA and 802.11 MAC models are compared to simulation data in 6 on page 111 and their strengths and weaknesses are discussed.

In order to benefit from the strengths of both the CSMA MAC model and the 802.11 MAC model, here especially the hidden node aspects and the accurate 802.11 MAC model, a solution that combines these models should be considered. Many authors have extended the work by [Bianchi, 2000] in various directions. For example, in [Duffy et al., 2005] the authors consider 802.11 networks in non-saturated conditions where nodes may belong to different traffic classes. However, the aspect of hidden nodes is not covered in this work. Only few authors have considered MAC broadcasting in non-saturated 802.11 networks where the effect of hidden nodes is included. In [Alizadeh-Shabdiz and Subramaniam, 2006] the authors have extended the work by [Bianchi, 2000] in order to model non-saturated networks, while they also have included the effect of hidden nodes, given different carrier-sense, transmission and interference ranges. Such a model could be useful for modeling the MAC functionality accurately, however the model might be too complex to handle in a CL optimization.

4.5 Flooding broadcast model

The final sub-model to consider is the flooding broadcast model based on the *simple broadcast* scheme introduced earlier. Flooding consists of a set of simple rules to propagate broadcast messages throughout the network. This is at the

cost of an excessive overhead. Typically, reliability gains from the increased overhead as there is an increase in the probability that a node successfully receives at least one copy of a message.
In this section the mechanisms of flooding broadcast are analyzed to provide a model definition which can be included in the overall model. The challenge is to analytically describe the flooding process when considering unreliable transmission due to collisions and channel errors. In accordance with the overall model the flooding broadcast model should be capable of describing the probability, P_{mf}, that a specific node will not successfully receive a message.

Currently extensive work has been committed to compare performance measures of different broadcasting schemes. Typically flooding broadcast is used as a baseline for such comparisons as its mechanisms are simple to consider while various practical implementations exist. A majority of the existing work is primarily based on simulation studies. In relation to flooding it seems only few studies consider analytical models. A relevant example is provided in the work of [Viswanath and Obraczka, 2006] where an analytical model for flooding is derived. The authors use the model to calculate the overall reachability of nodes defined by the fraction of nodes that are covered by a broadcast. Their model does not provide an evaluation of message reception probabilities for an individual node. In addition assumptions are made that excludes the effects of some retransmissions in the model. The potential impact on results from these assumptions are not evaluated nor discussed.
In this work the aim is to consider metrics of each node individually which requires a different modeling approach in relation to the studied cases of existing work. To pursue an analytical model, the approach is to consider the individual transmissions of a flooding broadcast in a physical node topology as described in section 4.2 on page 65. The analysis of the broadcast mechanism has been split in two steps: Initially a *fully connected network* is studied. This is based on the assumption that all nodes may transmit messages to each other. This simplifies the analysis as all nodes under the given assumptions can be treated equally. Clearly, a fully connected network assumption does not reflect the actual scenario where many nodes are not within transmission range of each other. Thus an attempt is made to extend the consideration from the fully connected network to analyze flooding broadcasts in a *partially connected network*.
The outcome of the analysis is finally extended to construct a model of the flooding broadcast which can be included in the overall model for optimization analysis.

4.5.1 Developing a fully connected flooding model

As mentioned the fully connected model considers a flooding broadcast environment where all nodes are capable of receiving all transmissions. To enable the analysis of this scenario a set of assumptions are introduced for the fully connected model.

Fully connectedness: All nodes may transmit a message to each other. Thus any transmission in the network will increase the probability of a successful reception for any node. This is clearly not the case in a large topology where some nodes obviously cannot transmit to each other. This assumption is re-considered later in this work.

Equal transmission probability: All nodes experience a similar amount of network contention and channel errors. Thus the expected probability of a successful transmission, P_s, is considered to be the same for any node in the network. This is clearly a rough assumption. For instance transmission probabilities vary with distance, as shown in the PHY model. In addition contention may not be equal throughout the network. Considering this model with varying P_s, e.g. in relation to the transmission distance, may be a future extension to this model.

Independent transmission probabilities: The probability, P_s, of the successful reception of a message in a transmission is independent for each node. This assumption is common in relation to simplify analysis with wireless channels. It may even be a reasonable assumption when wireless losses are caused by effects in the channel of for instance fast fading. However, in the studied environment this assumption may be optimistic. A major part of losses are caused by collisions due to the limited mechanisms for collision avoidance in the ad-hoc scenario. Such losses are typically highly correlated [Tseng et al., 2002]. The assumption of *independent transmission probabilities* does not necessarily simplify the modeling approach as independent losses must be evaluated in relation to each node. In future work, approaches to include correlated losses in the model should be considered. This might include a differentiation between the loss types in the model.

Constant P_s: The flooding broadcast of a message consists of multiple frame transmissions. Due to changing conditions from for instance mobility P_s may in reality vary in time throughout the broadcast operation. However, to reduce the complexity of this modeling approach P_s is considered constant throughout a broadcast.

The aim is to develop a general expression for the message reception probability $1 - P_{mf}$ in a fully connected network. Initially, models with few nodes are considered. Subsequently the considerations are extended to a network of any number of nodes.
Fully connected networks may be defined as in Figure 4.12. Network *(A)* contains five nodes and network *(B)* only contains three. Initially the five-node ad-hoc network in *(A)* is considered. One of the nodes in this network is a *source node* (So). Thus it has a message to broadcast with the probability of 1. The assumption has been made that there is an equal probability, P_s, of a successful transmission between any pair of nodes. Let O define a set containing all nodes in the ad-hoc network *(A)* excluding the source node. In this case $O = \{a,\ b,\ c,\ d\}$. P_{mf} can be derived considering any of the nodes in O. Node 'd' is used in the following example. In Figure 4.12 node 'd' has been denoted *sink* (Si) thus $O = \{a,\ b,\ c,\ Si\}$. A new set is defined $N = O \backslash Si$ which consists of neighbor nodes to Si. The approach is to calculate all potential failed transmission paths between the source and the sink. To distinguish transmissions three categories are defined. (1) The transmission from the source to all nodes

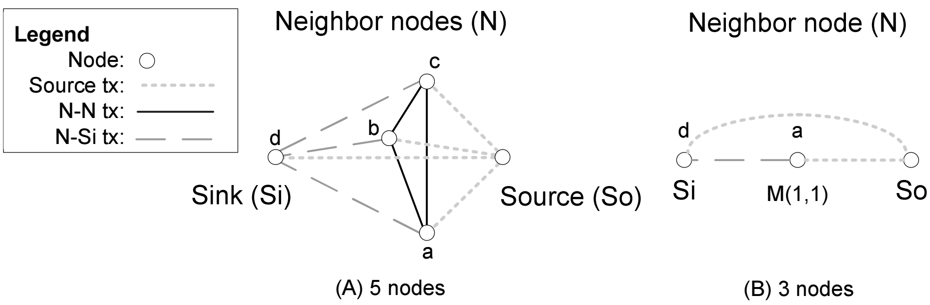

Figure 4.12: *A general view on a fully connected network with one source node.*

in O is denoted *source tx* (2) Transmissions between nodes in N are denoted *N-N tx* and finally (3) transmissions from N to Si are denoted *N-Si tx*.

To proceed the analysis of the problem a simplified network is given in Figure 4.12 *(B)* where $N = \{a\}$. Calculating the probability that all transmission paths to Si will fail is trivial, as given in Equation 4.35.

$$\begin{aligned} P_{mf} &= P_f(So \rightarrow Si)(P_f(So \rightarrow a) + P_s(So \rightarrow a)P_f(a \rightarrow Si)) \\ &= (1 - P_s)[(1 - P_s) + P_s(1 - P_s)] \end{aligned} \tag{4.35}$$

Where $P_f(So \rightarrow Si)$ is the probability that a transmission fails from 'So' to 'Si'. Next a new notation is introduced to ease the expressions: $M(i,\ j)$. i is the amount of transmitting nodes in a transmission and j is the amount of receiving nodes in N. $M(i,\ j)$ defines the probability that a broadcast fails to reach Si when it is relayed through the *neighbor nodes* in N. As a result Equation 4.35 may be re-written to:

$$\begin{aligned} P_{mf} &= P_f(So \rightarrow Si)M(1,\ 1) \\ where\ M(1,\ 1) &= [(1 - P_s) + P_s(1 - P_s)] \end{aligned} \tag{4.36}$$

The following components in the derivation are written in terms of $M(i,\ j)$. The problem in Figure 4.12 *(A)* with 5 nodes is more complex. Multiple transmission paths exist between neighbor nodes depending on which transmissions have failed and succeeded in the *source tx*, *N-N tx* and *N-Si tx*. This is exemplified in Figure 4.13 where two different progresses (α and β) in a flooding broadcast are depicted.

A common starting point is the transmission from So in *tx-round 1* (notice $M(1,\ 3)$). It is always assumed that $P_f(So \rightarrow Si)$ fails since this is a precondition to derive Pmf. Initially it is kept out of the expressions as it is easily included later. In *transmission round 1* some nodes in N either succeed or fail to receive the transmission from So. Nodes who have successfully received the message will commence a retransmission in *tx-round 2*. Recall that failed transmission paths are sought. Thus it is again considered that direct transmissions to Si fail. However nodes in N that failed to receive the transmission in *tx-round 1* now have another chance to receive the message and commence a retransmission. In the progress of α in *tx-round 1* only node 'a' successfully received the transmission. 'a' commences a retransmission which is successfully

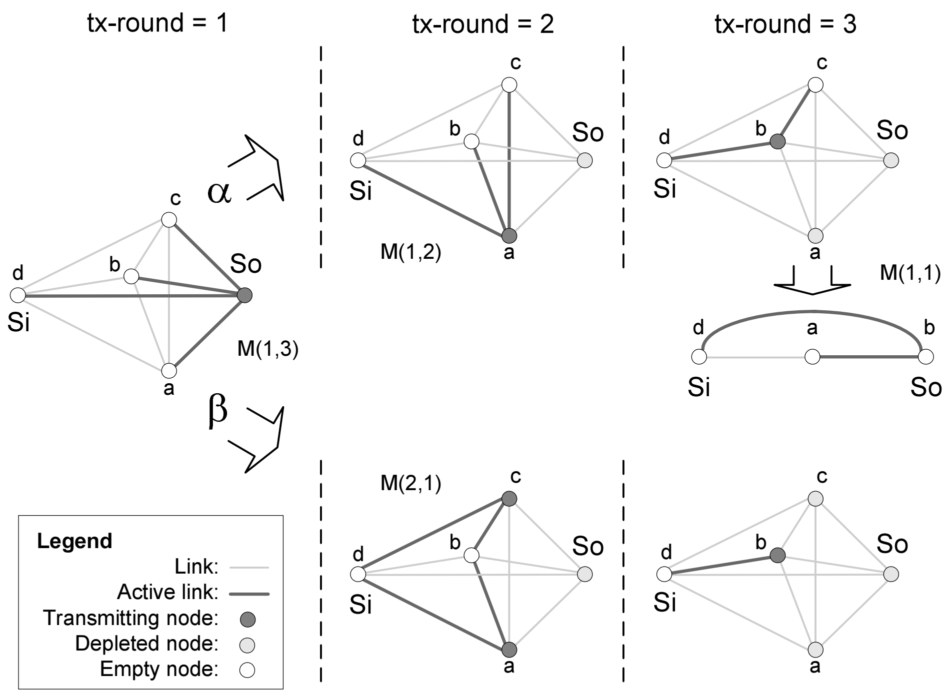

Figure 4.13: *Two different examples of how a broadcast may progress in a fully connected network.*

received at 'b' and fails at 'c' (and Si). Notice in this case $M(1,\ 2)$. In *tx-round 3* 'b' retransmits. Only 'c' has not received a message previously. As the nodes So and 'a' have completed their transmission/re-transmission they no longer represent a path to Si. The result is that the scenario can be considered exactly as in network *(A)* in Figure 4.12 with $M(1,\ 1)$. This re-occurrence of situations will prove useful as shown in the following.
The progress of β differs in *tx-round 2* where two nodes successfully receive the message. This results in a situation where node 'b' may receive transmissions from both 'c' and 'a' i.e. (2, 1).
Based on this understanding of the options of progress $M(1,\ 3)$ can be established.

$$\begin{aligned} M(1,\ 3) \quad &= \begin{pmatrix} 3 \\ 0 \end{pmatrix} P_s^0(1-P_s)^3 + \begin{pmatrix} 3 \\ 1 \end{pmatrix} P_s^1\mathbf{(1-P_s)^1(1-P_s)^2}M(1,\ 2) \\ &+ \begin{pmatrix} 3 \\ 2 \end{pmatrix} P_s^2\mathbf{(1-P_s)^2(1-P_s)^1}M(2,\ 1) + \begin{pmatrix} 3 \\ 3 \end{pmatrix} P_s^3(1-P_s)^3 \end{aligned} \tag{4.37}$$

Considering the second and third terms of Equation 4.37 the bold expression $(\mathbf{1-P_s})$ has deliberately been split in two. First part describes the failed re-transmission to Si. The second part describes the failed transmissions to nodes in N from So.
Notice that Equation 4.37 contains $M(1,\ 2)$ and $M(2,\ 1)$. $M(1,\ 2)$ can be

defined similarly as $M(1,\ 3)$ as a function of $M(1,\ 1)$. $M(2,\ 1)$ is given by:

$$\begin{aligned} M(2,\ 1) &= \begin{pmatrix} 1 \\ 0 \end{pmatrix} (1-(1-P_s)^2)^0(1-P_s)^2 \\ &+ \begin{pmatrix} 1 \\ 1 \end{pmatrix} (1-(1-P_s)^2)^1(1-P_s)^1 \end{aligned} \tag{4.38}$$

where $(1-(1-P_s)^2)$ is the probability that both or either of the transmissions succeed (See β tx-round 2 in Figure 4.13).
Recognizing the recursive elements in Equation 4.37, a general recursive expression for $M(i, j)$ can be defined for any $i,\ j \in N$ in Equation 4.39.

$$M(i,j) = \begin{cases} 1 & \text{for} \quad i=0,\ j>0 \\ 1 & \text{for} \quad i>0,\ j=0, \\ g(i,j) & \qquad otherwise \end{cases} \quad i,\ j \in N \tag{4.39}$$

$$g(i,j) = \sum_{q=0}^{j} \begin{pmatrix} j \\ q \end{pmatrix} (1-P_s)^q \left[1-(1-Ps)^i\right]^q \left[(1-Ps)^i\right]^{j-q} M(q, j-q) \tag{4.40}$$

Finally from Equation 4.41 P_{mf} can be calculated for any amount of nodes in a fully connected network that corresponds to the assumptions initially presented in this section.

$$\begin{aligned} P_{mf} &= P_f(So \rightarrow Si)M(1,\ j) = (1-P_s)M(1,\ j) \\ j &= number\ of\ nodes\ (neighbors)\ in\ \mathrm{N} \end{aligned} \tag{4.41}$$

Notice that $P_f(So \rightarrow Si)$ has been re-introduced.

Model results

To verify the correctness of the fully connected model a simple simulation based on the same assumptions has been created. A starting point is taken in the set O containing all nodes in a fully connected network excluding the *source node*. Further two additional sets are defined:

P Markings of nodes who have received a message.

S Markings of nodes who have not re-transmitted a message.

Both sets have the same size as O. The simulation is described in the following steps:

1. Source tx: Initialize markings in P to *1* with the probability P_s.
2. From P and S nodes who have received a message, and not yet commenced a retransmission, are chosen to transmit next.
3. For each transmission the markings in P and S are updated.
4. Steps 2 and 3 are executed until no new nodes are scheduled to transmit.

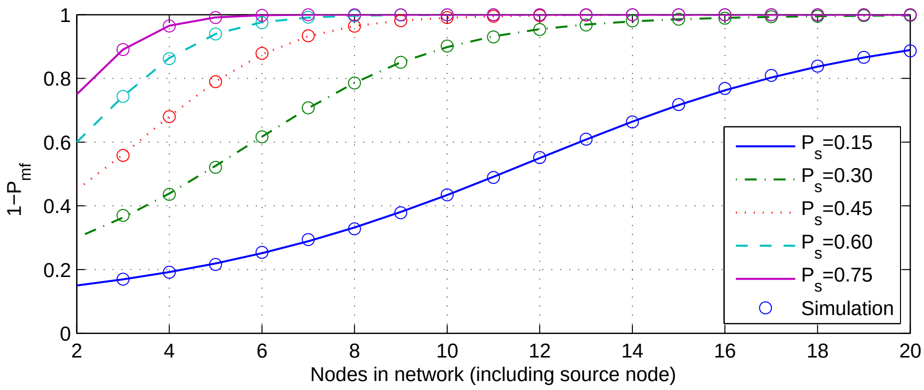

Figure 4.14: *Model and simulation results for varying probabilities of a successful transmission P_s.*

P_{mf} for a given value of P_s is calculated as the mean probability that any given node in O receives at least one transmission. The mean probability is calculated from multiple (>1000) simulation runs.

The results of this simulation and the model in Equation 4.39 are depicted in Figure 4.14. Clearly, the model accurately describes the probability of P_{mf} under the assumption that all nodes can successfully transmit to each other with P_s. Later the assumptions of the model are evaluated in relation to more realistic simulation conditions. However, next step is to move from flooding in a fully connected ad-hoc network to a partially connected ad-hoc network.

4.5.2 Flooding in a partially connected network

Next flooding is introduced in an ad-hoc network where there is not connectivity between all nodes. I.e. the network is partially connected. The corresponds to the studied MANET scenario.

The partially connected flooding model should enable an evaluation of P_{mf} for a particular node. The input for the model is P_s, the amount of reachable nodes in a transmission and a distance between the source node and the particular node being evaluated.

In the partially connected model emphasis is on the physical node placement as described in section 4.2. This enables a simple 1-dimensional topology view where no nodes are in the same location. As described in relation to the overall model the flooding model does not use a notion of physical distance. Instead a transmission radius is defined in the number of nodes that can be reached in average in one direction from a transmission source. This is exemplified in Figure 4.15 for a transmission radius of $R = 2$. The notation R is used for convenience as an abbreviation for the interface name `radius_nodes`. The notation of a source node (So) and a sink node (Si) is used again. From the source, nodes in the *right* section are denoted with positive integers n_{+K}. Nodes in the left section are denoted with negative integers n_{-K}.

The assumptions defined for the fully connected model are maintained for the partially connected model with the exception of fully connectedness. Subsequently approaches to obtain a model for the partially connected network are

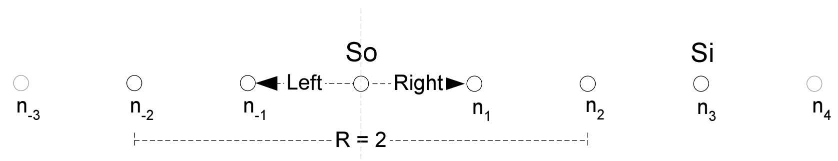

Figure 4.15: *Outline of the node placement. The node indices are defined in relation to the source node So. Si is the sink node.*

considered. The outcome is a model which will enter into the overall model.

A simple case where transmission radius equals one

The problem of modeling the flooding algorithm in a partially connected ad-hoc network may initially be considered in its most simple form with $R = 1$. This case has been depicted in Figure 4.16. A message is propagated in multiple hops in both sections from So. Each node that receives a message successfully will forward it. If a single transmission fails the broadcast is suspended in the particular section. Thus P_{mf} for node $n_{\pm}K$ is simply given as:

$$P_{mf}(K) = 1 - P_s^K \quad K = 1 \ldots U \tag{4.42}$$

Where U defines the number of nodes in each direction from So. With this definition it is possible to assume symmetry for P_{mf}.
It should be noted in this case that message propagation only moves away from the source node. There is a simple reason for this. The source node does not forward transmissions, thus $n_{\pm 1}$ has no influence on transmissions on the opposite side of So. Similarly, a forwarding node only retransmits a message once. Thus, no propagation toward the source node is possible. These two aspects are no longer true when $R > 1$.

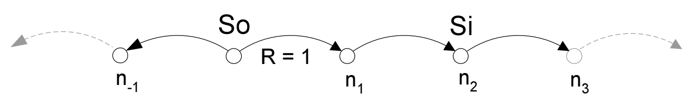

Figure 4.16: *A flooding broadcast in a network where* $R = 1$.

Increasing transmission radius

When the transmission radius is increased several challenges arise in the analytic description of the problem. To propel the further analysis the case where $R = 2$ is considered. Figure 4.17 depicts different example transmission paths for a flooding broadcast initiated at So. The first observation is that messages may propagate in either direction in relation to the source. I.e. nodes in the right and left section of the network may transmit to each other. In addition an arbitrary node may experience a transmission of the same message from different directions. Repercussions from these effects are considered in relation to the model construction.

Similar to the fully connected model it is desired to construct a recursive expression for the calculation of P_{mf} for a particular node K and P_s. Taking

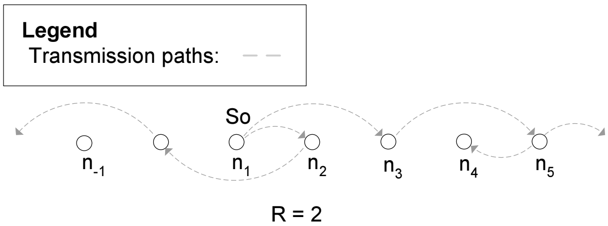

Figure 4.17: *An example of a flooding broadcast in a network where $R = 2$.*

advantage of the symmetry property $n_{+K} = n_{-K}$ only one section needs to be considered initially. A detailed analysis of this case is provided in Appendix C.2 on page 154. It shows that even for $R = 2$ this analysis approach becomes difficult. Further, as cases are considered where $R > 2$ new aspects are introduced. E.g. some of the transmission paths now needs to be considered across Si in both directions. This motivates an approach to consider if reasonable approximations can be found.

An approximative approach

To reduce the complexity of the considered problem it is desirable to consider where independence assumptions could be made. An approach is provided in Figure 4.18 for $R = 2$. This is a slice of five nodes anywhere in the right section.

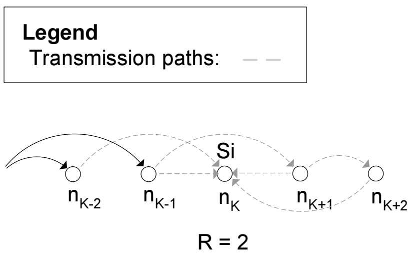

Figure 4.18: *Approximation with $R = 2$.*

Considering n_K as Si only nodes with direct transmission possibilities to Si are included. To enable this perspective it is assumed that n_{K-2} n_{K-1} retransmit messages independent of each other and values $P_{mf}(K-2) \wedge P_{mf}(K-1)$ are known. I.e. $1 - P_{mf}$ is the probability that the particular node will retransmit a message. The independence assumption enables a product form expression as given in Equation 4.43. To include contributions from $n_{K+1} \wedge n_{K+2}$ from Equation C.1, ω is included in the expression in relation to the probability that n_{K-1} receives a message. $P_{mf}(K-1)$ is basically dependent on $P_{mf}(K)$ which is being evaluated. This inevitably leads to inaccuracies. An attempt to compensate for this would be to approximate $P'_{mf}(K-1)$ independent of $P_{mf}(K)$ using the following expression: $P'_{mf}(K-1) = P_{mf}(K-1)\frac{1}{(1-P_{mf}(K))+P_{mf}(K)}$. Clearly $P_{mf}(K)$ is not known. Thus its closest known approximation is $P_{mf}(K-1)$ itself. This leads to the expression of Equation 4.18 which is used in Equation 4.43.

$$
\begin{aligned}
P_{mf}(K) \quad &= \begin{cases} (1-P_s)M(1,\ 2) & \text{for} \quad K=1 \\ h(K) & \text{for} \quad K=1\ldots U \end{cases} \\
h(K) \quad &= [(1-P_{mf}(K-2))(1-P_s)+P_{mf}(K-2)] \\
&\cdot[(1-i(K-1))(1-P_s)\omega+i(K-1)]
\end{aligned}
\tag{4.43}
$$

$$
i(K) = \frac{P_{mf}(K)}{(1-P_{mf}(K))(1-P_s)+P_{mf}(K)} \tag{4.44}
$$

As in the case of the fully connected model a simulation based on the assumptions of the partially connected model has been deviced. The details of this simulation are described in the following section.
This simulation has been used to verify the presented approximation for $R=2$. The results are presented in Figure 4.19.

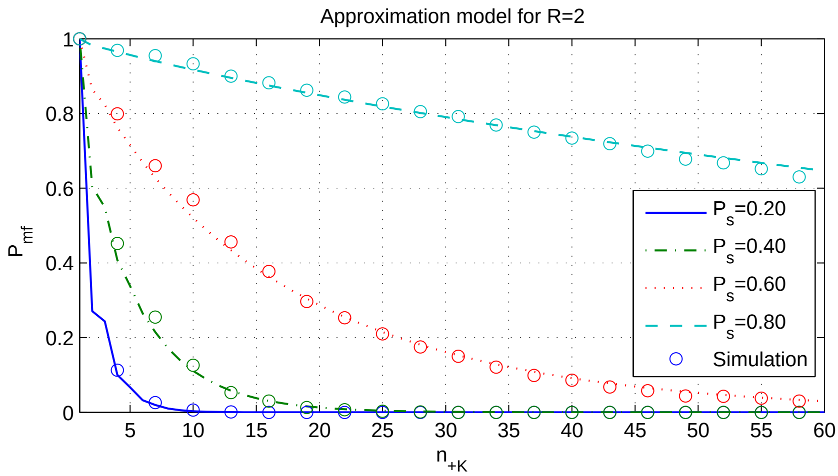

Figure 4.19: *Results from approximation model for flooding in a partially connected network where* $R=2$.

As shown this approximation model provides good results. Unfortunately it may not be very easy to extend for $R>2$. In these cases $n_{k+1}\wedge n_{k+2}\wedge\ldots$ will not only depend on n_{k-1} but also $n_{k-1}\wedge n_{k-2}\wedge\ldots$. This introduces new dependencies and further examinations of which approximations could be useful.

Clearly the presented approaches for modeling the partially connected flooding mechanism introduces some complex problems to solve. Good approximations may be difficult to establish due to the many dependencies. The approximation example however provides a motivation in future work to continue other approximation approaches. An option could be to model small parts of the network as fully connected networks and the combine these. A completely different approach would be to make use of modeling approaches at a higher abstraction level like stochastic petri nets.
In relation to the overall, model requirements are to evaluate P_{mf} for varying

P_s and varying R with $R > 2$. This is also needed, to discuss the applicability of the model approximations made in this work. To enable these capabilities an empirical model is established. This model is presented in the next section.

4.5.3 Empirical model approach

Since the mechanisms of flooding are basic, and the assumptions of our model are well defined, implementing a simulation of the system is simple. Since the conditions for the simulation are fairly basic it is also possible to quickly generate results with a high sample count for many different parameters. These properties enable a simple empirical model approach.

Simulating flooding in a partially connected network
The implemented simulation scheme is an extension of the simulation used to verify the model of flooding in a fully connected network. Still a node is allowed to retransmit a message once under the condition it has received one. The difference is that the transmission reach is now limited to R surrounding neighbor nodes. Again a transmission to a node may succeed with the probability P_s independently for each node in range. The simulation continues until all nodes that have received a message have retransmitted it once.
The simulation scheme is based on a round-based approach taking its starting point in the sets P and S as defined in relation to the simulation of the fully connected network. In a round all nodes who have received a message (and not forwarded it previously) are scheduled to transmit. All these transmissions are completed before P and S are re-evaluated to schedule nodes that will transmit in the subsequent round. In reality this round based approach does not exist. Due to the synthetic jitter, MAC and channel sensing the node retransmissions across nodes are not conducted in the order they have been received. However, it may be reasonable to assume that the time a node re-transmits to some extend is related to when it received the message. Thus, the assumptions of the round-based model approach may not lead to large inaccuracies. This aspect is considered in the verification of the broadcast model in Chapter 6.

An example simulation output of successful mean message reception probabilities based on 2000 independent simulations is provided in Figure 4.20 for $R = 5$. Node 40 is the transmission source i.e. its probability of *having* the message equals one. Around the source node there is an area where probabilities are approximately even due to transmissions from either side of the source. This is in accordance with results from the fully connected model. As the distance from the source increases the probability starts decreasing. The range of the network is limited i.e. node 1 and 80 are *edge nodes*. In the area around the edge nodes $1 - P_{mf}$ decreases slightly faster as fewer nodes exist in the neighborhood to increase the probability of a successful message reception.

Model design
The empirical model is partially based on results from the simulation scheme presented and results from the fully connected model.
The conditions in the area around the source node has a large resemblance to the fully connected network model. The source may transmit to nodes $\{n_{-R}, \ldots, n_{+R}\}$. This range is denoted the *Si neighborhood*. Nodes close to

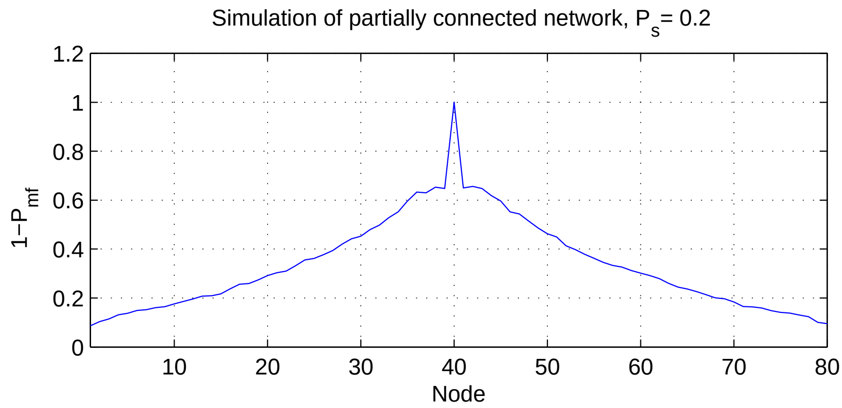

Figure 4.20: *Simulation of a flooding in a partially connected network with $R = 5$. Node 40 is the source of a broadcast.*

Si may also transmit to most of these nodes. Nodes close to the edges defined by $n_{-R} \wedge n_{+R}$ are not in reach of each other. However this is partially compensated from additional transmissions outside the Si neighborhood. These aspects enable an approach where the fully connected flooding model is used to calculate probabilities of the nodes $\{n_{-R}, \ldots, n_{+R}\}$ where $P_{mf} = (1 - P_s)M(1,\ 2R)$. The study of different simulation results has motivated an approach to model nodes outside the Si neighborhood by polynomials. By approximating the progress of $1 - P_{mf}$ in the range $\{n_{+R}, \ldots, n_{+U}\}$ by a polynomial of third degree $y = f(x), y = [0 \ldots 1], x = [1, \ldots, U]$, only four parameters a, b, c, d need to be stored. As a result a model can be created by simulating the flooding mechanism for different parameters of R and P_s, estimating the parameters a, b, c, d, and storing the values in a table. As the curve is approximately symmetric around Si only right side is parameterized. Results may then be mirrored to the left side. R is an integer of limited size, whereas P_s may be defined with some desired *resolution*, e.g. 0.01. The size of the table is consequently: $R \cdot P_s$, which is tractable to work with.

4.5.4 Capabilities of the empirical model approach

The empirical model enables approximate results for probabilities of P_{mf} at a given node. A comparison of the model output and independent simulations is given in Figure 4.21 on the following page for varying transmission radius R. As expected there is a good correlation between the model and the results. The simulations are conducted with $P_s = 0.246$. However, as the model in this case is generated with a resolution of 0.01 it must round to $P_s = 0.25$. In this case the difference seems to have insignificant influence on the deviation.

However this models has some limitations. Primarily the accuracy of this model depends on the resolution of the generated results in relation to P_s. There is a practical limit to this precision. However it is primarily in the generation phase where multiple simulations must be executed. Thus considerations must be made to the desired accuracy of the model when integrating it into the overall model.

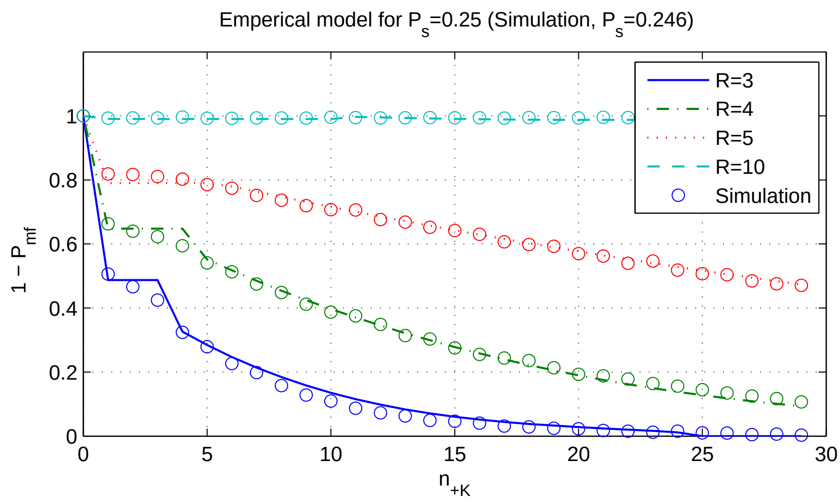

Figure 4.21: *Results of empirical model in relation to independent simulation data. Model is generated at a resolution of* 0.01.

The empirical model does not explicitly consider special cases where either the source node or the destination node lie close to the edge of the network. E.g. if the source node is situated at an edge only R nodes would be reached by its transmission as opposed to $2R$ had it been in the middle of the network. These aspects could be added to improve the overall accuracy and usefulness of this model. A practical approach would be to generate table entries for all special cases. However the table size would increase dramatically. For instance new entries would have to be generated for different distances for a source node from the edge. However, assumptions of fully connected networks in the neighborhood of Si would still be useful. Another approach is to create limited models of these special cases. These limitations are important to consider when evaluating the capabilities of the overall model. Again future work in mathematical models could eliminate these issues.

4.6 Model integration

The previous sections have described the functionality and design of the individual submodels introduced in the overall model design in section 4.2 on page 65. The overall model covers both the message failure probability P_{mf} and the end-to-end delay D_{e2e}. In the presented submodels only the calculation of P_{mf} has been described, while the end-to-end delay calculation is considered in Chapter 6. The model takes a set of layer parameters which may be used to adapt to changing conditions, which are represented by the node density. The model currently supports the layer parameters: *PHY mode*, *transmission power* and *FEC code rate* that may be used to obtain a higher degree of resilience in terms of the message failure probability. The model currently supports the PHY modes for 1 and 2 $Mbit/s$ modes.

In relation to employing the proposed model in CL optimization, a prototype of the overall model has been implemented in Matlab to enable test and evaluation. Based on the node density condition and the layer parameters mentioned above, the message failure probability P_{mf} may be calculated for an arbitrary node, which is defined by its distance from the source node.

In the development of the current model, an important factor has been to avoid complex and computational heavy components in the submodels. Here analytical derivations, approximations and table lookup procedures have been used to obtain efficient submodels. This allows the current model to be used for static search based optimization.

To be able to evaluate the accuracy and performance of the model, output from the model is validated against simulation data in Chapter 6.

Chapter 5

Evaluation framework

The purpose of the evaluation framework is to allow verification and testing of the proposed models, with a realistic representation of the considered service case. Further it allows performance and reliability metrics to be evaluated for simulation cases. Figure 5.1 shows an overview of the three main blocks constituting the evaluation framework.

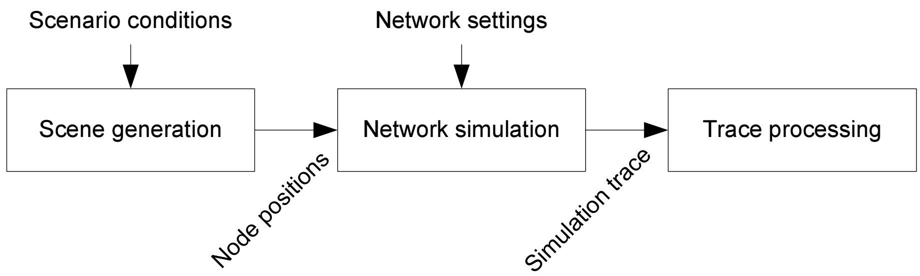

Figure 5.1: *Evaluation framework overview.*

The basis for the evaluation framework is the service case defined in section 2.1 on page 13. The specific road scenario defined in the service case imposes certain requirements and limitations to the mobility and placement of nodes in the scenario. These considerations will be discussed in section 5.1. The *scene generation* block in Figure 5.1 represents the generation of scenarios, which may be defined according to a specific model verification or according the service case scenario.

The second block in Figure 5.1 is the *network simulator*. Based on the *node positions* provided by the *scene generation* block the *network simulator* simulates how communication between nodes takes place. This includes physical phenomena such as radio propagation and interference and the functionality of the protocol stack. As defined in section 2.6 the implementation of the evaluation framework is based on the de facto standard network simulator *ns-2*. This simulator is highly regarded in research communities and is dominant in simulation based analysis of communication methods in ad-hoc scenarios. Being an open source tool with many different contributing parties ns-2 has been thoroughly revised and verified to deliver realistic network simulation results

[Robinson, 2004] [Floyd, 1999]. The use of ns-2 is discussed in detail in section 5.2.

The third and last block in Figure 5.1 denoted *trace processing* represents the extraction of information from ns-2 *simulation trace* files and the processing that is necessary to calculate relevant performance and resilience metrics. This will be discussed further in section 5.4.

5.1 Scene generation

The *scene generation* covers the task of defining the mobility aspects in accordance to the defined service case. To obtain a realistic representation of cars, mobility models which imitate the behavior of real cars could be considered. Appendix D has some considerations on how mobility models could be introduced.
In this work a starting point will be taken in a simplified representation of node positions, where the nodes are statically positioned. The nodes may be positioned equidistantly or according to a Poisson distribution. The equidistant positioning allows tests to be run in a deterministic setting, while nodes positioned according to a Poisson distribution better reflects the variability of node positions on an actual road.

5.2 Network simulator

The network simulator used is in this work is a patched version of ns-2 version 2.29. ns-2 uses the node positions specified in the scene generation to simulate a networked environment, based on a configuration of wireless settings such as transmission power, PHY mode and protocol functionality. Since the nodes' ability to communication depends on the radio propagation conditions in the considered environment, a suitable radio propagation model is needed. In [Takai et al., 2001] the authors show how inappropriate choice of radio propagation models lead to significant differences in the evaluation of ad-hoc routing protocols. The choice of a realistic propagation model is therefore considered in the following.

5.2.1 Radio propagation models

In the service case it is assumed that the nodes are moving on a straight stretch of road in a rural area where only a limited amount of reflections from the surrounding environment occurs. The cars in the scenario are thus assumed to have a direct LoS, however reflections from the road and other vehicles may be present. ns-2 in its standard configuration features two deterministic propagation models, the free space and two-ray ground provided by the CMU Monarch project [Monarch, 1998]. Further a non-deterministic shadowing model is also available. Section 4.3 on page 69 contains a brief introduction radio propagation and propagation models.

In relation to the service case, the CMU two-ray ground model is expected to predict the path-loss best. However, in order to model the multi-path phenomena induced by other vehicles and mobility, a non-deterministic fading model is desired. The built-in shadowing model imitates slow-fading, which is caused by e.g. a large building, however the fading present in the service case is believed to be fast-fading. Since ns-2 does not contain a fast-fading model this has been obtained by an ns-2 extension. The lack of fast-fading models in ns-2 has been addressed by the authors in [Punnoose et al., 2000], who describe how the Rician fading model can be implemented efficiently in a network simulator such as ns-2. They have produced a patch for ns-2 that adds the Rician model to any path-loss model available. This patch is used to obtain fast-fading.

5.2.2 Enhanced 802.11b error model

In the standard ns-2 version 2.29, the successful reception of a frame is determined based on the Signal to Interference plus Noise Ratio (SINR) of the received signal. Since the performance of a wireless adapter, in terms of the achieved BER depends on the used PHY mode, due to the differences in modulation and coding schemes, the BER for the PHY modes should be used to determine whether a frame is received successfully. This has been obtained using the extension by [Xiuchao, 2004], that converts the SINR into a BER depending on the used PHY mode. The extension includes two different tables for converting between SINR and BER. In Appendix E.1 on page 162 these are compared to the model of the physical channel. The values that are used in the evaluation framework, are based on extrapolated measurement results in [PRISM, 2001]. The figure also shows that these values fit rather well to the prediction of the physical layer model presented in section 4.3 on page 69.

In addition to the SINR to BER conversion, this extension has also improved the interference calculation, since the power of all transmitting nodes is considered, in contrast to only the strongest received signal in the standard ns-2 implementation.

5.2.3 Applied ns-2 extensions

In addition to the extensions to the standard ns-2 implementation described above, bug-fixes from [Schmidt-Eisenlohr et al., 2006] related to discordances between the functionality of the ns-2 implementation and the 802.11 standard specification has been included in the ns-2 simulator.

The extensions for the 802.11b error model by [Xiuchao, 2004], the Ricean fast-fading [Punnoose et al., 2000] and the bug fixes by [Schmidt-Eisenlohr et al., 2006] are all included in a patch available at:
`http://www.tlc-networks.polito.it/fiore/`
This patch has been used to obtain these features.

5.2.4 ns-2 802.11b functionality

After having added the Ricean fading and improved error model extensions, the affected parts of the procedure for receiving a frame in ns-2 is as follows:

1. The power level of the received frame is calculated using the deterministic path-loss model, in this work the CMU two-ray propagation model. The Ricean fading gain is multiplied onto the power level of the received frame.

2. If the power level of the received frame is below the carrier sense threshold, the frames is dropped silently.

3. If the power level of the received frame is below the receive threshold, the frames is marked as erroneous. In both cases the frame is forwarded to the 802.11b sublayer layer, which results in the channel being marked as busy for the duration of the frame.

4. The SINR is calculated individually for the 3 parts PLCP preamble, PLCP header and MAC frame, based on the the power level of the received frame, the number of other simultaneous transmissions and the noise floor corresponding to the modulation scheme used for each of the 3 parts.

5. For each part, the BER is determined by table lookup for the relevant modulation scheme.

6. The frame error probability is calculated as

$$P_{fer} = 1 - (1 - P_{pre})^{n_{pre}} (1 - P_{hdr})^{n_{hdr}} (1 - P_{frame})^{n_{frame}}$$

   where n_x is the number of bits in part x, and P_x is the BER of part x.

7. The frame is dropped if P_{fer} exceeds a random uniform number drawn in the range [0..1]. Else the frame is

An important feature in relation to collisions, is that ns-2 uses a capture threshold to determine whether a transmission that overlaps the current transmission causes a collision. The capture threshold is defined as the minimum ratio between the received power level of the initial transmission and the power level of the overlapping transmission. The effect of using this capture threshold, is that the amount of collisions due to hidden nodes may be decreased. ns-2 has this option enabled per default set to a ratio of 10 dB, but during some tests it may be relevant to disable the threshold in order to make the results easier to interpret.

Finally, it should be mentioned that the used ns-2 does not have an implementation of FEC. It is therefore not possible test the effect of FEC by simulation.

5.2.5 Simplified MAC and channel

An accurate channel and MAC layer implementation aims at capturing a majority of the effects in an actual wireless environment with 802.11b. However, the increased complexity also complicates result interpretation and decreases controllability of event in the environment due to the complexity of interacting stochastic processes. To enable a simplified view on MAC and PHY a *simple MAC* scheme is implemented. This allows an intermediate step in tests where a model can be evaluated under controllable conditions.
The simple MAC implementation is based on simple assumptions of independent losses and a common loss probability P_s for any transmission between two nodes within reach of each other. This enables simulation tests in an environment where:

1. Losses are guaranteed to be independent.

2. Loss probabilities are controllable.

To achieve this the original *simple MAC* component of ns-2 has been adapted to provide the desired features. The original simple MAC component offers a basic MAC layer implementation based on a two-ray path-loss model. If receivers are within range of the transmitter perfect channel conditions are assumed. The model implements a simple collision scheme.
To provide the desired features simple MAC has been modified. Instead of assuming perfect channels it incorporates a loss process where loss events are independent. The loss events describe both channel errors and collisions. As a result a single transmission between any pair of nodes is successful with the probability P_s. To enable controllability and independence in the loss process explicit collisions have been disabled.
The simulation tests may now be conducted with either the *simple MAC* or the *802.11b MAC*.

5.3 Flooding broadcast implementation

The flooding broadcast mechanism has been implemented in ns-2 to mimic a layer 3 service. In this layer it is, as discussed earlier, assumed that a node can obtain its own physical position. The flooding broadcast mechanism has been implemented to comply with the functional specifications defined in section 3.1 on page 45. In NS2 context the implemented mechanism is referred to as the *flooding broadcast agent*.

The flooding broadcast agent has two functional properties. It may initiate a message broadcast with a predefined periodic broadcast frequency. Further it handles all received messages and forwards them if either of the following conditions are met:

1. It has not yet forwarded the message with the specific Message Id.

2. It must be located within the zone of the source node.

When a message is received the node uses a sample of its own position and the source position information to calculate the Euclidean distance between the two. If (distance > zone range) the message is dropped.

To enable this operation the relayed broadcast message must contain information about the *source node position*, a *unique message ID* and the *actual payload data*. A unique message ID is readily available in the IP header which includes a 16 bit `Identification` field to enable identification of fragmented packets. The source node position information could additionally be included in the `optional` field of the IP header. These considerations are already included in the assumptions of overall message sizes. Thus they do not affect the total frame size of 92 B including a 30 B payload and IP/UDP headers. A final summary of pre-configured and configurable parameters in the entire simulation environment is given in Table 5.1 on the next page.

5.4 Simulation methodology

In this section the general methodology used to conduct simulation based tests in the ns-2 environment is introduced.
The terminology of simulation based testing is introduced in Figure 5.2.

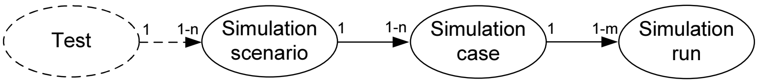

Figure 5.2: *Terminology of tests conducted in the evaluation framework.*

In a *test* a simulation is used to investigate a certain aspect. As an example the aim of a test may be to compare results from the overall model to simulation results. In a test various *simulation scenarios* may be defined where a *simulation scenario* describes a specific configuration of the simulation environment, i.e. density, MAC layer type, PHY mode, etc. In a simulation scenario the desire may be to examine the effects of varying one or more specific parameters like *Tx power* and *BC frequency.* Each examined setting of a set of parameters

Channel and PHY			
Path loss: CMU Two-ray		*Transmission*	
Frequency	2.472 GHz	Power*	-10 ... 30 dBm
Loss factor L	1	PHY mode*	1, 2, 5.5, 11 $Mbit$
Fast fading: Ricean		*Antennas*	
Rice value (K)	6	Elevation	1.5 m
Noise floors		Type	Omni-directional
1 $Mbit$	-104 dBm	Gain	1
2 $Mbit$	-101 dBm	*Thresholds*	
5.5 $Mbit$	-97 dBm	Receive	-75 dBm
11 $Mbit$	-92 dBm	Carrier sense	-85 dBm
		Capture*	off, 10 dB
Media Access Control			
Tfw_{max}	0.5 ms	*802.11 MAC*	
Simple MAC		CW_{min}	31 *slots*
P_s*	$0 \ldots 1$	CW slottime	20 μs
		Preamble*	*Short/Long*
DCAD service			
BC frequency*	$10-50$ Hz	Payload size*	$0-100$ B (30 B)
Zone range	300 m		
Source node*	All/Single(id)		
Conditions			
Node positioning*	*Equidistant /Poisson*	Density*	$1/10 \ldots 1/40$ nodes/m
Road length	1000 m		
**Configurable parameters*			

Table 5.1: *Parameters of the simulation environment.*

leads to the definition of a *simulation case*. A simulation scenario may consist of one or more simulation cases. Finally a simulation case may consist of one or more *simulation runs*. A simulation run is the actual conduction of a simulation. Depending on the type of parameter being tested the simulation run(s) must be adapted in both *duration* and *quantity*. This must ensure that a sufficient amount of independent statistics can be gathered to achieve the desired significance in results. Minimizing the duration and quantity of simulation runs must on the contrary ensure datasets of a tractable size. The actual settings of test runs are specified individually for the conducted simulation cases in the following chapter.

Random seeds in ns-2
The statistical independence is ensured by following the guidelines in the ns-2 manual [Fall and Varadhan, 2002]. Each random variable used in the simulation is based on an independent substream of the *default random generator*. By seeding only the default random generator for each run, the substreams are automatically seeded independently of previous runs.

Trace parsing and statistics
To enable the simulation based testing the evaluation framework incorporates automated simulation and statistics processing tools. These are depicted in Figure 5.3.

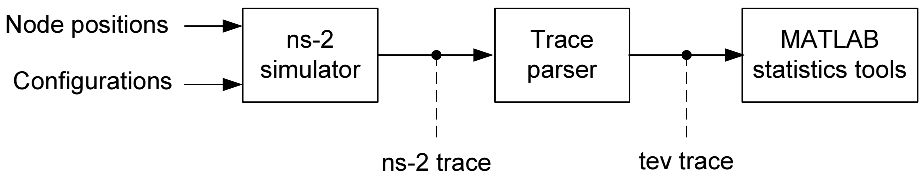

Figure 5.3: *Simulation tools and file formats.*

As described previously a simulation takes it starting point in a simulation scenario defining a given configuration. Based on the configuration and optionally varying parameters in different *simulation cases* a set of simulation runs are conducted. The result is one or more ns-2 trace files which contain dynamic information about *Node positions* and *Transmission events* at layers 1-3. The ns-2 trace files are subsequently parsed in a *trace parser* which sorts out irrelevant events correlates frame transmissions to the messages they convey and creates a new trace format called '*tev*' which can be efficiently parsed in MATLAB. Finally the desired statistics can be calculated in MATLAB. The tests conducted in the subsequent chapter will clarify which metrics can be deduced based on the simulation results.

Chapter 6

Verification and Results

Having established sub-models and a simulation platform the goal is now to analyze optimization problem. Part of this task is to verify model components against simulations to evaluate model assumptions in relation to simulated results. Subsequently, the optimization problem is studied from simulation results and model results. This enables a discussion of model capabilities in relation to how accurately the models matches simulation data and how suited they are for optimization analysis.

In the following the verification and tests are described and the results are discussed. Appendix E on page 162 contains additional test that supports the discussions in the following.

6.1 Verification of sub-models

In order to justify a discussion of model capabilities, the following presents verification tests that establish the correctness and accuracy of the used sub-models. When necessary the sub-model results are compared to simulation results and discussed in relation to reality.

6.1.1 Physical layer sub-model

The first part of this test is to expose any differences between the results of the ns-2 error model and the physical layer sub-model used in this work. Appendix E.1 on page 162 contains comparisons of the path-loss and SNR to bit error probability functionalities.

Second, the complete physical layer sub-model is compared to simulation results. The purpose of this test is three-fold. (1) Determine how accurate the channel error probability P_{cherr} calculation is. (2) Comparison of combined Free space and two-ray model against the Ricean fading model. (3) Exemplify the effect of chosen physical channel parameters.

Test description

Silent listening nodes are placed equidistantly along a line, 5 meters apart to be able to measure the frame loss at different distances. A single transmitter node is positioned at 0 meters, which broadcasts messages at 20 Hz for 5 seconds with a transmission power of 10 mW. All other nodes are silent, which means that there is no interference in this scenario. The ns-2 simulation uses Ricean fading. The frame loss probability is calculated from the number of frames that each node receives.

It is expected that the Ricean fading causes a smoother transition from the states where all or none frames are received, than what would be the result if only the deterministic free space/two-ray model was used.

Figure 6.1 shows simulation and model results for both the free space/two-ray and Ricean models, and model results for the frame error probability P_{fer} due to bit errors and the outage probability P_{out}. These plots are based on the parameters specified in Table E.1.

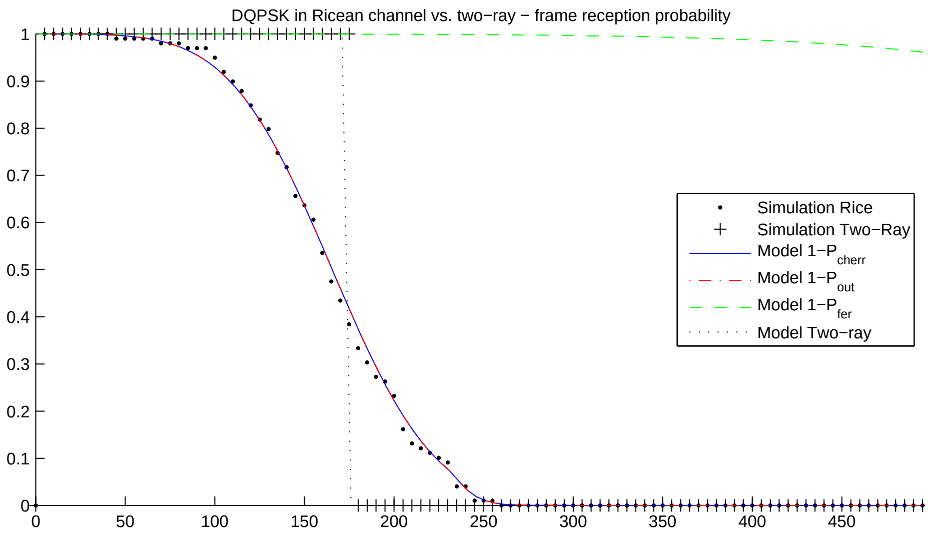

Figure 6.1: *Frame loss probability of model and simulation versus two-ray model, based on* -75 *dBm receive threshold and transmission power of* 10 mW.

Discussion

The plots show a very good resemblance between model and simulation results. The results show as expected that the Ricean propagation model delivers a more random frame error loss than the free space/two-ray model, which reflects the multi-path fading effect. As the P_{out} line is atop of the P_{cherr} line, this shows that frame losses are rarely caused by bit errors. This is also shown by the P_{fer} line that does not start to drop significantly until the outage probability is almost 1.

Figure 6.2 shows a similar plot for the frame loss probability based on model and simulation results. These have been generated for values used in other work, where especially a lower receive threshold of -95 dBm affects performance. The effect of this lower receive threshold is an increased transmission distance, however here the frame loss is not only determined by the effect of outage but to a higher degree caused by bit errors.

The resemblance between the simulation results and the model prediction is less

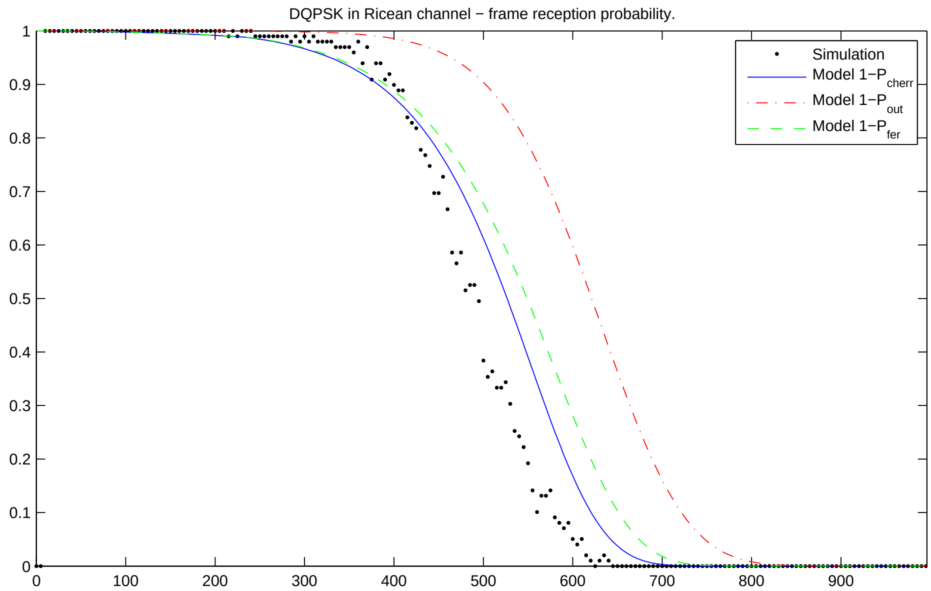

Figure 6.2: *Frame loss probability for model and simulation for receive threshold of* $-95\ dBm$ *and* $10\ mW$ *transmission power.*

precise in this plot than in Figure 6.2. This is possibly due to the difference between the SNR to BER values used in ns-2, shown in Figure E.2 and the analytical expressions for bit error probability used in the model. Figure 6.2 may therefore be seen as a validation of the assumption regarding independent outage and bit error probabilities.
Further, this figure gives a preview of how the model would be able to model the higher rate PHY modes of 5.5 and 11 $Mbit/s$ if suitable model expressions were implemented. At these rates the bit error probability is expected to be dominant as seen in the figure, due to the higher sensitivity towards bit errors, inherent in the high-rate CCK PHY modes.

The physical channel model has shown to accurately model the behavior of frame losses due to channel errors, in terms of outage and bit error probabilities. This test has not included the aspect of interference between multiple transmitters. Since the physical channel sub-model currently does not take this aspect into consideration, this is expected to be a source of deviation.

6.1.2 MAC sub-model

In the MAC sub-model design section two MAC models have been described. The purpose of this test is to determine how well these MAC models compare against simulation results, to determine their applicability as part of the overall optimization model. The test contains two simulation cases, which seek to investigate how the models react to varying transmission power and varying broadcast frequency. However, since the 802.11 model only describes saturated network conditions, it does not vary with the broadcast frequency. Figure 6.3 shows a plot of the collision probabilities for varying transmission power and a broadcast frequency of 150 Hz. Appendix E.2 on page 163 contains detailed test description and additional results for varying transmission powers with a fixed broadcast frequency of 20 Hz and for varying broadcast frequencies.

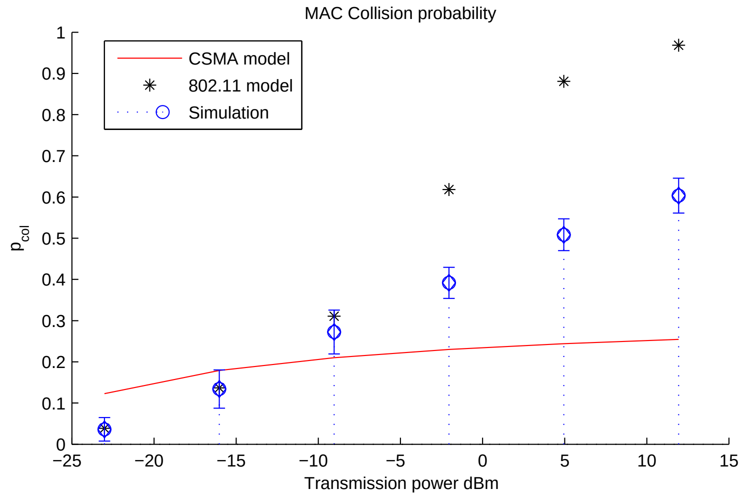

Figure 6.3: *Collision probability for varying transmission power. Based on broadcast frequency* 150 Hz.

Discussion
An important observation in these results is that as the number of contending nodes increases, the fixed contention window size means that nodes start to collide due to the initiation of transmissions in the same slots. Effectively this means that the network may be rendered useless if the collision domain is too large.

For near-saturated network conditions when the number of contending nodes is increased, the 802.11 MAC model overestimates, but seems to capture the course of the collision probability well, which indicates that it would be accurate in a fully contended setting. On the contrary the non-persistent CSMA model does not predict the effect of increasing number of contending nodes well, since it fails to reflect the increase in the amount of collision.

6.1.3 Flooding broadcast sub-model

In this section the flooding broadcast model is verified in relation to ns-2 simulation data. The model, consisting of a fully connected variant and a partially connected variant, has been verified and partially created from a simple simulation scheme with good results. The main aim of this verification is to establish how well the assumptions on which the models were created compare to ns-2 simulation data. The empirical flooding model defined in section 4.5 is used throughout these tests.
This verification consists of two main cases. First the flooding model is compared to simulation data where the simple MAC scheme is used. This model

corresponds to the simple assumption of uncorrelated losses. In this setting the influence of equidistant versus exponentially distributed nodes is tested as well. Finally the simple MAC is interchanged with the more realistic 802.11 MAC scheme where losses are mainly caused by collisions and consequently become increasingly correlated.

Using simple MAC

The broadcast model is tested initially with the simple MAC layer implementation introduced in Chapter 5 on page 104. This implementation uses the same assumptions as the model in terms of independent losses and a common successful transmission probability P_s. This enables a test of the model in the simulation environment where loss conditions are controllable. Further it defines a baseline for comparison with the realistic 802.11 MAC scheme in relation to functional differences between the two and which consequences these differences have in the results. It should be noted that the simple MAC component does not consider realistic MAC access delays and transmission times limiting the analysis options to consider P_{mf}.

Varying reception probabilities and transmission power

In this verification a comparison is established between the model and different simulation cases. In these cases two parameters are varied: (1) the probability of a successful transmission P_s and (2) the transmission power α_1^1. In both cases a predefined density is assumed, thus the transmission power also controls the mean amount of neighbor nodes. In this test cars are distributed equally to ensure that a transmission always reaches the amount of nodes as the flooding model assumes.

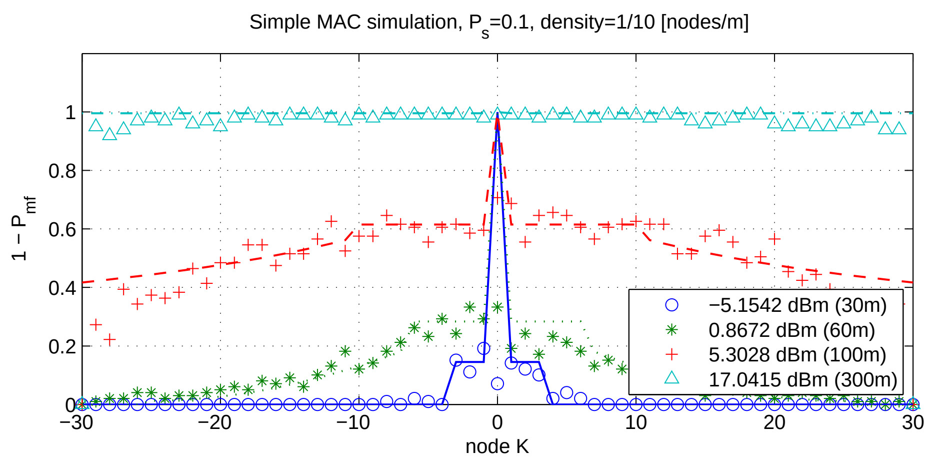

Figure 6.4: *Comparison of flooding broadcast model and simple MAC simulation where $P_s = 0.1$. Node $K = 0$ is the broadcast source.*

In each simulation case only one broadcasting node is considered. It is placed in the middle of the road stretch ensuring that an equal amount of cars exist in either direction. The zone range is 300 m leading to 60 nodes being within range excluding the source node. Only one simulation run is conducted for each test case as there is no variability in the placement of nodes. Each test run is conducted for 5 seconds leading to 100 broadcasts (20/s). It has been verified that a broadcast is completed before a new one is instantiated. In all tests P_s is verified to be as expected. It is further verified that no queue losses exist in either of the tests.

The results are depicted in Figure 6.4 on the preceding page for $P_s = 0.1$. The lines represent model output whereas simulation results are illustrated with points. Clearly there is a good correlation between the model and the simulation result. This generally verifies that the established simulation environment behaves as expected and that the model under the given assumptions is correct. Clearly the model does not capture the drop-off in probability at the edges of the zone. The model assumes more nodes retransmitting in the neighborhood than the limitation of the zone provides. Notice that the probability the source node receives a copy of its own transmission is also included. Similar results for $P_s = 0.2$ are available in the appendix.

Random node locations

Clearly equidistant node locations do not correspond to cars in a stretch of road. To examine the impact of varying distances between the cars the previous verification test has been repeated where nodes are distributed exponentially. As a result there is a variation in the amount of nodes a transmission can reach. This may affect results significantly in relation to the model.
As there is a variation in the distance between cars 30 different simulation runs are conducted with a duration of 3 seconds. With a BC frequency of 20 Hz in total statistics are based on 180 broadcasts.

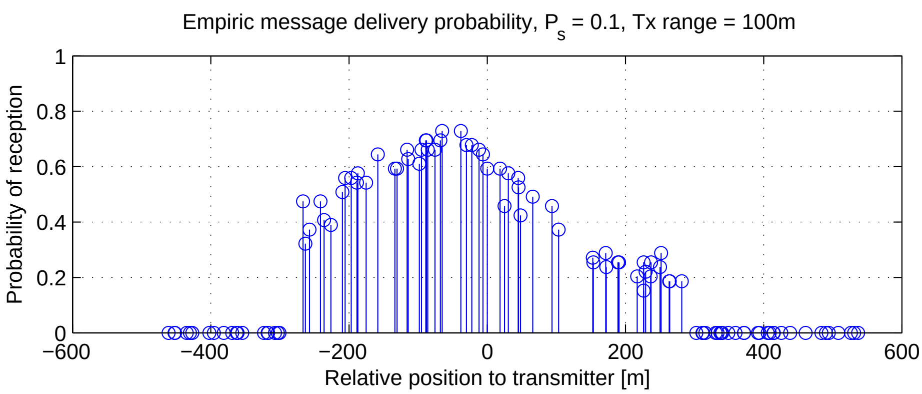

Figure 6.5: *An example distribution of message reception probabilities for nodes exponentially distributed.*

The mean results are depicted in Figure 6.4 on the previous page. Apparently simulation results are a bit lower than in case of the equidistant simulations and the model. This is expectedly due to the effect of the exponentially distributed positions. There is a significant probability that there will be a gap between two subsequent cars reducing the amount of nodes reached by a broadcast. This is shown in Figure 6.5 for a single simulation case. This may have an effect on the efficiency of the optimization approach. As optimization considers the average case for a given density it may deviate significantly from the specific case.

Using 802.11 MAC

The final step in verifying the broadcast model is to use a more realistic MAC layer and channel model. The main difference between this verification test and the previous test, based on the simple MAC layer, is that frame losses are not expected to be independent when using the 802.11 MAC.
Like in the previous test, the broadcast model output P_{mf} is compared. Since the transmission success probability P_s cannot be controlled deterministically in the 802.11 MAC ns-2 component, the simulations are run and processed initially to obtain the P_s parameter. The P_s probability is calculated as an average over all conducted transmissions within a sub-area of the zone. This sub-area does not include the edges of the simulation environment within one transmission radius. This should avoid the inclusion of transmissions from an area where fewer contending nodes exist and P_s is not representative.

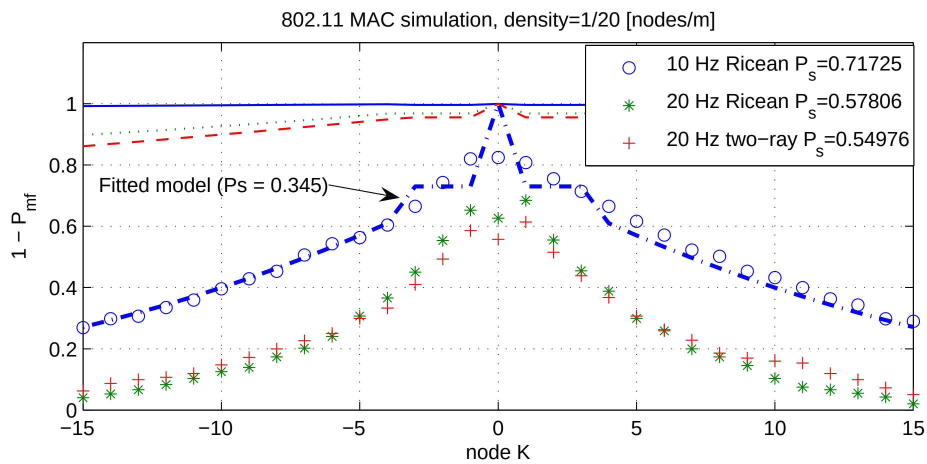

Figure 6.6: *Simulation results of mean message reception probability using 802.11b MAC. Correlation of losses leads to a deviation of the model assumptions with significant results.*

The simulation results using 802.11 MAC are depicted in Figure 6.6. There is a significant difference if the simulation results in relation to the model. Considering the most significant difference between the simple MAC model and the 802.11 MAC is the fact that losses are due to collisions. Specifically correlated losses minimize the probability that a lost transmission to a node, could take another path through the neighbor node. Thus a relatively large p_s is not a guarantee that that a message can reach its destination in this environment. As most of the message losses are caused by collisions at low transmission rates this degrades the model accuracy.

Flooding broadcast model evaluation

Overall a flooding broadcast model has successfully been developed. It corresponds well to assumptions of simple MAC and channel conditions.

In cases where collision losses are dominant, the assumption of independent transmission probabilities causes inaccuracies in the simulation environment with a realistic MAC and channel conditions. As a result the fully connected model underestimates the probability of a failed message delivery P_{mf}. However, the model can be empirically fitted to the results as seen in Figure 6.6. Thus the descending behavior of P_{mf} appears to be consistent, independent of the loss type.

In the RTS/CTS-free MAC transmission scheme collisions are highly likely to occur. Figure 6.7 (A) depicts for a given transmitting node how many of its neighbors lose a message in a single collision. The transmission radius is $R = 3$ meaning that 6 collision losses lead to a complete blocking of the message from this particular node. At 5 collision losses all 6 neighbors will typically not receive the message as the neighbor causing the collision cannot receive it either. Clearly multiple correlated collision losses appear. If this happens early in a broadcast, the message may fail to be received by multiple nodes.

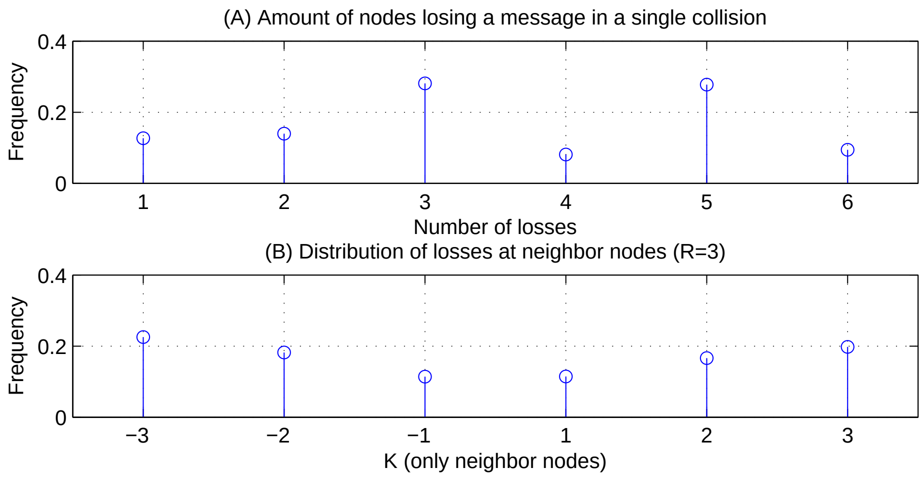

Figure 6.7: *Amount and distribution of collision losses for neighbors of a transmission when* $R = 3$.

Figure 6.7 (B) depicts the distribution of losses in relation to neighbor nodes. Neighbors farthest away from the transmission node have a higher probability of losing a message due to hidden nodes.

6.1.4 Integration

Having tested the sub-models individually, the models are now combined to enable CL optimization. The following discussion summarizes the findings of the sub-model verifications.

In the previous tests, the physical channel model has proven to provide good results. However, since the overall model is based on an average channel error probability for all nodes within transmission range, the accuracy of the model is not fully utilized. This average P_{cherr} assumption means that the overall model overestimates the channel error probability for nodes nearby, while the channel error probability for nodes close to the transmission range are underestimated.

Two MAC models have been presented and evaluated. The verification tests have shown that the non-persistent CSMA model overestimates P_{col} as the broadcast frequency increases. As the simulation saturates at a given level due to the backoff mechanism limiting the collision probability, the CSMA model continues to increase. Further the CSMA model underestimates P_{col} with increasing node count.
The 802.11 model only characterizes a saturated network, and therefore overestimates P_{col} when the level of contention is low. However, since the DCAD service is expected to operate in a well utilized network, the 802.11 MAC model seems applicable, and is used in the following as part of the overall model. However, since the 802.11 model does not take the hidden node aspect into consideration, it may underestimate the amount of collisions in saturated conditions.

The broadcast sub-model overestimates P_{mf} significantly, due to a belief in many redundant paths. An attempt to fit the broadcast sub-model showed that the shape of the model output was similar. Even though the output of the broadcast model is not correctly scaled, the output may still be usable for determining optimal settings.
The broadcast model is further based on the assumption that the number of reachable neighbor nodes is an integer number, which may lead to the overall model output being jagged for low numbers of neighbor nodes, since the effect of an added neighbor node is greatest here.

6.2 Numerical analysis and simulation results

Metrics
In the following tests the primary metrics are given by the reliability metric, message failure probability P_{mf} and the performance metric, end-to-end delay D_{e2e}. As mentioned earlier these metrics define the multi-objective function. Both of these metrics are considered from a the perspective of a single node.

6.2.1 T1 - Simulation of a high contention case

This test is a simulation based analysis of the DCAD service case and the flooding broadcast algorithm. The aim is to establish how the simulation behaves when varying layer parameters and the density. These different simulation cases will enable a discussion of which optimization results can actually be achieved. This is a test of the assumptions on which the design of the optimization approach has been created. Further they provide a basis to compare the overall model output to simulation data. The starting point for this test is a baseline simulation case, which is then used as a reference for the following simulation cases.

Basic test considerations
In order to have reliable simulation results a number of considerations must be made. Firstly, when a network becomes fully contended, nodes want to transmit

more broadcast messages than they are able to actually transmit, and the size of the message queue in each node increases. This unstable condition leads to increasing end-to-end delay as the network becomes congested.

Here, a rate adaptation could be used to limit this effect by ensuring a stable level of delay. This would only have a small influence on the number of collisions, however that would result in the broadcast rate becoming a metric rather than an end-user service parameter.
Besides using rate adaptation, the messages could simply be dropped as the queue length is exceeded. However, for small queue lengths this would mean that frames would be dropped by due to causes that the model does not include. By setting the queue length to a very large value compared to the simulation time, frames are not dropped, and delays increase.
In the following tests the queuing delay has been set to large values, in order to include the congestion effect in the end-to-end delay performance metric.

All simulation run traces have been investigated, to ensure that no occurrences of queue overflows were present. In order to capture the effect of increased delay due to queuing, the tests are run for 5 seconds. Since each run generates a large amount of test data, the number of repetitions of each test has been limited to 5.

For the results between the model and the simulation to be comparable, it is necessary to consider the definition of a failed transmission. In relation to the flooding broadcast model this is simply defined by whether a transmission is successful or not, given the amount of neighbor nodes within reach. However, having based the simulation on a realistic physical layer means that the number of neighbors is not fixed and the definition of a transmission is therefore not as simple. When the transmission power fades below the receive threshold due to the Ricean fading, a transmission cannot succeed. However, it may still cause collisions. Since the physical channel model has been shown to accurately model the outage aspect, the physical channel model is used to correct the probability of a successful transmission P_s in the simulation results, in order to compensate for this effect. This may lead to inaccuracies, however they are expected to be minimal. This compensation does not affect the simulated performance and reliability metrics.

In the following result plots the transmission distances on the horizontal axis of the plots has been estimated using the two-ray model. This means that nodes near the transmission distance experience losses due variations in the received power.

In the following tests a common configuration is used with respect to the following parameters:

Parameter	Value
Broadcast frequency	10 Hz
Propagation model	Ricean (K=6) and free space/two-ray
Capture threshold	off

Baseline simulation case
In this simulation case the transmission power is varied and the the outcome is studied for different nodes. In the evaluation of the results a reference is taken in a specific source node. The performance and reliability metrics are calculated

for 3 receiving nodes, relatively positioned from the specified source node. The first node that is considered is the neighbor node n_1 which is 1 position, i.e. 20 meters from the source. The second is the node n_8 in the middle of the zone range, being 8 positions, i.e. 160 meters away. Finally the node n_{15} at the edge of the zone range, which is 15 positions, i.e. 300 meters away is considered. Considering the metrics for these three nodes allows to test the assumption that the node farthest away from the source is the node that experiences the worst QoS. The specific test parameters may be found in Appendix E.4 on page 171.

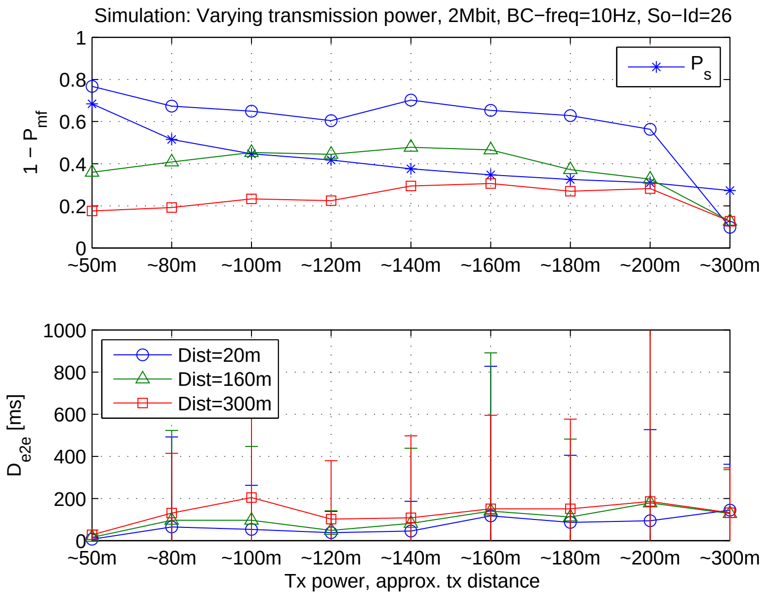

Figure 6.8: *Message reception probability and end-to-end delay for a contended network where the density is* 1/20.

A result from the test where the node positioned in the middle of the considered road stretch is shown in Figure 6.8. For comparison a similar result has been generated for its neighbor node that, like any other node, also provides an instance of the DCAD service. These results may be found in Figure E.8. The results based on these 2 reference nodes vary, but some general tendencies have been observed.

The most important observation in this test is that even with rather conservative broadcast frequency per node of just 10 Hz (the DCAD service is specified in the range $20 - 50$ Hz), the performance and reliability requirements of the DCAD service are not fulfilled. Correlated losses means that even a high P_s value is not sufficient to ensure a successful message delivery, as discussed in the verification results for the broadcast model. The outcome of the test is not expected to improve significantly for higher rate PHY modes, since the limiting factor is are collisions.

Generally, as contention increases, the probability of successful transmission P_s drops due to increasing level of frame losses. However, from 120 m to 180 m

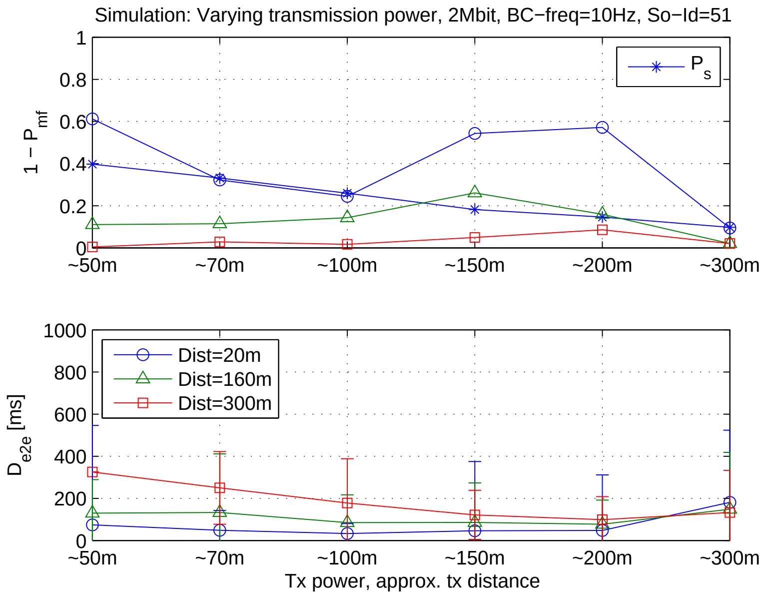

Figure 6.9: *Message reception probability and end-to-end delay for a heavily contended network where the density is* 1/10 *(node 51).*

the advantage of many neighbor nodes seems to outweigh the drawbacks of increased level of contention.
For the neighbor node n_1, it can be seen that it has the highest probability of reception.
The reception probability of node n_8 becomes increasingly better as the transmission distance increases and direct transmission becomes more likely. However, after this the reception probability degrades with increased contention and collision probability.
For node n_{15} the reception probability increases slightly as the transmission distance increases from 100 to 200 meter. Hereafter the increasing level of contention causes drop in message reception probability. That is, direct transmissions are not the best choice for this node. In relation to the assumption of optimizing for node n_{15} in order to achieve the best overall QoS, Figure E.8 on page 172 shows that n_{15} has its optimal point at 200 m, while the two other nodes achieve optimal QoS at 120 m. Thus, this assumption does not seem to hold in general.

With regards to the experienced end-to-end delay, this increases quickly as the transmission distance increases above 50 meters. Further, the nodes farther away from the source node experience the larger delays, as it would be expected. The end-to-end delay metric depends on many factors from queues, contention and collisions. Generally, increased contention leads to increased MAC delays and queuing. However, as the transmission range is increased fewer hops are needed to deliver a message. E.g. at 100 meters there is an increase in delay, but at 120 meters the delay drops again as more nodes are included and fewer hops are needed. Additionally, a higher reception probability should lead to lower mean delay since less transmissions are needed to deliver a message.

When considering P_{mf} and D_{e2e} jointly, it seems that the best QoS, i.e. the highest P_{mf} and lowest D_{e2e}, is achieved with a transmission distance of 140 m. However, this is not completely clear-cut, since Figure E.8 shows that the node at 300 m has its optimal point with relation to P_{mf} at 200 m transmission distance.

However, since the assumption in the chosen optimization approach is to choose the optimal value based on P_{mf} for the node at 300 m, the optimal transmission distance is 160 m based on Figure 6.8 and 200 m based on Figure E.8, which corresponds to transmission powers of 9.39 dBm and 11.32 dBm, respectively. This further shows that there might be some variation between runs for the optimal values.

Increasing density

In this simulation case, the node density is increased from 1/20 $[nodes/m]$ to 1/10 $[nodes/m]$ in comparison to the baseline simulation case. The results of this case are shown in Figure 6.9 on the facing page and E.9 on page 172.

Generally, it can be seen that the level of contention has increased significantly compared to the lower density case above, since the number of nodes within reach for each transmission distance setting has approximately doubled. The successful transmission probability P_s, has dropped in comparison to the baseline simulation case, which has also been expected based on the results from the MAC sub-models. Here it was shown that increasing the number of contending nodes, increases the risk of collisions. Most damage is done to n_15 where nearly no messages can be delivered successfully.

As the transmission range increases node n_1 achieves a significantly increased reception probability. This may be due to the fact that more neighbor nodes may retransmit, even though the probability of collision is high.

With regards to the end-to-end delay, it seems that the level of contention is very high from beginning, which causes high delays, since many hops are required for delivering a message with a short transmission distance. As the number of hops required decreases, so does the end-to-end delay. However, it would be expected that the increasing level of contention would make queue delays larger. Due to the low reception probabilities, the delay plots may actually be based on a very limited data set, and these results should therefore be interpreted cautiously.

In these results the optimal point for the node at 300 m is a transmission distance of 200 m for both plots. However for the node at 160 m the best reception probability is achieved at a transmission distance of 150 m.

Since the optimal values are defined for the node at 300 m, the optimal transmission power is 11.32 dBm.

DCAD end-user service evaluation

The results of the simulation cases above have shown that the DCAD service is very unlikely to function well when based on flooding broadcast since many broadcasts are never delivered, even at low loads. When two nodes within the same neighborhood transmit at the same time, neither of the transmitted frames are likely to be received. If these collisions occur at the broadcast source node this effectively limits the possibility of the broadcast messages to be spread. To be able to prevent these destructive collisions, a feature related to the RTS/CTS scheme used when unicasting, would be extremely helpful. This topic is discussed further in Chapter 7.

Redundancy
The flooding broadcast scheme used in this work, lets all nodes rebroadcast each received message once. This behavior means that the redundancy rate is high, compared to other more selective broadcasting schemes. Figure E.10 shows a plot of the number of copies of each message that a given node receives, for the scenario defined in Table E.7 on page 173. It should be noted that in this scenario only a single broadcast source exists. With an average of approximately 6 deliveries for each node, the network becomes contended much earlier than if for e.g. a CDS-based broadcasting scheme was used. Since contention is a main cause of collisions, the number of redundant messages should be limited. This motivates the use of a more advanced broadcasting scheme in the DCAD service.

6.2.2 T2 - Model verification

The purpose of this test is to determine how well the overall model complies to simulation result from T1 for varying transmission power. In order to do that, the overall model is configured for the two simulation cases tested in T1, and P_{mf} is calculated for varying transmission powers. Since the broadcast model has been shown to be too optimistic since it assumes that losses are not correlated, also P_s is plotted, to enable a comparison of the physical and MAC model outputs. P_{mf} is calculated for the three node positions from T1 at 20 m, 160 m and 300 m.

The physical channel sub-model has shown to be very accurate when interference is not considered. Since the overall model uses a mean value of the channel error probability for all nodes within range, this may lead to slight imprecisions. The MAC sub-model is based on the 802.11 Bianchi model, which has been shown to overestimate the collision probability in near full contention cases, however with capture threshold enabled. Since the considered scenarios are high-contention scenarios, the Bianchi model is expected to deliver a good prediction of the collision probability, possibly a bit low, since this model does not consider the effect of hidden nodes. As the broadcast model assumes uncorrelated frame losses, it is expected that the overall model output P_{mf} is optimistic.

The overall model results are shown in Figure 6.10 and Figure 6.11. Notice, in Figure 6.10 no nodes are within range at 10 m. This is just depicted as $1 - P_{mf} = 1$.

The most important thing to notice is that the model captures that as the transmission power is increased, the increased contention level entails a lower message reception probability due to collisions. As the number of nodes within reach increases, the probability that a frame is received increases. However, due to the optimism of the broadcast sub-model, the overall model predicts this helpful effect earlier than in the simulation results.

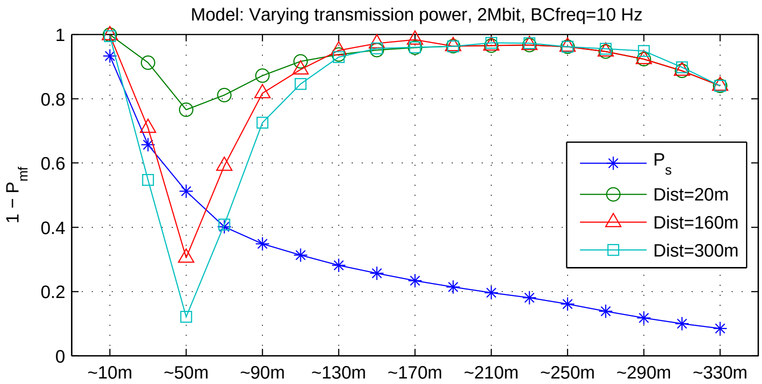

Figure 6.10: *Message reception probability from overall model for contended case where the density is* 1/20.

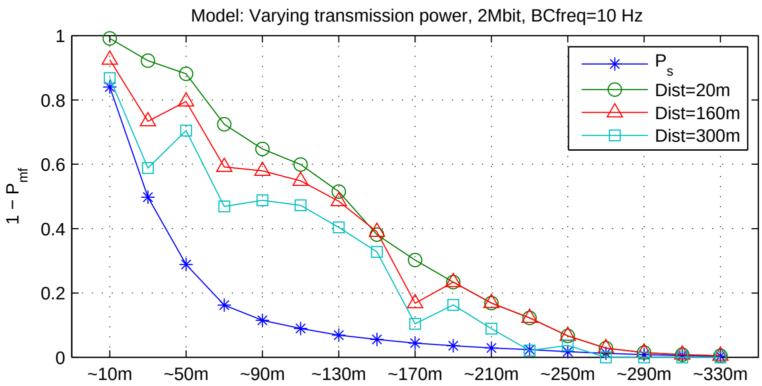

Figure 6.11: *Message reception probability from overall model for heavily contended case where the density is* 1/10.

In the model output, P_s has a wider span than in the simulation results. Initially the probability of a successful reception is higher, however already at a transmission distance of 80 m it has dropped below the P_s obtained from simulation. Here it seems as if the collision probability is overestimated in MAC sub-model.
Further, the model predicts a much lower difference between the reception probabilities of the three considered nodes as the number of nodes within reach increases. This is again due to the assumption of uncorrelated losses in the broadcast sub-model.

The model suggests an optimal point at 210 m corresponding to a transmission power of 11.74 dBm. This result should be compared to the 160 m of the simulation case in Figure 6.9 and 200 m that is the optimal point in the simulation case of figure E.8. Longer and more simulation runs could reduce the variability in the simulation results. However, optimizations of the simulation conduction would be required to make this tractable. These results are still expected to provide reasonable basic comparisons. In this case there is an interesting correlation between the model and simulation results.
This result shows that the overall model may be accurate enough to allow decisions regarding optimal transmission power settings to be made for the considered scenario. In the following the heavily contended case is considered.

In Figure 6.11 the model results of the heavily contended scenario with a density of $\frac{1}{10}$ is depicted. Here it is clearly seen that the increased contention level makes the MAC sub-model overestimate the probability of collision. The result is that P_s quickly drops to a level where not even a high number of neighbor nodes is sufficient for the broadcast mechanism to increase the reception probability.

Besides transmission power, the overall model also allows the PHY mode and FEC code rate to be adjusted to obtain an optimal parameter selection. However, in relation to selecting the optimal PHY mode, the 2 $Mbit/s$ mode is the optimal, since the DQPSK modulation scheme is sufficiently resilient against errors in the considered scenarios. This also means that applying FEC in the considered scenario would not increase the reliability of the system, rather it would decrease the performance since each transmission would take longer time. If the model is extended with the higher rate PHY modes of 5.5 and 11 $Mbit/s$, the FEC code rate parameter would also come into play.

End-to-end delay

In the overall model the end-to-end delay D_{e2e} has been presented as being a function of the mean hop count H and delays present in different parts of the system as

$$D_{e2e} = J_{fw}(H - 1) + (D_{mac} + D_{tx} + D_q)H \tag{6.1}$$

In the following this is considered in relation to simulation results. A scenario has been generated where a single source node broadcast and all other equidistantly placed nodes are acting as forwarding nodes. This enables an isolated view on the end-to-end delay, where the contention level is low. The specific test configuration is given in Table E.7 on page 173.

An approximation of H is provided by the simulation scheme used to create the empiric flooding broadcast model. It is a mean value for the round in which the first reception of a message is registered.
In this low contention case where only a single source node broadcasts at a frequency of 20 Hz, the medium is never busy when a broadcast is initiated. Thus, the source node does not experience a MAC delay. As the neighbor nodes receive the initial broadcast, the level of contention increases and the nodes must back-off for each other, which means that they at least on average must wait for a DIFS period and $\frac{CW_{min}}{2}$ slots before being able to rebroadcast. Further, before a node rebroadcasts, a small random forwarding jitter of $0 - 0.5$ ms is added. Finally the transmission time of a frame, as specified for the DCAD service, transmitted at $2Mbit/s$, takes 0.464 to transmit. Finally, since this is a low contention case where each flooding broadcast dies out before a new is initiated, the queue delay is considered irrelevant and set to zero.

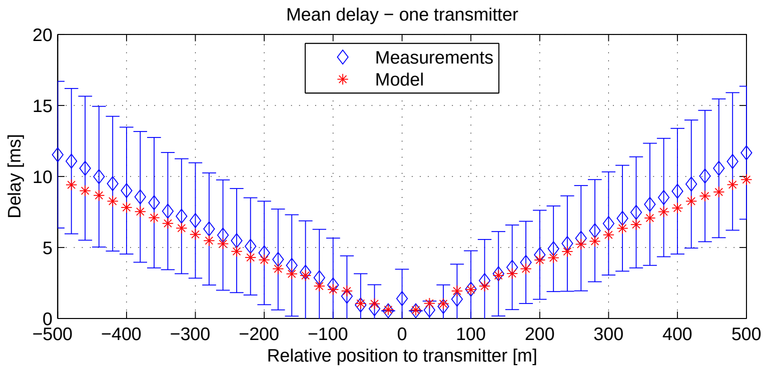

Figure 6.12: *End-to-end delay for different nodes, when a single broadcast source is present.*

In Figure 6.12 the expected delay for a node calculated using eq. (6.1) is plotted together with the calculated average delay for the simulated case. The plots show a high degree of resemblance, however, the model-based delay is slightly lower at far distances than the simulation result. As the calculated MAC delay does not include the busy time a node must wait when backing off, this is a probable cause for the deviation between the plots. Further the may be some queueing delays involved which ar not included in the model.

6.2.3 T3 - Resource consumption

The prototype of the overall model that has been implemented in Matlab. In order to estimate the efficiency of the approach, the overall model has been profiled. It takes approximately 0.5 s to calculate the message reception probability, given specific conditions and configuration of layer parameters, when using the approximative approach for calculation the the bit error rate for DQPSK in a Ricean channel. The part of the model that takes the longest time to calculate is the calculation of the outage probability, since it is defined as an integral of the Ricean pdf, which Matlab needs to approximate in order to solve.
When using the exact calculation of the bit error probability of DQPSK in a

Ricean channel, which is defined as an integral over both the Ricean pdf and the Marcum Q function in the DQPSK expression, the calculation of a single message reception probability takes approximately 1.5 s.
In this context it should be mentioned that $\frac{3}{4}$ of the time spent is used to calculate the transmission distance. This part has not been optimized in any way, and it as expected that the execution time may be decreased by a factor of 3-4 with modest effort, which would mean that each calculation takes approximately 0.1 s.

Chapter 7

Discussion

Cross-Layer resilience optimization provides a holistic approach to jointly improve dependability and performance by providing optimal settings of existing system parameters. In this work the aim has been to consider approaches to apply CLRO optimization to ensure service continuity in an inherently unreliable wireless environment. The starting point has been to consider a mission critical DCAD service in a car-to-car ad-hoc environment. A solution has been proposed to optimize its performance and reliability. In this chapter the properties of the proposed solution are discussed and it is established how the solution may be deployed to improve dependability of service provisioning in wireless environments.

7.1 Resilience in broadcasting

The starting point for the analysis and development of CLRO approaches has been to consider the mission critical DCAD (Distributed Car-Data for Assisted Driving) end-user service. It defines a set of functional requirements for its operation. In addition it also establishes a set of QoS requirements that define a lower boundary to what level of performance and faults it can accept in the application layer level.
The end-user service is established on a multipoint-to-multipoing broadcasting service with geographical constraints given by a broadcast zone. Thus, the focus of this work has been to consider how to improve reliability of broadcasting. Clearly, there is a general challenge of ensuring service provisioning in an unreliable wireless environment. In addition the nature of this service where all nodes broadcast makes this a particularly considerable challenge.
The service is considered in the perspective of readily available COTS systems. This entails a standard 802.11 based wireless network system. In 802.11, layer 2 broadcasts are best effort operations. The only condition to execute a broadcast is that the medium is sensed idle. Adding layer 3 broadcasting capabilities also ensures that messages may be forwarded to nodes outside transmission range. However, currently no standards exist for layer 3 broadcasting and de facto approaches are yet to appear. Thus, much research is currently committed to establishing good methods for providing reliable broadcasting in wireless networks.

The background for this work to study broadcasting in MANETs, has been to review the state of the art in current approaches considering layers 1-3 in context of the CL approach. This provides a new perspective to improve existing broadcast mechanisms.

Optimizing resilience of a basic broadcasting scheme

To enable an analysis of the basic properties of broadcasting a simple layer 3 flooding scheme has been used as a starting point. The flooding broadcast scheme is popular for being easy to implement while providing effective broadcasts with a good coverage. This is at the cost of efficient operation due to a large overhead. However the ease of implementation and its simple assumptions of operation currently makes flooding broadcasting the most prevalent layer 3 broadcasting mechanism deployed in real implementation. This also makes it relevant for analysis in relation to potential cross-layer optimization.
Studying the improvement of flooding broadcast from a CL perspective is readily interesting when considering the basic functionalities of an 802.11 based protocol stack. In this work focus has been on the the following mechanisms. From the bottom and up layer 1 enables *transmission power control* to adjust transmission range and *PHY mode rate adaption* to handle different SNR conditions. Layer 2 enables *FEC* to correct bit errors. Each of these mechanisms are represented by *layer parameters*. Resilience optimization has in this work been defined as the task of finding the optimal settings of these layer parameters. The optimality is finally evaluated in relation to optimization metrics of reliability and performance. For the defined DCAD service, reliability is measured at each end-user node as the probability of a failed reception of a broadcast message, P_{fm}, where performance is the end-to-end delay encountered, D_{e2e}.
The establishment of these resilience mechanisms have enabled the definition of a multi-objective function aiming at ensuring the desired values of P_{fm} and D_{e2e} in relation to QoS requirements.

Performance and reliability of the DCAD service

The DCAD service is a mission critical service. In case of a failure, it may lead to accidents. Thus, strict requirements are set for its operation i.e. a low message loss and delay of messages is required. As expected the multipoint-to-multipoint broadcasting mechanism is capable of causing serious contention and consequently collisions. Particularly a single broadcast may cause all nodes in a zone to retransmit. This leads to $N^2 \cdot \omega[transmissions/s]$ within a zone. However, even at high transmission bit-rates with low broadcast rates ($1 - 2Hz$) ensuring a 100% coverage of transmissions may be difficult. The multiple transmitting nodes will unavoidably lead to collisions. It has been shown how using varying densities to emulate different traffic conditions has a large impact on the performance and reliability metrics.
Optimization analysis has been successfully conducted to study optimal layer parameter settings under these conditions.

7.2 Overview of CLRO approach

An all-important property of introducing CLRO is to allow communication systems to dynamically adapt to their operating conditions. For instance, in case of the broadcast operation it may be useful to adjust transmission range in relation to the density of nodes. Thus, an input to the optimization approach is *observations* of the current conditions. These observations may be available from different layers or from remote nodes. Another input to the optimization process is the *QoS requirements* from the end-user service. The identification of optimal settings of the layer parameters should take all critical requirements into account. This must ensure that the optimization aims at balancing the achievable performance and dependability.

In this work the approach to analyze and provide optimal settings of layer parameters is based on the development of an overall system model. The system model describes how the different system components operate individually and in combination. Having successfully defined a systems model it may be used to analytically identify optimal settings for layer variables. These optimal settings may then be used to specify a joint optimal configuration for the resilience mechanisms. To achieve these capabilities the challenge is to define:

I - A useful system modeling approach.

II - A method that enables optimization analysis.

As discussed later these challenges may be considered separately or jointly. In the following sections the capabilities of the model developed in this work are discussed.

7.3 Models for optimization analysis

The development of the overall model has been based on a sub-model design where system components are modeled individually. Clearly this decreases the problem complexity. In addition, it also enables a starting point in existing modeling approaches for previously studied topics like physical channel and MAC behavior. The outcome of this approach is a set of elaborate models. These models allow a detailed study of the mechanisms that are important to include in the optimization analysis.

7.3.1 PHY and Channel models

The channel is an important part of this model as it represents a basic cause of faults that may occur during operation. Multiple conditions affect the transmission conditions including path loss, multipath fading and interference from other nodes. This affects the energy levels and SNR at receiver stations. The channel model has been consided in close relation to the layer 1 properties of an 802.11b NIC. Aspects have been included to how a channel with fast fading components affects frame loss probabilities and consequently the flooding broadcast operation. This is relevant to enable a joint adaptation of the modulation scheme (PHY mode) and amount of redundant information required for

FEC to handle bit errors. Thus, the channel model has an important impact in relation to choose optimal layer parameters.
From a theoretical perspective the channel model used in this work includes many aspects of signal propagation and its impact on the modulation schemes applied. This has, however, not been verified in a real environment of car-to-car communication. Instead simulation schemes, partially based on the same models, are used. This leads to good model results that ultimately would require real measurements to verify.
The accurate models applied to describe bit error probabilities induce complex integrals that cannot be solved in a closed form expression. In addition they are computationally heavy to solve numerically. This makes them inefficient for dynamic optimizations where new parameters must be calculated continuously. Approximations have been studied and implemented that can reduce complexity. However, these considerations have been limited to DBPSK and DQPSK, 1Mbit and 2MBit respectively. Future work should consider how much impact approximate approaches have on the model precision and the overall optimization analysis. This should particularly include high rate modulation schemes to consider the DCAD service in a realistic deployment with transmission rates in the area of 54 Mbit (802.11a/g) and beyond.

7.3.2 MAC models

Occurrence of collisions is the most significant cause of frame losses in the broadcast environment. The basic channel access mechanisms of 802.11 in broadcast mode provide little protection against collisions. As a result the accuracy of the MAC model determines the accuracy of predicting collision probabilities.
In this work two existing modeling approaches have been adapted. The 802.11 (Bianchi) MAC model which accurately describes 802.11 MAC in highly contended networks and a general non-persistent CSMA model. The latter model in contrast to the Bianchi model includes hidden nodes and varying transmission rates. However its assumption of the backoff process are not completely similar to 802.11. This especially degrades its accuracy when contention is high. In high contention cases however the Bianchi model has shown good results in the context of this work even though it does not consider hidden nodes explicitly. As most of the analysis cases considered in this work are based on high contention cases the Bianchi model has been applied. In future work improvements could be made to join the properties of the two approaches to enable accurate analysis of collision probabilities when contention is low i.e. when the channel transmission rate is increased.

7.3.3 Flooding broadcast model

In this work an effort has been spent on creating a model to analytically describe the mechanisms of flooding broadcast. The model must be capable of producing an estimate of the message reception probability at a given node when considering a given broadcast source.
An accurate model has been derived to define message reception probability in a fully connected network. Extending this approach to a partially connected network has been increasingly complex. As a result the final model of flooding in a partially connected network partly consists the fully connected network

model and partly of an empirical approach. The empirical approach is based on a simple simulation model where the output is fitted to a third degree polynomial with very good results. The parameters of the polynomial are then stored in a table. Many various cases of transmission ranges and successful transmission probabilities can be generated easily. This makes this approach tractable and the models very efficient as only a table lookup and few multiplications are required.
A main assumption in the derivation of the flooding broadcast model has been to consider individual node losses being independent in a transmission. This may be true for frame losses caused by channel errors. However the dominating cause of a frame loss in the broadcast environment is a collision. As collisions are heavily correlated this significantly affects the model results.
The correlated losses mean that multiple redundant paths between the source and sink nodes are blocked concurrently. As a result the advantage of flooding with multiple forwarding nodes is significantly degraded. However there are clear similarities in the model results and simulation results. Thus, it is expected that the flooding model can be developed further to include correlated losses. This may even simplify the analytic modeling approach as multiple transmissions paths can be considered concurrently.

7.3.4 Overall system model

The three sub-models have been integrated in the overall system model to initially provide an estimate of the message failure probability P_{mf}. The model offers adjustment of the following parameters:

- *Layer parameter:* Transmission power
- *Layer parameter:* PHY mode (1 and 2 $Mbit/s$)
- *Layer parameter:* FEC code rate
- *Observation:* Density
- *End-user service parameter:* Payload size
- *End-user service parameter:* Broadcast frequency

Discussing the overall model capabilities can be done in three stages:

Applicability of model - Evaluating the applicability of the proposed model is a question of whether it is suitable for optimization analysis or not. A motivational factor in this work is to establish a dynamic adjustment of layer parameters. This requires a model that can deliver results to calculate new parameter settings quickly as conditions change. If new parameter settings are delayed the results may degrade significantly. The overall model presented in this work is relatively resource requiring, since it uses approximately 0.5 s to compute P_{mf} for a given configuration of parameters and conditions. However, it is expected that it can be optimized significantly.

Qualitative results - The model has been verified in relation to simulation result. In general there is a large deviation in the calculation of P_{mf}. This is mainly due to the inaccurate assumption of independent losses made in broadcast model. However, there is a clear correlation between the model and simulation result in terms of optimal parameters. A future prototype implementation of the model for dynamic adjustments must enable a qualitative analysis of the identified parameters used to improve reliability.

Future improvements - The proposed model and sub-models provide several improvements. As mentioned the optimal parameter calculation can expectedly be improved significantly. In addition the sub-models should be extended to provide en estimate of the end-to-end delay.
An interesting approach in future work is however to use the model knowledge gained in the elaborate modeling approach to pursue other modeling approaches. Specifically Markov Decision Processes have been considered as they enable a joint approach to modeling and the decision process required to identify and set optimal parameters. Such an approach could simplify model complexity.
A general perspective is that observations used in the observation process may be unreliable, i.e. missing or ambiguous, as they are based on unreliable network traffic. Probabilistic model types like Bayesian Networks could be applied to enable optimization decisions under such conditions. These aspects are future work to consider.

7.3.5 Optimization approach

The developed system model does not enable a closed form solution. Thus, the approach to finding optimal parameters is to calculate the model for varying parameters and return the maximum value. This search based analysis becomes increasingly complex as multiple layer variables are adjusted simultaneously. However structured optimization approaches may be applied to improve efficiency.
Several assumptions have been made to realize the CLRO approach. These assumptions are discussed in this section.

Single node optimization to provide global optimization
The definition of the optimization problem has been based on deriving performance and reliability metrics for an outer node placed close to the edge of the geographically bounded broadcast zone. The assumption is made that the node at the edge in average experiences the lowest level of QoS. Thus, if optimization is conducted in relation to this node a global improvement is achieved. In accordance to the flooding broadcast model and measurements of mean end-to-end delays this has been shown to be a reasonable assumption. However, simulation results of P_{mf} have also provided cases where the successful message reception at outer nodes increase while reception probabilities close to the source node degrade. This indicates that this approach will not consistently provide a global improvement of reliability and performance.

Mobility and unevenly distributed cars
A highly relevant discussion in the evaluation of the optimization approach concerns mobility of nodes. Primarily mobility leads to variations in the density of cars motivating a dynamic adjustment of layer parameters. An additional aspect is that cars are not evenly distributed on the stretch of road throughout

a zone. The latter aspect may have a impact on which performance can be expected with the proposed optimization approach.
In this work the differences are analyzed by considering cars evenly distributed with a fixed and constant distance between cars versus cars positioned with exponentially distributed distances. The latter case provides variations in the car locations over varying simulation runs. In average cases the mean results of P_{mf} for the exponentially distributed distances between cars correspond to the evenly distributed case. However the single simulation run may deviate as gaps emerge between some cars while others drive extremely close i.e. besides each other in two lanes. As the optimal parameters are chosen for the average case they may not lead to the optimal results for the specific case.

7.3.6 Results of optimization

As established initially in this chapter, the DCAD service is in the current deployment of the broadcast mechanism not capable of achieving a high level of reliability. Many messages are lost due to correlated collision losses increasing P_{mf} significantly. Still there exist optimal parameter settings maximizing P_{mf}. Identifying optimal parameters by simulation is an inherently tedious task. Simulation times and sizes of datasets define a practical limit to how many cases can be generated. In the model this is not a problem. It can easily be recomputed with varying parameter settings.
Direct comparisons of the model output and simulation results have been made for varying transmission powers. Mainly due to the inaccurate assumption of independent losses in the broadcast model, the overall model predicts P_{mf} too low in comparison to simulation results. Still this does not prevent a model estimate of an optimal transmission distance where the loss probability is minimized.
Results of P_{mf} have been considered in relation to a node at the edge of the zone as the optimization approach defines. The following interesting conclusions can be made:

- Both the model and the simulation results state that a direct transmission to the source is less efficient than reducing the transmission power and relying on multiple hops.

- In both the model and simulation results the needed transmission power must be high enough to reach enough nodes to outweigh the negative influence of high contention.

Based on model verifications, the properties of the sub-models have been clarified. This enables clear explanations in the differences in the model and simulation results. It is expected that the suggested improvements of particularly the MAC model and broadcast model will increase the accuracy of results significantly. Further improving the model with models of modulation schemes for higher transmission rates increases its practical use to provide optimal parameters.

7.4 Providing CLRO to ensure reliable multipoint-to-multipoint broadcasting

In this work models for CL resilience optimization and analysis have been proposed. Next advancement is to include the model in a dynamic optimization approach considering mobility cases. This should also enable an overall evaluation of the reliability of the multiple instances of the DCAD service running in the various mobile nodes.
These steps also entail a consideration of CL architectures where the used layer parameters and observations must be made available to a CL optimization mechanism that makes decisions of optimal parameters. This introduces new issues where coexistence of different end-user services should be considered in the optimization.

Ensuring dependable broadcasts
In the analysis of this work different approaches to increase reliability of broadcasting have been introduced. With the proposed model as a basis adding additional resilience mechanisms can be evaluated. The major improvements considered interesting based on the simulation results in this work are:

CDS based broadcasting - Decreasing the amount of forwarding nodes is an important step to reduce the amount of re-broadcasts and thereby the contention level. A CL approach may be applied to aid the selection of forwarding nodes based on position information and knowledge about link conditions.

RTS/CTS - The lack of an RTS/CTS scheme in 802.11 layer 2 broadcasts means that the destructive collisions from neighbor nodes cannot be avoided. However work has been conducted that proves the possibility of using RTS/CTS schemes in layer 2 broadcasts as well. The price is increases timely overhead for RTS/CTS communication. However, a significant reduction in collisions may compensate for this.

ARQ - Clearly, adding ARQ mechanisms can also be used to improve reliability at the cost of timely overhead.

These resilience mechanisms would introduce new layer parameters and submodels in the overall model. Further the future developments of the broadcast model must be adjusted to comply to the frequency of different loss types as the mentioned schemes are introduced.

Chapter 8

Conclusion and outlook

In the following the conclusion of the project work is presented followed by an outlook on future work. The conclusion is initiated with the achievements made in the project.

8.1 Achievements

Providing dependable services in a MANET environment is a challenge as wireless links are inherently unreliable. In addition, mobility causes constantly changing conditions making fault occurrences frequent as the network topology changes. To improve both performance and dependability in such communication environments, protocol optimization schemes are subject for much research. Recent studies challenge the traditional approach that optimizations should be contained in the different layers of the protocol stack. By adding joint approaches across layers, potential has been revealed in improving both service performance and dependability.
In this work a Cross-Layer resilience optimization approach has been studied to enable dependable services in car-to-car scenarios.

Defining a mission critical end-user service case
A part of the service types expected to appear in future deployments of car-to-car networks are mission critical. To consider these aspects a DCAD (Distributed Car-data for Assisted Driving) service has been specified. This service enables e.g. car platooning and evasive maneuvering. The service is based on cars propagating their information about position and velocity to other cars within a geographically limited area called a *zone*. It defines strict requirements for QoS that must be kept to avoid accidents. In this work QoS have been considered in relation to two metrics (1) the performance metric *end-to-end delay* (D_{e2e}) and (2) the reliability metric *message loss probability* (P_{mf}).
Based on this service definition a requirement of a reliable multipoint-to-multipoint broadcast mechanism has been established.

Resilience mechanisms in ad-hoc broadcasting
Standardized or de facto standards for reliable broadcasting in 802.11 based ad-hoc networks does currently not exist. An analysis has been conducted to

establish the current state of the art in both layer 2 and layer 3 broadcasting mechanisms that may deliver efficient and/or reliable broadcasting. Results of the analysis have been used to establish a simple flooding based broadcast scheme with geocasting capabilities. It has been based on a COTS protocol stack with assumptions of HIDENETS middleware. The scheme defines an entry point to analyze CL optimization of multipoint-to-multipoint broadcasting.
Based on the COTS protocol stack and the functional properties of broadcasting, mechanisms have been identified that can be optimized to improve resilience toward faults. They encompass transmission power control, FEC code rate and PHY mode rate control.
The control variables for the resilience mechanisms have been defined as *layer parameters*. In conjunction with the performance and reliability metrics finding optimal settings of these *layer parameters* defines the aims of optimization. These aims have been formulated in a multi-objective function enabling a joint optimization of both performance and reliability.

Optimization approach and system model
Based on the multi-objective function an optimization method has been specified that aims at improving the metrics of optimization globally across nodes in a zone. This has been conducted in the context of dynamic adjustments where the optimal layer parameter setting are continually sought. This enables an adaptation of the broadcasting process to varying conditions of node densities in the zone.

To enable the optimization, a model based approach has been devised. It relies on a system model where the parameter space can be searched to find an optimal setting of the layer parameters. The input to the optimization process is an observation of node density and knowledge about the DCAD end-user service parameters *broadcast payload size* and *broadcast frequency*.

A system model has been designed that entails a sub-model based approach. Each sub-model individually represents the functionalities of respectively the channel and PHY layer, MAC and the broadcast mechanism. This has allowed a decoupled model design approach where suitable model types have been applied to each problem. The outcome is a set of elaborate models that provide useful insight into the studied optimization problem. The models have successfully been integrated to deliver performance and reliability metrics enabling search based optimization of optimal settings for the layer variables.
The sub-models represent the following achievements.

PHY and channel model - This model encapsulates the interaction of the wireless channel and the physical properties of the communicating NICs at layer 1. It is based on a combination of existing models to describe bit error rates (BER) and frame losses as an outcome of the applied modulation scheme and a Rician channel. Approximation approaches have additionally been applied to reduce computational times of the models.
A model of FEC has been included to derive an approximation of the BER in relation to a given code rate.
The model has been verified to successfully predict frame error probabilities via simulation.

MAC model - Two different MAC models, *general CSMA* and *Bianchi 802.11* with different properties, have been applied to provide an estimate of collision probabilities under varying conditions loads and amount of nodes in the collision domain. The models have been adapted from existing work to comply to the properties of layer 2 broadcast transmissions. Both models have been successfully implemented in the system model to enable an assessment of their accuracy for varying conditions.

Flooding broadcast model - To asses both P_{mf} and D_{e2e} for particular nodes in the MANET environment, a model is required that incorporates the functional properties of layer 3 broadcasting. In this work both an analytical and an empiric model has been proposed. The analytical model accurately describes P_{mf} for nodes in a fully connected ad-hoc network under the assumption of independent frame loss probabilities. The analytical model has been included as a part of an empirical model to deliver an estimate of P_{mf} in a partially connected network. The empirical model has been obtained from a basic simulation scheme and is based on parametrization of polynomials.

Evaluation framework
An evaluation framework has been established. It provides simulation and statistical tools to enable an evaluation of the model accuracy and analysis of the optimal layer parameter settings in a deployment of the DCAD end-user service. The network simulation is conducted in ns-2. However, ns-2 is originally based on simple assumptions of channels. Thus ns-2 has been modified using existing patches and own modifications to deliver Rician channel properties, channel errors on a bit level, correct 802.11 MAC behavior and the needed trace information.

Model implementation and results
Finally the model has been implemented to enter into the evaluation framework. The outputs have been verified in relation to simulation results and basic optimization analysis cases have been successfully conducted providing useful results.

8.2 Conclusion

Provisioning of highly dependable services in unreliable wireless environments with mobile nodes is a key motivation to study CL optimization approaches. The starting point of this work has been to consider a relevant service case. The functional properties of the DCAD end-user service specify attractive features in future car-to-car scenarios. However, it also emphasizes the challenges present in enabling such a service.

Introducing a broadcast scheme
Relying on multipoint-to-multipoint broadcasts the DCAD service represents the worst use case scenario imaginable in a wireless 802.11 based ad-hoc network. Primarily, no RTS/CTS is available when layer 2 broadcasts are exe-

cuted. This increases the probability of collisions significantly. Secondarily, the dynamic widespread topologies of ad-hoc networks means that hidden node problems are very common.
A basic broadcast setup has been used as a starting point to consider these aspects. It implies a layer 3 flooding broadcast scheme where every node retransmits a received message once. It induces much redundancy which increases probabilities of message delivery at the cost of an increased overhead.

Functional mechanisms and improvement of resilience
The fundamental basis for CL resilience optimization in this work has been to consider the functional mechanisms of a COTS protocol stack including the flooding broadcast mechanism and a HIDENETS middleware layer. This implies 802.11(b) PHY and MAC functionalities.

From this protocol stack a set of mechanisms have been identified that affect the performance and reliability of the considered multipoint-to-multipoint broadcast. They are referred to as *resilience mechanisms* as they either actively or passively affect the capabilities of a node to prevent and handle occurring faults.
In layer 1 the PHY mode offers an adjustment of the transmission rate to SNR conditions to minimize bit errors. In the same layer, transmission power control can increase the SNR and transmission range but also increase the collision domain. In layer 2 FEC enables correction of bit errors at the cost of increased frame sizes and higher transmission delays. The interface to each of these mechanisms are represented by a *layer parameter*.
The capabilities of the mechanisms have been studied to enable a joint CL optimization in relation to QoS requirements from the end-user service and the varying conditions for nodes operating in a MANET. The optimization aims have been considered in relation to optimization metrics defined by the DCAD service. D_{e2e} is a performance metric describing the end-to-end delay between the broadcast source and a service user. P_{mf} is a probability that a broadcast message is not received at the service user. A multi-objective function has been defined to seek a minimization of both D_{e2e} and P_{mf} depending on settings of the layer parameters, observed operating conditions and end-user service parameters.

Introducing a system model
To realize the CL optimization a model-based approach has been applied. It is based on an overall system model that encompasses the functionalities of the studied resilience mechanisms. The system model has been partitioned in three sub-models to enable a decoupling of the model development. This has allowed different modeling methods to be applied in conjunction with the type of mechanism being modeled.

Modeling PHY and channel behavior
The *PHY* model describes the interaction between the wireless NIC and the channel to describe bit and frame errors in relation to a given channel condition. This sub-model is responsible for supplying the frame error probability P_{fer}, which is related to bit errors and outage probability P_{out}, which is the

probability that the received signal power level drops below a certain threshold. With a starting point in the channel properties, a deterministic combined free space and two-ray path-loss model is used to model the large scale path loss. Further, Ricean fading has been used to model the multi-path effects caused by the surrounding environment. To capture the effect the Ricean channel has on the wireless NIC, expressions for calculating the BER in a Ricean channel for the 1 and 2 $Mbit/s$ modulation schemes has been sought. In order to make the model computationally tractable, closed form solutions have been desired. For the 1 $Mbit/s$ mode, an analytical expression has been derived, which enables an efficient calculation of the bit error probability in a Ricean channel. However, in the case of the 2 $Mbit/s$ mode, this is not easily done, and an approximate solution is used for calculating the bit error probability. The 5.5 and 11 $Mbit/s$ modes are even more complex and are currently not supported by the model.
A simple FEC scheme based on Reed-Solomon coding is included in the model for coping with channel conditions where bit errors are limiting the transmission distance.
The physical channel model has been validated against simulation results and has shown to predict performance very well.

802.11 MAC models

The MAC sub-model describes the interaction between nodes in the wireless channel on the MAC layer. The considered aspect is the collision probability P_{col}, which is in general based on the number of nodes within collision distance. Two MAC models have been considered in this work. A general CSMA model that particularly includes hidden nodes and varying transmission rates, as opposed to the 802.11 MAC (Bianchi) model that considers only saturated network conditions and does not take hidden nodes into consideration.
In verifications the general CSMA model has been found not to represent the results obtained by simulation very well, since it was not very sensitive to the number of nodes within reach, but rather dependent on the transmission frequency. The 802.11 MAC (Bianchi) model was found to match the tendency in the simulation results better, since it accounts for the fixed size contention window back-off used when broadcasting in 802.11. The 802.11 Bianchi model has been used in the overall model evaluation, since the considered DCAD service is expected to operate at a near saturation level, at least while it is coupled with the flooding broadcast mechanism.

Flooding broadcast model

The flooding broadcast model defines the properties of the flooding broadcast mechanism to deliver the optimization metrics of D_{e2e} and P_{mf}. It is based on a frame transmission failure probability P_{loss} aggregated from collision losses and channel losses. Further it uses topology knowledge from the PHY model to establish the mean amount of nodes that are within range of a single transmission.
The flooding broadcast model consists of both an analytic model and an empiric model. The analytic model has been developed to describe P_{mf} under the assumptions of a fully connected network where P_{loss} is equal for each node and losses in a broadcast transmission are independent between nodes. By simulation analysis this model has been found to accurately describe P_{mf} under

the given assumptions. Deriving an analytical model for flooding in a partially connected network has been found increasingly difficult. Instead, an empirical model has been developed. It uses a basic simulation scheme to generate an estimate of P_{mf}. The properties of the flooding broadcast scheme results enables a parameterization of polynomials to store results efficiently. Joining the analytical and empirical models results in a flooding broadcast, verified independently in ns-2 with good results using simple MAC assumptions.
Besides providing an estimate of P_{mf}, the basic simulation scheme has also been successfully used to derive a mean hop count path H and enable useful approximations of P_{e2e}.

Although good models of flooding broadcasting have been achieved, the assumption of independent losses is not accurate. Independent losses may reasonably be assumed from channel losses. However, collisions cause heavily correlated losses. In relation to flooding broadcast this means that the amount of redundant paths are reduced significantly. In a broadcast scenario without RTS/CTS schemes this problem is specifically pronounced as neighbor nodes are capable of eliminating each others broadcast completely. A simulation study has been conducted where the mean transmission range is three nodes ($R = 3$). In 10% of the collision cases none of the 6 neighbors receive a transmission. In additionally 30% of the cases only 5 of the neighbors receive the message. In these cases the last neighbor is the cause of the collision, thus no retransmission is commenced.
These differences in the model assumptions to the simulation scenario leads to a model that is too optimistic in its predictions of P_{mf}. It is however believed that a differentiation of loss types can be included in the flooding broadcast model. An assumption of correlated losses may even lead to some simplifications in construction of the analytical model. This must be considered in future work.

DCAD service capabilities

The capabilities of running the DCAD service in multiple nodes have been studied in simulation analysis independently of the optimization approach. This provides an idea of what performance and reliability can be expected with the studied broadcast scheme and available resilience mechanisms.
At a broadcast frequency of 10 Hz in a 2 $Mbit$ channel with a density of $1/20$ $[nodes/m]$, nodes farthest away in the broadcast zone, in best cases, only receive the message with the probability $P_{mf} \approx 30\%$. This should be related to the QoS requirements of $P_{mf} = 99\%$. A real deployment with higher channel transmission rates would expectedly improve these results. However even in non-contended situations the multipoint-to-multipoint broadcast operations cause correlated losses that makes it difficult to reach the required QoS requirements.
Still the cross-layer optimization approach can be introduced to achieve the best possible results.

Capabilities of the developed system model

The system model has been implemented to evaluate its capabilities of providing estimates of P_{mf} and D_{e2e}. The model offers adjustment of the layer parame-

ters: *transmission power*, *PHY mode* and *FEC code rate*. This can be done in relation to varying *densities* and end-user service parameters of *broadcast frequency* and *data payload*.
Due to the limited amount of modulation schemes supported, the model parameter space is limited. However, a comparison can verify the general capabilities of the model.
A simulation study of the DCAD service has been extended to evaluate P_{mf} and D_{e2e} at varying transmission powers for a given density at a node close to the edge of the zone at approximately 300 m. The same parameters used in the simulation have been inserted into the model. As the considered case is under high contention the 802.11 MAC model is used.
As expected the model predicts P_{mf} higher than the simulation results. This is primarily due to the correlated losses not considered in the flooding broadcast model. However, the model provides a result with a unique solution to maximize P_{mf}. This could be valid even though the qualitative comparison of the two result sets fail.
For a density of 1/20 $[nodes/m]$ the model suggests a transmission power of 11.74 dBm which corresponds to a transmission range of approximately 210 m. In a similar simulation run the best result is achieved at a transmission range of approximately 160 m corresponding to a transmission power of 9.39 dBm. However, the model suggests that near optimal results can be achieved within a transmission range of $170 - 260$ m. A similar analysis of the simulation results suggest the same within a range of $140 - 200$ m. These result indicate an interesting correlation between the model and the simulation results. Based on verification results on the individual sub-models and these comparisons it is expected that the suggested system model can be improved to deliver useful optimization parameter settings.

A central aspect in deriving the D_{e2e} in the model, is an estimate of the mean hop count distance H to nodes from a broadcast source. A basic estimate of H has been provided from the same flooding broadcast simulation scheme on which the empirical model is based. The model results have been compared to simulation results at low contention where queueing is minimal. There is a good correlation between the model and simulation results. Future effort is required to include analytical models to derive D_{e2e} for various load situations.

Optimal strategies in the DCAD end-user service

Considering the optimal parameters of the DCAD service in multipoint-to-multipoint environment it has been shown that an optimal transmission range is not necessarily when attempting to transmit a message directly to the targeted node of a broadcast. Instead a more reliable approach is to transmit the message in multiple hops. This has been shown both in simulation and by the model for a density of $1/20[nodes/m]$. As the density increases so does contention. From simulation results it has also been shown that a density of $1/10[nodes/m]$ degrades D_{e2e} significantly when considering the outer nodes in the zone. The system model provides the same results, however, its optimistic estimate of D_{e2e} suggests that a small transmission range of a few nodes ($R < 5$) will allow a reception probability above 60% at 300 m. Clearly a relization of such a scenario requires additional resilience mechanisms in the protocol stack. This is a motivation to include ARQ schemes and possibly RTS/CTS to handle the heavily contended situations.

Optimization approach
An overall aim in the applied optimization approach has been to improve reliability and performance globally for all nodes within a zone. The mean to provide optimal parameter settings has in this work been to consider the performance and reliability metrics of a node situated farthest from the source node. Based on model results this approach seems reasonable. There are most hops toward this node leading to increased mean delays. Further it experiences the least amount of retransmissions, decreasing its probability of a successful reception. Many of the conduced simulation results support this assumption. However it has not been consistent in simulation results. In particular during high contention cases the outer node may actually experience better QoS, while nodes close to the source are degraded as the transmission power is varied. This aspect should be studied further.

Mobility is clearly a large topic in MANETs. This aspect has been considered on a principle level in this work by studying varying node densities and exponentially distributed distances between cars. The latter aspect has shown how variability on car positions may significantly affect results of successful message receptions. This means that the optimal parameter settings calculated from a mean case may not be ideal for a specific case. Evaluating these aspects requires an implementation of the suggested model in a dynamic environment. This is considered a part of future work.

Applicability of elaborate models
In this work the approach has been made to construct a system model and individual elaborate sub-models. This has provides a good starting point to study the functional aspects of channel, PHY and MAC mechanisms that are highly relevant when attempting to improve reliability and performance in MANETS. Further, analytic modeling approaches to describe the performance and reliability of the flooding broadcast scheme has provided valuable insight into the mechanisms affecting its functionality. However, the resulting elaborate models may not be well suited for CL optimization. Specifically not if they should be implemented in wireless devices with limited resources.
On a standard 1.7 GHz Centrino system, a single calculation of the output metrics is expected to take approximately 100 ms in an optimized implementation. To make a search based optimization approach, multiple calculations would be required, increasing the time significantly to calculate a single set of optimal parameters using these models. In a dynamic optimization approach a lag in providing the optimal estimates, in relation to the changing conditions, may actually degrade performance of the optimized process. Thus, future work should consider how these models could be simplified while maintaining capabilities to provide optimal parameter settings.

8.3 Outlook

The physical channel model could be extended to support the 5.5 and 11 $Mbit/s$ modes, by looking into approximative approaches, such as it was possible for the 2 $Mbit/s$ mode. This would also enable the benefit of of applying FEC, since the higher rate modes are more sensitive to the degrading effect of varying channel conditions.

From an overall perspective it would be interesting to formulate the optimization problem as for example a Markov Decision Process. This would possibly allow more efficient choice of optimal parameters, which could be efficient enough to be integrated into COTS products with limited processing capabilities.

In relation to the MAC sub-model, the models considered in this work have shown room for improvement. The desired properties of an enhanced model would probably be an extension of the Bianchi model that takes hidden nodes and non-saturated conditions into consideration.

Further, the assumption of having a single mean value of the channel error and collision probabilities to represent all nodes could be replaced by an approach where the nodes are evaluated on a more individual basis. However, the downside to such an approach is an increased degree of complexity.

An obvious topic for further work is to consider a more efficient broadcast scheme for the DCAD service. This would entail a lower degree of contention in the network by reducing the number of redundant deliveries.

In relation to this it would also be interesting to introduce mobility traces. This would enable a more realistic environment for the DCAD service case, and the mobility aspect would impose new challenges on the broadcast mechanism.

The flooding scheme is attractive to study as it primarily is well defined and simple to establish and secondarily is used in real deployments of broadcasting e.g. in the DSR ad-hoc routing protocol. It is further expected that it may be possible to apply experiences from this work to more elaborate broadcast schemes in future work.

Appendix A

Optimization variables and layer parameters

A.1 Optimization variables

A.1.1 Performance metrics

The following list presents the performance metrics that influence the user's experience of the end-user service.

Delay Important to consider in interactive real-time communication such as a A/V conference, where a too high delay degrades the user experience.

Jitter The amount of variations in delay, which is also a relevant metric in interactive real-time communication or streaming services.

Throughput Nearly all service types have more or less strict requirements to the rate with which they are able to transmit and receive.

Packet loss Only relevant for services using connection-less transport. In a video streaming session the loss of packets could result in degradation of the perceived video quality.

Energy consumption For battery driven mobile nodes the energy consumption has a direct influence on the life-time of the node.

Resource usage In devices with limited resources that run multiple simultaneous services, the use of CPU, memory and storage is relevant to consider.

A.1.2 Resilience metrics

From the end-user perspective the following resilience metrics are relevant to consider:

Blocking probability The probability of being blocked when requesting a service.

Service loss probability The probability of the QoS dropping below the QoS that the service requires to function acceptably.

Detection and recovery time The time that is used to detect and recover from a loss of service.

A.2 Layer parameters

In this work the considered layer parameters cover the following parameters in layers 1 and 2.

A.2.1 L1

Physical modes (802.11x) Specifies in which physical mode the wireless node operates. Physical modes define *modulation scheme*, *code rate* and *data rate* used [Zhao et al., 2003].

TX Power level (802.11x) Defines the transmission power level in dBm or Watt.

A.2.2 L2

FEC Code rate Defines the amount of error correction code bits as the ratio $\frac{K}{N}$ between the number of information bits K in relation to the total codeword length N.

Appendix B

Application cases

The following application cases are based on applications defined in HIDENETS deliverable 1.1.

B.1 Blackboard service

B.1.1 Service description

The blackboard is made up of a collection of information of various types, e.g. video, audio, pictures, documents, etc. This information is relevant within a limited geographic region, where it should be made available to any ad-hoc node (car) that comes sufficiently near. Since information about a point of interest should be available before the point of interest, the distribution of data should happen via nearby cars. The distribution of information between cars would happen in wireless ad-hoc networks, where connections are being created when the cars get sufficiently near each other.

B.1.2 Service requirements

- No errors should be present above transport layer, to avoid degeneration of data. In streaming video/audio applications some degree of loss can often be accepted, but since the information is continuously being transferred between cars, degeneration is not desired.
- High throughput is needed to be able to transfer audio and video within the possibly short period of time a connection between two cars exists.
- Jitter and latency are less important factors.
- Priority classes for messages could be necessary to ensure that e.g. traffic reports are received.

B.1.3 Protocol stack

Blackboard service
Connection-oriented (TCP/SCTP)
IP-routing, ad-hoc topology control
Error control, multi-channel MAC
WLAN

B.1.4 Themes

- Connection-oriented communication in ad-hoc networks.
- Session/connection hand-over.
- Proximity/geographically based routing methods.

B.2 Video conference - access to medical expertise

B.2.1 Service description

This service is used while the ambulance is at the scene of accident and while heading back to the hospital with the injured. It may be of vital importance to keep the multimedia session running while driving, preferably at highest possible quality. This involves seamless handovers between access points and even handovers between access points of different technologies. WLAN probably has a limited range, whereas cellular technologies as 3G, EDGE, etc. have wider coverage, but lower bandwidth.

B.2.2 Service requirements

- Uninterrupted audio/video streams with seamless hand-over.
- Strict timing requirements (latency/sync/jitter)
- Adaptive audio/video coding that selects coding parameters/algorithm based on the current conditions.
- Connection technology handover depending on current coverage.

B.2.3 Protocol stack

Video conference service, codec
UDP/RTP
IP-routing, multihoming
Error control, multi-channel MAC, multi-radio management
WLAN, 3G, EDGE, ...

B.2.4 Themes

- Wireless to infra-structure.
- Handover between access points and technologies.
- Codec selection/parameters.

- real-time service.

B.3 Distributed car-data for assisted driving

B.3.1 Service description

Cars exchange information of positions, movements, etc. to form a picture of the current traffic situation. The cars communicate in ad-hoc networks using broadcasting or multicasting. Each car uses this information to discover potential hazards. The driver is warned so that an accident can be avoided, or the safety systems in the car are prepared for a crash or safety systems take control of the throttle, brakes and steering of the car. On longer ranges the system can provide information about the traffic situation, e.g. queues or slippery roads. Such information would come from a central service and be distributed to the ad-hoc networks via access points connected to infra-structure.

B.3.2 Service requirements

This service has been selected as the service case considered in this work. Thus requirements have been quantified. A list of the service parameters and their requirements is given in Table 2.2 on page 18. In this section a discussion of the background for the requirements is presented.

The *data payload* contains the information that is transmitted to surrounding cars in the environment regularly. The regularity is given by the transmission frequency which defines when newly sampled updates of the data payload is transmitted. The transmission rate is defined from how quickly dynamics in the physical environment change. The main factors are changes in velocity (acceleration) and how fast information needs to be updated to allow other cars to react in a timely manner. Based on frequency components in car mobility ($2-3\ Hz$ [G. et al., 2000][Lin and Ulsoy, 1995]) and smoothness in the control mechanisms [Franklin et al., 1997] an upper bound approximation is defined at $50\ Hz$. However, the *transmission frequency* may be varied depending on the acceleration and velocity of the car.
With the assisted driving service, platooning [Svinnset et al., 2006a] and adaptive cruise control would be relevant functionalities to include. For instance this would allow cars to drive closer while maintaining safety. This is possible as the task of e.g. emergency braking is no longer delayed by the slow ($1\ s >$ [McGehee et al., 2000]) reaction time of humans. Instead the reaction time $latency_{reaction}$ is given by latencies in the system from a car starts braking until the car behind reacts:

$$latency_{reaction} = latency_{transmission} + latency_{sampling} + latency_{processing} \quad \text{(B.1)}$$

$latency_{sampling}$ is the potential latency introduced by the discrete transmission frequency of car-data information. E.g. at a transmission frequency of 50 Hz a message is in worst cast first sent $1/50\ Hz = 20\ ms$ after a car starts breaking. $latency_{processing}$ is the latencies imposed by the computing system in both a transmitting and receiving car to sample sensors, issue a message to be sent, interpret a message and initiate a control command. This latency is considered

insignificant compared to $latency_{transmission}$ and $latency_{sampling}$. Finally the largest contribution is expected from the $latency_{transmission}$. As an example two cars both travel at a speed of 150 km/h 12 meters apart. The leading car brakes abruptly at the position x_0 with a deceleration of 9 m/s^2 (adapted from [Svinnset et al., 2006a]) until a complete halt. The following car starts braking as well, and for a safety margin the following car must have come to a complete halt at least 3 meters after the leading car. Consequently the following car must start its deceleration at least 3 meters before x_0. It uses approximately 200 ms to travel the 9 meters (12 m − 3 m) meaning that $latency_{reaction} \leq 200\ ms$. Consequently $latency_{transmission} \leq 200\ ms - 2latency_{sampling} \leq 160\ ms$ if it is also required that two consecutive messages may not be lost due to transmission error.
Requirements for *jitter* are given from the potential impact on real-time control systems that jitter has. Jitter can degrade control performance significantly and in worst case lead to instability[Bortoff, 2000]. In this case bounds are defined for allowable jitter in relation to maximum mean latency.
To ensure reliable operation of the mission critical DCAD service only few faults are allowed in the application layer. Specifically a packet loss rate of 1% is tolerated when further requiring that two consecutive packets may not be lost. Further a low Bit Error Rate of $10^{-}4$ is accepted. It is expected that these faults can be handeled by observer-based filtering techniques in the control algorithms [Franklin et al., 1997].
The final requirement defines a *zone range* with its origin in a car with the end-user service. This range defines an area in which cars, heading in the same direction, must receive the messages within the given requirements.

B.3.3 Protocol stack

DCAD service
UDP/RTP
IP-routing - ad-hoc topology control
Error control, multi-channel MAC
WLAN

Appendix C

Modeling

C.1 Overall model

Table C.1 presents the interfaces of the overall model.

Interfaces in the overall model	
MAC model	
Name:	*Description and Unit:*
$\rightarrow$ `tx_freq`	Transmission frequency of the DCAD end-user service [(frames/s)/node].
$\rightarrow$ `frame_size`	Size of the frame to transmit including headers and FEC overhead [bytes].
$\rightarrow$ `col_nodes`	Average number of neighbor nodes within collision range including source node, floating point [nodes].
$\rightarrow$ `phy_mode` (α_2^1)	Modulation scheme, code rate and data rate: 1e6, 2e6, 5.5e6, 11e6 [Mbit/s].
$\leftarrow D_{mac}$	Delay distribution of frames from it is ready to send until it can access the medium successfully [μs].
$\leftarrow J_{fw}$	Delay distribution of frames caused by MAC forwarding jitter (uniform distribution 0-0.5 ms) [μs].
FEC model	
$\rightarrow$ `code_rate` (α_1^2)	Relation of original data to the entire data amount including redundant data.
$\rightarrow$ `payload_size`	Amount of data in the payload. I.e. DCAD service data [bytes].
$\leftarrow$ `frame_size`	-
$\leftarrow G_{coding}$	FEC coding gain.
$\rightarrow P_{ber}$	Probability of a bit error.
$\leftarrow P_{fer}$	Probability of a frame loss due to bit errors after FEC.
PHY model	
$\rightarrow$ `frame_size`	-
$\leftarrow G_{coding}$	-
$\rightarrow$ `phy_mode` (α_2^1)	-
$\rightarrow$ `pow_tx` (α_1^1)	Transmitted signal power [dBm].
$\rightarrow$ `density` (λ)	Density of nodes in a stretch of road in one direction (including both lanes). Parameter controls exponentially distributed distances between cars [nodes/m].
$\leftarrow D_{tx}$	Delay of a transmission on the wireless medium from the first bit is sent until the last bit is received [μs].
$\leftarrow P_{out}$	Probability of a frame loss due to insufficient received signal power.
$\leftarrow$ `rad_col_nodes`	-
$\leftarrow$ `phy_mode` (α_2^1)	-
Broadcast Mechanism	
$\rightarrow P_{loss}$	Mean probability of frame loss.
$\rightarrow$ `node_src`	Source node. The node count position of the considered source node relative to an edge node. An edge node is generally defined as the last reachable node within the range of the zone.
$\rightarrow$ `node_dest`	Destination node. The node count position of a considered destination node relative to the source node.
$\rightarrow$ `zone_range`	The radius of the zone in meters [m]
$\rightarrow$ `rad_nodes`	Radius of nodes within transmission range.
$\leftarrow H$	Hop-count distribution between `node_src` and `node_dest` [hops].
$\leftarrow P_{mf}$	The probability of a message broadcast from `node_src` not reaching its destination `node_dest`.

Table C.1: *Interfaces in the overall model.*

C.2 Flooding broadcast model

This section contains an extended analysis of a flooding broadcast in a partially connected ad-hoc network where $R = 2$.

C.2.1 Extended analysis

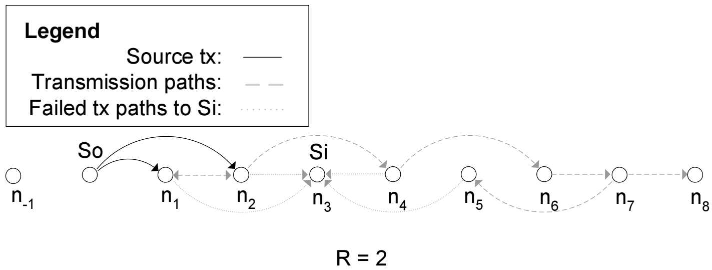

Figure C.1: *A studied example of a flooding broadcast in a network where* $R = 2$.

The analysis takes it starting point in the flooding scenario of Figure C.1. So initiates a broadcast and the aim is to derive P_{mf} for Si which in this case is defined to be node n_3. To simplify the analysis initially transmission from the left section are not included. Thus nodes n_{-K} do not retransmit messages in this example.
Subsequently a broadcast progress is analyzed in the order from So issues the first transmission until all relevant paths toward Si are covered. In this example $P_s = 0.6$. Intermediate calculations of the message error probability are denoted P'_{mf}.

Step 1. So transmits a message which is received by n_1 and n_2 with the probability P_s. To calculate the probability that Si (n_3) receives the message from either n_1 or n_2 a fully connected network with nodes $n_K, K = 1 \dots 3$ can be considered. However, So cannot transmit directly to Si thus only probabilities from neighbor nodes are considered. As a result $P'_{mf} = M(1,\ 2) = 0.340$. Using a fully connected network for this calculation is possible in this special case as probabilities of a successful transmission are equal for n_1 and n_2. General cases, where this assumption cannot be made, are discussed shortly.

Step 2. At $R = 2$ n_2 may also transmit to n_4. If Si has failed reception of a transmission from either n_1 or n_2 chances are to successfully receive a retransmission from nodes n_4 or n_5. This requires that n_4 successfully receives a message from n_2. In the example of Figure C.1 n_4 receives the message. Its retransmission fails to reach Si and n_5. Instead n_6 receives and forwards the message resulting in a path $n_6 \rightarrow n_7 \rightarrow n_5 \rightarrow Si$. In a hypothetical case a message could have continued to propagate on every second node i.e. $n_8 \rightarrow n_{10} \dots$ and then return again $\dots n_9 \rightarrow n_7 \rightarrow n_5 \rightarrow Si$. Basically this shows that indefinitely many paths exist assuming the network is infinitely large. However, most of these paths are highly improbable allowing them to be excluded from the calculations. Equation

C.1 is an expansion of $M(1,\ 2)$ which includes paths over n_4 and n_5. Common for these nodes are that they may transmit directly to Si. ω is the probability that a message propagates from $n_2 \rightarrow n_4 \wedge n_5$ and fails to reach Si. Consequently $P'_{mf} = 0.277$. Including n_6 in Equation C.1 (not shown) causes $P'_{mf} = 0.274$. Thus it is presumably reasonably to exclude paths on nodes outside transmission range of Si.
In this step it should be noted that a message can only pass Si $n_2 \rightarrow n_4$ and not the other way around.

$$\begin{aligned} P'_{mf} &= (1-P_s)^2 + [P_s(1-P_s)[P_s(1-P_s)\omega + (1-P_s)] \\ &\quad + P_s(1-P_s)[P_s(1-P_s) + (1-P_s)]\omega](1-P_s) + P_s^2(1-P_s)^2\omega \\ where, &\quad \omega = [P_s(1-P_s)[P_s(1-P_s) + (1-P_s)] + (1-P_s)] \end{aligned} \tag{C.1}$$

Step 3. To continue the analysis the P_{mf}-probabilities of $n_2 \wedge n_3$ should be updated depending on each other and paths over $n_3 \rightarrow \ldots$. This requires derivations of n_3 independent of n_2 to calculate P_{mf} for n_1 etc.

Step 4. Finally focus can be shifted setting $n_4 = Si$. At this point the probability that n_2 and n_3 transmits a message is different. Thus the calculations can no longer be carried out as if $n_2 \wedge n_3 \wedge n_4$ are a part of a fully connected network. Instead contributions from $n_2 \wedge n_3$ to n_4 must be considered individually. This require a probability of a retransmission from either n_2 or n_3 independent of the other. However as these nodes depend on former transmissions from e.g. n_1 their independent probability of a retransmission is not easily obtained.

C.3 Physical channel

C.3.1 Signal to Noise Ratio definition

Since different variants of the Signal to Noise Ratio (SNR) is used extensively in the following, the following defines these. The SNR is generally defined as the ratio between the received signal power P_r and the received noise power. The noise is considered white Gaussian noise with mean zero and power spectral density $N_0/2$. With the bandwidth of the transmitted signal being $2B$, the noise power is given by N_0B [Goldsmith, 2005]. Thus the SNR is given by:

$$SNR = \frac{P_r}{N_0 B} \tag{C.2}$$

Typically, the signal to noise relation is expressed in terms of the energy per chip E_c, energy per symbol E_s or the energy per bit E_b. These relate to the SNR as:

$$SNR = \frac{P_r}{N_0 B} = \frac{E_c}{N_0 B T_c} = \frac{E_s K}{N_0 B T_s} = \frac{E_b H}{N_0 B T_b} \tag{C.3}$$

where T_c is the chip time, T_s is the symbol time and T_b is the bit time. G is the number of chips per symbol while H is the number of chips per bit. For 802.11b the signal bandwidth $2B$ is $2 \cdot 11\ MHz$ and the chipping rate is $11\ Mchips/s$, i.e. the chip time T_c is 11 μs [IEEE, 2003]. This simplifies the SNR expression to:

$$SNR = \frac{E_c}{N_0} = \frac{E_s G}{N_0} = \frac{E_b H}{N_0} \tag{C.4}$$

Here, E_c corresponds to the received power resulting from path-loss, shadowing and multi-path effects, while the noise power is defined by thermal noise or electrical noise in the receiver equipment. The values of G and H depend on the used 802.11b PHY mode. For 1 $Mbit/s$ DBPSK $H = G = 11$ since only one bit is sent in each symbol with a length of 11 chips. 2 $Mbit/s$ DQPSK transmits 2 bits in each 11 chips symbol, thus $H = 5.5$ while G is unchanged at $G = 11$. For the CCK based PHY modes the symbol length is 8 chips, and thus $G = 8$. 5.5 $Mbit/s$ CCK transmits 4 bits per symbol resulting in $H = 2$, while 11 $Mbit/s$ CCK transmits 8 bits per symbol, resulting in $H = 1$.

C.3.2 Average BER of DBPSK in Ricean channel

The expression for the average BER in Ricean channel is derived analytically in the following. The starting is to insert eq. (4.4) and eq. (4.8) into eq. (4.12), which gives

$$P_e = \int_0^\infty \frac{1}{2} exp(-x) \cdot \frac{(1+K)}{\frac{E_b}{N_0}} exp\left(\frac{-(1+K)x}{\frac{E_b}{N_0}} - K\right) I0\left(2\sqrt{\frac{(K^2+K)x}{\frac{E_b}{N_0}}}\right) dx \tag{C.5}$$

From [Spiegel, 1999, eq. 27.32] the following relation is known

$$I_n(x) = e^{-n\pi i/2} J_n(ix) \tag{C.6}$$

For $n = 0$, this expression may be reduced to

$$I_0(x) = J_0(ix) \tag{C.7}$$

Further, by using the following relation from [Spiegel, 1999, eq. 27.94]

$$\int_0^\infty e^{-ax} J_0(b\sqrt{x}) dx = \frac{exp\left(\frac{-b^2}{4a}\right)}{a} \tag{C.8}$$

and identifying

$$a = \frac{(K+1) + \frac{E_b}{N_0}}{\frac{E_b}{N_0}} \tag{C.9}$$

$$b = 2i\sqrt{\frac{(K^2+K)}{\frac{E_b}{N_0}}} \tag{C.10}$$

eq. (C.5) may be written as

$$P_e = \frac{M}{2} exp(K(M-1)) \tag{C.11}$$

Where $M = \frac{(1+K)}{(1+K)+\frac{E_b}{N_0}}$. Eq. (C.11) has been verified numerically against eq. C.5.

C.4 CSMA model

The following contains the derivation of the CSMA MAC model based on [Wu and Varshney, 1999] referenced from the medium access sub-model in section 4.4 on page 82.

C.4.1 Derivation of CSMA model

The wireless channel around a node may be modeled as a Markov Chain as depicted in Figure C.2. The duration of the busy state B is the time it takes to transmit a frame $D_B = T$, and the duration of the idle state I is a slot time $D_I = a$. As specified in the list of assumptions, a silent node tries to transmit in each slot with probability p according to a Bernoulli process. Since a node may only transmit if it senses the channel idle, the achieved transmission probability p' is lower than the ready probability p. The achieved transmission probability may be expressed as

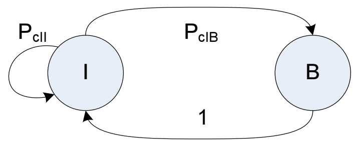

Figure C.2: *The channel has an idle state I and a busy state B.*

$$p' = p \cdot P_{cI} \tag{C.12}$$

where P_{cI} is the limiting probability that the channel is idle. In order to determine P_{cI}, the transition probability P_{cII} is needed. The precondition for the transition P_{cII} is that no nodes in the channel transmits. Thus,

$$P_{cII} = \sum_{i=0}^{\infty} (1 - p')^i \cdot p(i) \tag{C.13}$$

where $p(i)$ is the probability that there are i nodes within hearing range. Since the nodes are distributed according to a Poisson process along a line, $p(i)$ may be calculated as

$$p(i) = \frac{(\lambda N)^i exp(-\lambda N)}{i!} \tag{C.14}$$

The transition probability P_{cII} may now be calculated as

$$P_{cII} = \sum_{i=0}^{\infty} (1 - p')^i \frac{(\lambda N)^i exp(-\lambda N)}{i!} = exp(-p' \lambda N) \tag{C.15}$$

From the MC in Figure C.2, it can be seen that $P_c(I) = P_c(I) \cdot P_{cII} + P_c(B) \cdot 1$. Since $1 = P_c(I) + P_c(B)$, the steady-state probability of the channel being idle is

$$P_c(I) = \frac{1}{2 - P_{cII}} \tag{C.16}$$

The limiting probability that the channel is idle P_{cI} is given by the expected amount of time spent in the idle state in one cycle in relation to the expected time spent in all states in one cycle.

$$P_{cI} = \frac{D_I P_c(I)}{D_B(1 - P_c(I)) + D_I P_c(I))} = \frac{a}{a + T(a - P_{cII})} \tag{C.17}$$

The achieved transmission probability p' may now be obtained as

$$p' = p \cdot P_{cI} = \frac{p \cdot a}{a + T(1 - P_{cII})} = \frac{p \cdot a}{a + T(1 - exp(-p'\lambda N))} \tag{C.18}$$

In order to evaluate the performance obtained by a single node, a node is modeled as a three-state Markov Chain as depicted in C.3. A node may be in the idle state I while it does not transmit. The probability that it does not transmit is $P_{II} = 1 - p'$. The duration of the idle state is $D_I = a$. When a transmission occurs, the node may experience a successful transmission and enter state S or a colliding transmission and enter state C. The duration of a transmission is $D_S = D_C = T$.

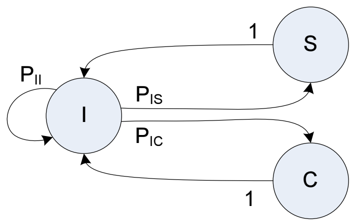

Figure C.3: *The three states of a node; idle I, successful transmission S and collision C.*

The aim of this analysis is to calculate the probability that a receiver node experiences a collision given that a transmission has occurred. To determine this probability the limiting probabilities of the states S and C, P_S and P_C are needed. With a starting point in the MC of a node in the figure, the steady-state probability of state I may be expressed as

$$P(I) = P(I) \cdot P_{II} + 1 \cdot P(S) + 1 \cdot P(C) \tag{C.19}$$

Since $1 = P(S) + P(C) + P(I)$,

$$P(I) = P(I) \cdot P_{II} + 1 - P(I) \tag{C.20}$$

which yields

$$P(I) = \frac{1}{1 + p'} \tag{C.21}$$

The probability that a successful transmission occurs depends on the preconditions described initially. The transition probability $P_{IS}(r)$ may be expressed as

$$\begin{aligned} P_{IS}(r) = P[&x \text{ transmits}] \\ &\cdot P[y \text{ does not transmit in the same slot}] \\ &\cdot P[\text{nodes in } C(r) \text{ do not transmit in the same slot}|r] \\ &\cdot P[\text{nodes in } B(r) \text{ do not transmit in vulnerable period}|r] \end{aligned} \tag{C.22}$$

The probability that nodes in $C(r)$ do not transmit in a slot, may be calculated as in eq. (C.14)

$$P_c(r) = exp(-p'\lambda C(r)) = exp(-p'\lambda(2R - r)) \tag{C.23}$$

The probability that nodes in $B(r)$ do not transmit in a slot is calculated similarly

$$P_b(r) = exp(-p'\lambda B(r)) = exp(-p'\lambda r) \tag{C.24}$$

This gives that the transition probability $P_{IS}(r)$ may be expressed as

$$P_{IS}(r) = p' \cdot (1 - p') \cdot exp(-p'\lambda(2R - r))^1 \cdot exp(-p'\lambda r)^{2\tau+1} \tag{C.25}$$

To determine this transition probability for any node, the average over $0 \leq r \leq R$ should be calculated. As listed in the assumptions, sender and receiver nodes are uniformly chosen within the radius R. Thus, the pdf of the distance r is a uniform distribution

$$f(r) = 1/R \tag{C.26}$$

P_{IS} is given by

$$P_{IS} = \int_0^R \frac{1}{R} p' \cdot (1 - p') \cdot exp(-p'\lambda(2R - r))^1 \cdot exp(-p'\lambda r)^{2\tau+1} dr \tag{C.27}$$

In [Wu and Varshney, 1999] the corresponding expression for P_{IS} is used to derive the throughput of the channel. However, in this work the desired output is the probability that a transmission leads to a collision, that is, the probability that a collision occurs, given that a transmission has occurred. Since the duration of a successful transmission equals the duration of a collided transmission, this collision probability may be obtained from the limiting probabilities of the node being in the successful and collision states, as described in the following.

The steady state probability of the state C may be specified in terms of the transition into state C. Further, from the transitions from state I it is known that $1 = P_{IC} + P_{IS} + P_{II}$. Thus,

$$P(C) = P(I) \cdot P_{IS} = \frac{1}{1 + p'}(p' - P_{IS}) \tag{C.28}$$

The limiting probability of state C may be expressed as

$$P_C = \frac{D_C P(C)}{D_C(1 - P(I)) + D_I P(I)} = \frac{p' - P_{IS}}{p' + \frac{a}{T}} \tag{C.29}$$

and the limiting probability of state S is

$$P_S = \frac{D_S P(S)}{D_S(1 - P(I)) + D_I P(I)} = \frac{P_{IS}}{p' + \frac{a}{T}} \tag{C.30}$$

The probability that a transmission results in a collision may be expressed as

$$P_{col} = \frac{P_C}{P_S + P_C} = \frac{p' - P_{IS}}{p'} \tag{C.31}$$

Appendix D

Mobility model

Generally, the features of a mobility model can be categorized into macro and micro-mobility [Haerri et al., 2007]. When focusing on a macroscopic point of view, motion constraints such as roads, streets, crossroads, and traffic lights are considered. Also, the generation of vehicular traffic such as traffic density, traffic flows, and initial vehicle distributions are defined. The microscopic point of view focuses on the movement of each individual vehicle and on the vehicle behavior with respect to other vehicles and pedestrians.

Macroscopic properties of service case
The considered service case has the following macroscopic properties:

- Motorway with two lanes in each direction.
- 1 km of straight road.
- The speed limit is the same for the whole stretch of motorway.
- Traffic density for each direction should be adjustable parameters.
- Initial vehicle distribution should resemble the typical distribution.

Microscopic properties of service case
The considered service case has the following microscopic properties:

- Cars should behave as cars with human drivers, however, they should never crash.
- Cars may overtake.

In relation to future work, the following presents two approaches for obtaining realistic mobility model. One approach to obtaining a realistic mobility model that is mentioned in the survey [Haerri et al., 2007], is to use mobility trace files generated from the DaimlerChrysler-internal micro-mobility simulator Farsi. The generation of a fixed set of trace files is described in [Krüger et al., 2005]. However, this solution does not allow adjustment of traffic density, besides what is available from the selection of trace files.

A mobility generator that allows the adjustment of mobility parameters is the VanetMobiSim [Fiore et al., 2007]. This is based on the Intelligent Driver Model

(IDM) that describes the behavior of each car via mathematical expressions. The implementation in VanetMobisim has lane changing capabilities and has been validated with respect to several real-life phenomema in [Fiore et al., 2007]. According to the survey in [Haerri et al., 2007], this mobility simulator is the only freely available that is able to generate ns-2 traces based on a sufficiently realistic model of human driver behavior.

Appendix E

Model verification and test

This appendix covers specific verification tests for for each sub-model and overall tests related to optimization. For the physical channel model the verification tests cover *path loss*, *SNR to bit error probability* and *frame loss probability due to channel errors*. For the medium access sub-model the two considered models are compared to simulation results. The last verification tests concern the flooding broadcast sub-model, which is compared to simulation results based on a simple MAC scheme and a realistic 802.11 MAC scheme.

Details and discussion of tests are also found in the report Chapter 6.

E.1 V1 PHY models

E.1.1 V1.1 Path-loss model

The purpose of this test is to verify that the deterministic path-loss predicted by the physical channel sub-model matches the simulation. The test is based on the combined free space and two-ray path-loss model. The results are expected to be accurately matching, since the path-loss model used in sub-model and simulation is the same.

The scenario used in this tests is configured with equidistant nodes placed 5 meters apart. At 0 meters, a transmitter nodes is located, that broadcasts messages. For each node that receives the broadcast, the SNR value of the received signal is output from ns-2. These results have been produced and plotted for transmission powers of 1 and 100 mW. The path-loss model of the physical channel sub-model has been fed with the transmission powers and node positions, to obtain the predicted received SNR. The results are shown in Figure E.1. The results have been found to be matching within the relevant scope of values. For the 1 mW simulation plot, the values are not plotted as the received signal power drops below the receive threshold.

E.1.2 V1.2 PHY modulation model

The Figure E.2 depicts the two sets of SNR to BER values discussed in Chapter 5 on page 104 and the expression for the bit error probability in an AWGN channel for the DBPSK modulation scheme. The evaluation framework uses the values

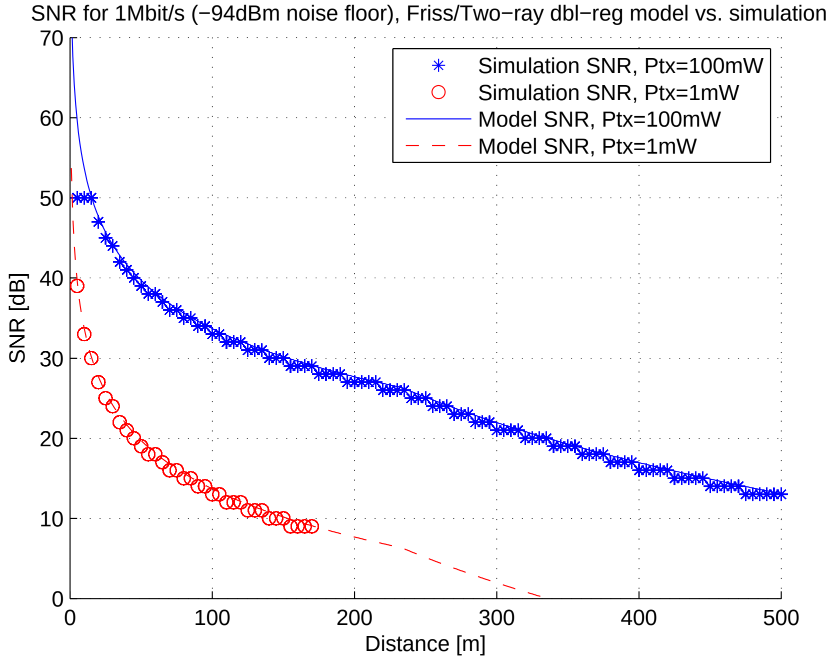

Figure E.1: *Deterministic path-loss verification.*

labeled 'Intersil empirical', which are also the values that match the analytical expression best.

E.1.3 V1.3 PHY frame loss probability

Table E.1 contains the parameters used in this test.

E.2 V2 MAC model

The main goal of this test is to validate the assumptions of the two MAC model against simulations. The non-persistent CSMA and 802.11 MAC models are compared to determine their applicability.

The simulations are based on the realistic PHY layer described in Chapter 5 on page 104.

E.2.1 V2.1 Varying transmission power

In the first MAC test the transmission power is varied. Since the node density is constant, the level of collisions is expected to increase as the collision domain increases in size. The scenario is configured as stated in Table E.2. All nodes are performing layer 2 broadcast with a given frequency, however no forwarding is done, and the broadcast frequency of each node is therefore not dependent on the amount of successfully received transmissions. The test has been performed

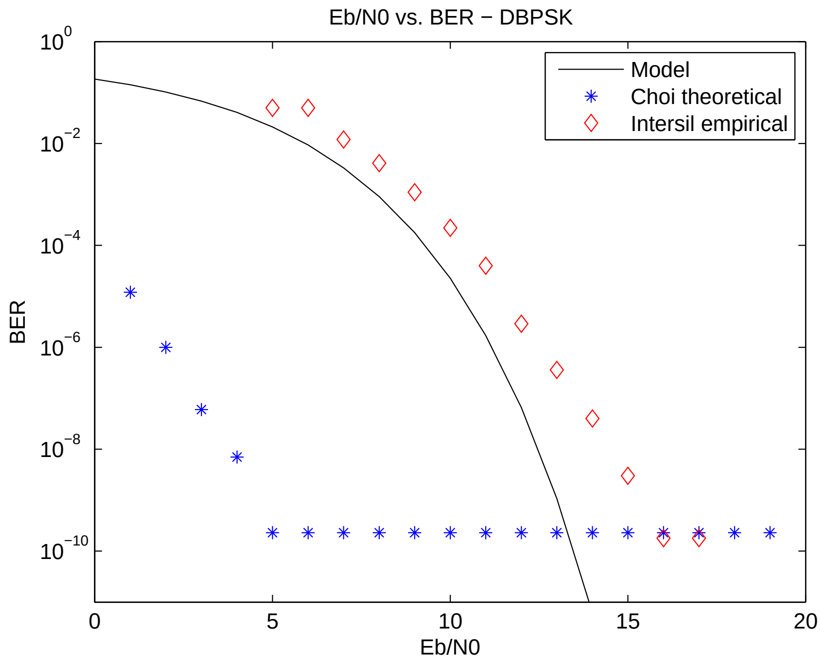

Figure E.2: *Comparison of SNR to BER conversion in ns-2 and analytical model.*

Parameter	Values
PHY mode α_2^1	2 Mbit.
Density β_1^1	$\frac{1}{5}$ [nodes/m]
TX pow α_1^1	10 mW
Broadcast frequency	20 Hz
Node distribution type	Equidistant
Radio propagation	Ricean fading (K=6), combined free space and two-ray
Noise floors	-91 dBm vs. -101 dBm
Receive thresholds	-75 dBm vs. -95 dBm
Carrier sense thresholds	-85 dBm vs. -104 dBm

Table E.1: *Simulation and model parameters for the varying physical layer sub-model test.*

for 2 different broadcast rates, to investigate both a non-congested scenario and a scenario with a higher degree of congestion.

Parameter	Values
PHY mode α_2^1	1 Mbit.
Density β_1^1	$\frac{1}{10}$ [nodes/m]
TX pow α_1^1	$-23.01\ dBm$, $-16.02\ dBm$, $-9.03\ dBm$, $-2.04\ dBm$, $4.94\ dBm$, $11.94\ dBm$
Broadcast frequency	$20\ Hz$, $150\ Hz$
Node distribution type	Poisson
Radio propagation	Ricean fading (K=6), combined free space and two-ray
Capture threshold	$10\ dBm$

Table E.2: *Simulation and model parameters for the varying tx-power test of the MAC models.*

Results
The results of the test are shown in Figure E.3 and 6.3. The simulation results show that the number of collisions increase as the transmission power is increased and a higher number of nodes and especially hidden nodes exist. The tests have been run with the capture threshold in ns-2 set to 10 dB in order to reflect a realistic scenario. The result of this setting is that the receiver nodes in the simulations are less sensitive simultaneous transmissions, which reflects the behavior of an actual wireless adapter.

In Figure E.3, where the network is clearly not saturated, the 802.11 MAC model overestimates the number of collisions, since it is based on the assumption of saturated network conditions. The non-persistent CSMA model initially overestimates the probability of collisions. Since the shape of the curve is different from the simulation results, the non-persistent CSMA model underestimates the collision probability significantly at the higher transmission powers.
In Figure 6.3 the broadcast frequency has been increased and the network is closer to a saturated state. Here the 802.11 MAC model is closer to the simulation result, however the results are still overestimated, but clearly shows the correct tendency. In relation to the non-persistent CSMA model, the model still overestimates for the low transmission powers, while it underestimates the collision probability at the higher transmission powers.

E.2.2 V2.2 Varying broadcast frequency

In addition to varying the transmission power, also the broadcast frequency has been varied. The scenario has been configured as stated in Table E.3.

Results
In general, at low contention levels, the difference between the 802.11 and non-persistent CSMA schemes is not very pronounced, since the contention window backoff in 802.11 does not have significant effect. However, as the level of con-

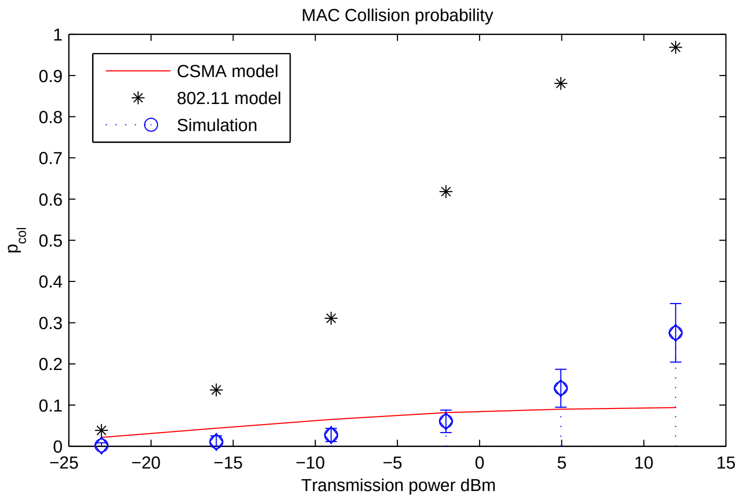

Figure E.3: *Collision probability for varying transmission power. Based on broadcast frequency* 20 *Hz*.

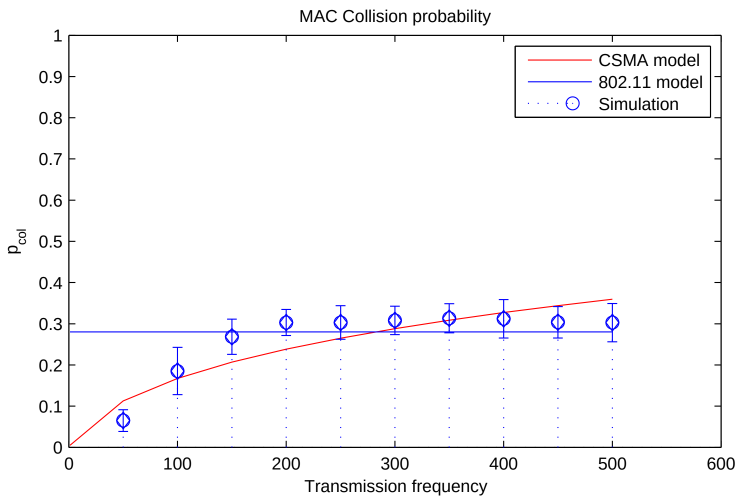

Figure E.4: *Collision probability for varying broadcast frequency. Based on node density* 1/10.

Parameter	Values
PHY mode α_2^1	1 Mbit.
Density β_1^1	$\frac{1}{10}$, $\frac{1}{5}$ [nodes/m]
TX pow α_1^1	$-20.00\ dBm$
Broadcast frequency	50 Hz, 100 Hz, 150 Hz, 200 Hz, 250 Hz, 300 Hz, (350 Hz, 400 Hz, 450 Hz, 500 Hz)
Node distribution type	Poisson
Radio propagation	Ricean fading (K=6), combined free space and two-ray
Capture threshold	10 dBm

Table E.3: *Simulation and model parameters for the varying broadcast frequency test of the MAC models.*

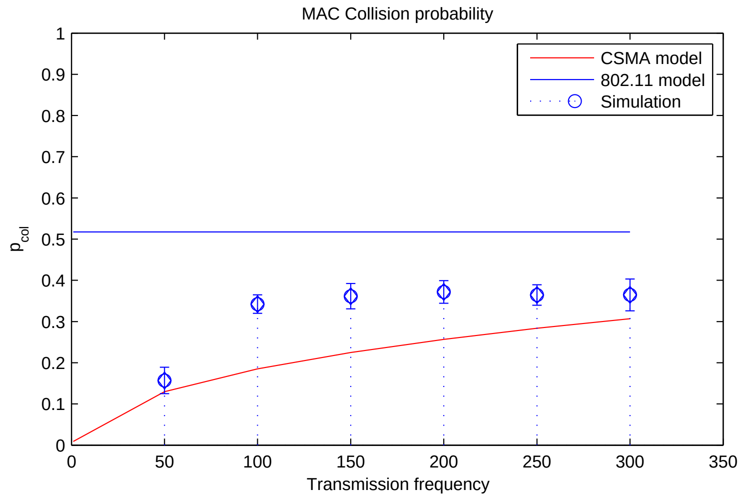

Figure E.5: *Collision probability for varying broadcast frequency. Based on node density* 1/5.

tention increases the contention window in 802.11 limits the frequency at which a single node is able to initiate transmissions, and thereby also cause collisions.

For the simulation, the collision level saturates, possibly due to the fact that also hidden nodes must back-off for the nodes within their hearing range and due to the limiting effect the capture threshold has on the collision probability. As the broadcast frequency increases, the medium becomes fully contended and messages starts to become buffered in interface queue. This means that the actually achieved broadcast frequency does not increase further.
For the non-persistent CSMA sub-model in Figure E.4 the collision probability keeps increasing above the saturation level of the simulation, due to the increased risk of hidden nodes transmitting in the vulnerable period. In the case with higher contention in Figure E.5, the non-persistent CSMA model underes-

timates the collision probability.

The 802.11 predicts the probability of collisions well for the saturated case in Figure E.4. However, as the density increases in figure E.4, the model overestimates the probability of collision. This may be due to the capture threshold setting in the simulation that limits the effect of hidden nodes.

E.3 V3 Flooding broadcast model

This section contains simulation and test results for the verification of the flooding broadcast mechanism. Initially the simple MAC component is used. It enables a verification of the flooding broadcast mechanism under the same assumptions it has been developed regarding independent losses and a given successful transmission probability P_s. This enables a baseline for verifying the flooding broadcast model to simulation results based on a more realistic 802.11 based MAC scheme.
The simple MAC verification analysis is conducted with both equidistant and exponentially distributed nodes.

E.3.1 V3.1 Simple MAC

In summary this test enables a verification of the flooding broadcast model under the following desired conditions:

1. Losses are guaranteed to be independent.
2. Loss probabilities are controllable.

In the simple MAC verification test only P_{mf} is evaluated as the implemented MAC model does not consider realistic MAC access delay and transmission times.

V3.1.1 Varying reception probabilities and transmission power
In this test simulation results are conducted based on the simple MAC component for varying reception probabilities and different transmission powers. These results are compared to the flooding broadcast model.
An overview of the parameters used to conduct this test is defined in Table E.4.

Parameter	**Values**
PHY mode α_2^1	2 Mbit. Has a small significance as real transmission times are not used.
Density β_1^1	$\frac{1}{10}$ [nodes/m]
TX pow α_1^1	$-5.1542\ dBm$ (30m/$R = 3$), 0.8672 dBm (60 m/$R = 6$), 5.3028 dBm (100m/$R = 10$), 17.0415 dBm (300m/$R = 30$)
P_s	0.1, 0.2
Broadcast frequency	20 Hz
Node distribution type	Equidistant

Table E.4: *Simulation and model parameters for the simple MAC test of the flooding broadcast scheme.*

In each simulation case only one broadcasting node is considered. It is placed in the middle of the road stretch ensuring that an equal amount of cars exist in either direction. The zone range is 300 m leading to 60 nodes being within range excluding the source node.

One single simulation run is conducted for each test case as there is no variability in the placement of nodes. Each test run is conducted for 5 seconds leading to 100 broadcasts (20/s). It has been verified that a broadcast is completed before a new one is instantiated. In all tests P_s is verified to be as expected. It is further verified that no queue losses exist in either of the tests.

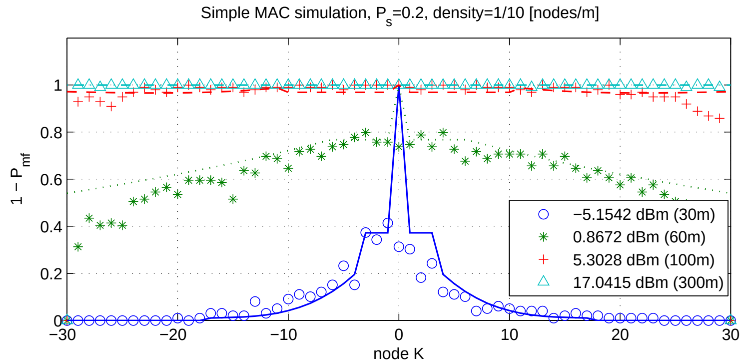

Figure E.6: *Comparison of flooding broadcast model and simple MAC simulation where $P_s = 0.2$. Node $K = 0$ is the broadcast source.*

The results are shown in figures 6.4 on page 115 for $P_s = 0.1$ and E.6 for $P_s = 0.2$. There is a good correlation between the model and the simulation results.

V3.1.2 Exponentially distributed nodes

In this test the distribution used to position cars is changed from equidistant to an exponential distribution. This means that there will be variations in the amount of nodes a transmission can reach. This is intuitively closer to what would be experienced in a real environment where cars travel with different distances.

As there is a variation in the distance between cars 30 different simulation runs are conducted with a duration of 3 seconds. With a BC frequency of 20 Hz in total statistics are based on 180 broadcasts. The same scenario configurations as in verification test V3.1.1 are used.

Mean results are depicted in Figure E.7 on the facing page. Apparently simulation results are a bit lower than in case of the equidistant simulations and the model. This is expectedly due to the effect of the exponentially distributed positions. There is a significant probability that there will be a gap between two subsequent cars reducing the amount of nodes reached by a broadcast. This clearly affects the distribution of the reception probabilities. This may have an effect on the efficiency of the optimization approach as the model does not capture this aspect.

E.3.2 802.11 MAC

Finally the broadcast flooding model is compared to simulation results using the more realistic 801.11 MAC component. The 802.11 MAC component in ns-

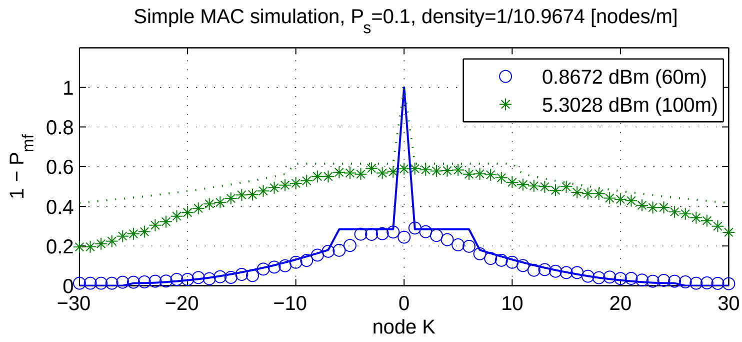

Figure E.7: *Comparison of flooding broadcast model and simple MAC simulation where $P_s = 0.1$ and nodes are exponentially distributed. Node $K = 0$ is the broadcast source.*

2 has been used, patched according to the description in Chapter 5 on page 104. In this test case losses are not directly controllable but must be extracted from simulation data. The parameters used for simulation are given in table E.5.

Parameter	**Values**
PHY mode α_2^1	2 Mbit.
Density β_1^1	$\frac{1}{20}$ [nodes/m]
TX pow α_1^1	0.8672 dBm (60 m/$R = 3$)
Broadcast frequency	20 Hz, 10 Hz
Node distribution type	Equidistant
Radio propagation	Ricean fading (K=6), combined free space and two-ray

Table E.5: *Simulation and model parameters for the 802.11 MAC test of the flooding broadcast scheme.*

The results of the simulation with the 802.11 MAC layer are given in Figure 6.6 on page 117. Clearly correlated losses lead to a heavily increased probability of message faults (P_{mf}).

E.4 T1 - High contention case simulation

The purpose of this test is to consider the simulation environment in the execution of the DCAD service with multiple services. The test must emphasize the effects of adjustment of layer parameters in relation to varying density conditions. The metrics to consider are the probability of message delivery P_{mf} and the end-to-end delay D_{e2e} at different nodes in the zone. The parameters used in this test are given in Table E.6.

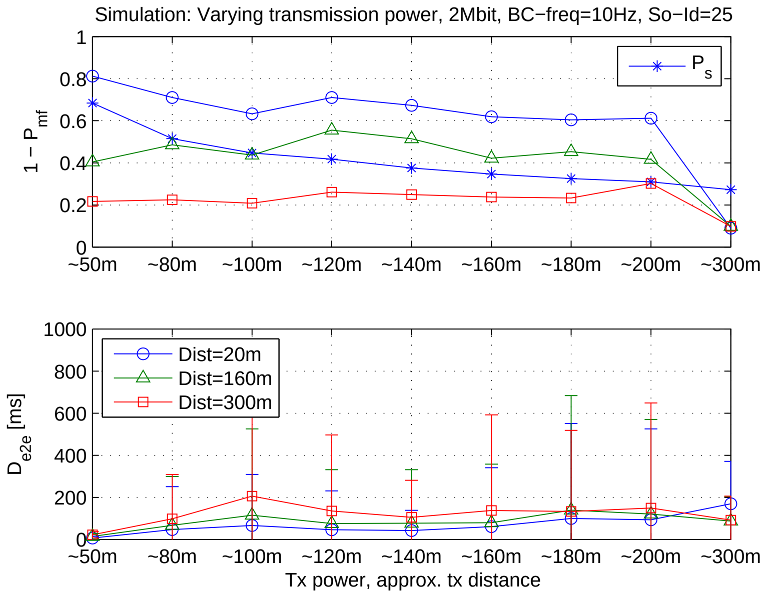

Figure E.8: *Message reception probability and end-to-end delay for a contended network where the density is* 1/20.

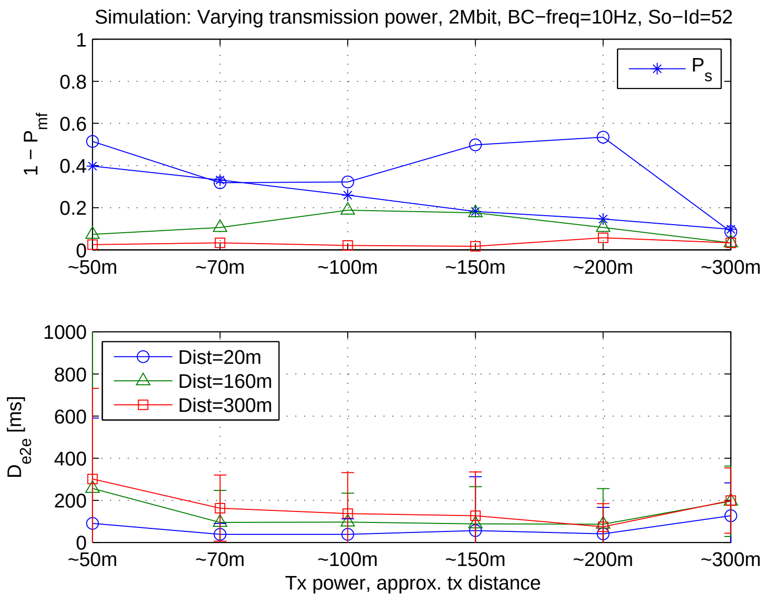

Figure E.9: *Message reception probability and end-to-end delay for a heavily contended network where the density is* 1/10 *(node 52).*

Parameter	**Values**
PHY mode α_2^1	2 Mbit.
Density β_1^1	$\frac{1}{20}$, $\frac{1}{10}$ [nodes/m]
TX w α_1^1	$-0.72\ dBm(50m)$, $3.36\ dBm(80m)$, $5.30\ dBm(100m)$, $6.89\ dBm(120m)$, $8.23\ dBm(140m)$, $9.39\ dBm(160m)$, $10.41\ dBm(180m)$, $11.32\ dBm(200m)$, $17.04\ dBm(300m)$
Broadcast frequency	$10\ Hz$
Node distribution type	Equidistant
Radio propagation	Ricean fading (K=6), combined free space and two-ray
Capture threshold	off

Table E.6: *Simulation and model parameters for the T1.*

E.4.1 T2 - Model verification

End-to-end delay

Table E.7 presents the parameters used in the simulation for establishing the end-to-end delay.

Parameter	**Values**
Source node	26
PHY mode α_2^1	2 Mbit.
Density β_1^1	$\frac{1}{20}$ [nodes/m]
TX pow α_1^1	$2.2mW$ ($\tilde{8}0\ m$)
Broadcast frequency	$20\ Hz$
Node distribution type	Equidistant
Radio propagation	Ricean fading (K=6), combined free space and two-ray

Table E.7: *Simulation parameters for the end-to-end delay test.*

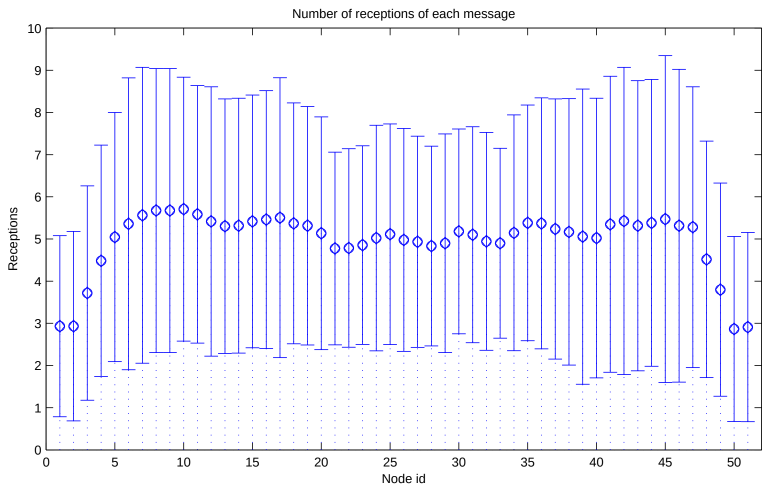

Figure E.10: *Number of received copies of a message for each receiver node.*

Appendix F

Abbreviations

AODV Ad-hoc On-demand Distance Vector

ARQ Automatic Repeat reQuest

AWGN Additive White Gaussian Noise

BER Bit Error Rate

BN Bayesian Network

CCK Complimentary Code Keying

CDS Connected Dominating Set

CL Cross-Layer

CLO Cross-Layer Optimization

CLRO Cross-Layer Resilience Optimization

COTS Commercial Off-The-Shelf

DBPSK Differential Binary Phase-Shift Keying

DCF Distributed Coordination Function

DPSK Differential Phase-Shift Keying

DQPSK Differential Quadrature Phase-Shift Keying

DSR Dynamic Source Routing

DSSS Direct Sequence Spread Spectrum

ELN Explicit Loss Notification

FEC Forward Error Correction

FER Frame Error Rate

HMM Hidden Markov Model

IDM Intelligent Driver Model

LoS Line of Sight

MAC Media Access Control

MANET Mobile Ad-hoc Network

MDP Markov Decision Process

OFDM Orthogonal Frequency-Division Multiplexing

PDF Probability Density Function

PDU Protocol Data Unit

PLCP Physical Layer Convergence Protocol

POMDP Partially Observable Markov Decision Processes

PSK Phase-Shift Keying

QoS Quality of Service

RF Radio Frequency

RTT Round-Trip Time

SINR Signal to Interference plus Noise Ratio

SNIR Signal to Noise plus Interference Ratio

SNR Signal to Noise Ratio

SSI Signal Strength Indication

TDMA Time Division Multiple Access

VBR Variable Bit-Rate

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