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Performing Gateway Load Balancing in MANETs

Thesis for the degree of Philosophiae Doctor

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Abstract

During the last decades, the advances in Information Technology have formed the basis for increased interest and research activity in the field of ad hoc wireless multihop networks or simply ad hoc networks. This emerging technology enables internetworking between wireless nodes that are deployed in an ad hoc and temporary manner. All nodes in an ad hoc network take the role as both hosts in an end-to-end communication session, or as routers to collaboratively relay data traffic in a multihop fashion on behalf of other nodes. Furthermore, ad hoc networks are highly dynamic in nature, i.e. nodes can join or leave the network at any time, and additionally, the nodes have also the flexibility to move around while being in the network. The fact that ad hoc networks can be rapidly deployed with minimal prior planning, cost, and without the need of any pre-existing infrastructure makes this technology very attractive and suitable in a number of applications, including emergency and rescue operations, and military operations.

Although ad hoc networks represent a promising technology that offers a broad range of potential useful applications, this technology is still in an immature phase. There are yet many issues and challenges that need to be resolved, which mainly arise from the inherent unreliability of wireless communication, the dynamic nature of these networks, the limited availability in resources with respect to bandwidth, processing capacity, battery power, and from the possibly large scale of these networks. These challenges require that the networking protocols at all layers in the network stack, that in many cases were originally designed for wired networks, must be modified or optimized, in order to adapt to the characteristic of the wireless environment.

The focus of this thesis has been devoted to the investigation of two specific issues within the field of ad hoc networking, i.e. *node mobility* and *load balancing*. The aim is to provide solutions in order to improve the overall performance in ad hoc networks.

Node mobility is one of the most important features in ad hoc networks, however, it

is also the reason for frequent link breaks and the constant change in the topology. An ongoing data transmission that is interrupted by a link break, must be rerouted to alternative paths in order to circumvent the broken link. However, this process of rerouting traffic takes a certain amount of time, which is referred to as the *rerouting time*. Minimizing the rerouting time is essential in order to reduce packet loss and improve network performance. In this thesis we investigate the factors that affect the rerouting time in proactive routing protocols and propose solutions for minimizing it.

Load balancing refers to the process of distributing traffic load more evenly in the network in order to minimize congestion and to optimize the usage of network resources. Performing load balancing in ad hoc networks is generally very challenging due to the inherently interfering nature of the wireless medium. In this thesis we therefore investigate the feasibility and the potential benefits of performing load balancing in ad hoc networks. We consider two scenarios, i.e. load balancing for *intradomain* and *interdomain* traffic.

Intradomain traffic is traffic between nodes inside an ad hoc network. Performing load balancing on intradomain traffic can be done in two ways. The first is referred to as *multipath load balancing* where a traffic flow between a source and destination pair is distributed over multiple alternative disjoint/semi-disjoint paths. The aim is to maximize throughput and reduce the risk for packet loss. However, a number of previous work has investigated and reported that this type of load balancing can only provide a rather limited improvement in performance due to the interference between the paths [1] [2]. Due to this reason, multipath load balancing is therefore not considered in this thesis. Instead we focus on the second way which is referred to as *transit routing*. Transit routing is about routing part of the local traffic over a *backbone* network in order to relief the traffic load in the ad hoc network. The assumption behind this concept is a network architecture similar to a Wireless Mesh Network (WMN), where a high capacity backbone network is an integrated part of the Mobile Ad hoc Network (MANET). This backbone network is commonly used to provide Internet-connectivity services, but can also be exploited to alleviate the traffic load in the MANET. In addition, for certain source and destination pairs, performing transit routing can considerably increase the throughput compared to if the traffic is routed within the ad hoc network.

Interdomain traffic refers to traffic between a node inside the ad hoc network and a remote node outside of the ad hoc network. Load balancing for interdomain traffic considers the potential of distributing interdomain traffic among multiple gateways in order to avoid congestion at the gateways and maximize the capacity for interdomain traffic. This type of load balancing is commonly referred to as *gateway load balancing* in the literature. Furthermore, interdomain can either be

inbound or outbound traffic. The work in this thesis mainly focuses on performing load balancing for outbound traffic. However, we believe that the results in our work are also applicable to inbound traffic as well.

The main contributions in this thesis are the investigation and the proposals of different solutions for intradomain and interdomain load balancing.

Preface

This thesis is submitted to the Norwegian University of Science and Technology (NTNU) in Trondheim in partial fulfillment of the requirements for the degree of Philosophiae Doctor.

The work on this thesis began in December 2005, and was carried out at the University Graduate Center (Unik) at Kjeller, under the supervision of Professor Øivind Kure at the Norwegian University of Science and Technology and Professor Paal Engelstad at the University of Oslo (UiO), Telenor R&I and Simula Research Laboratory. Professor Knut Øvsthus at Bergen University College was also partly involved in the supervision during the first year.

The research is supported by the Research Council of Norway (NFR) and is related to two EU projects, the Quality of Service in Ad-hoc Networks (QUAD) and Deployable High Capacity Gateway for Emergency Services (DeHiGate) projects. The goal of these projects is to devise solutions for future broadband communication system for emergency rescue operations.

Dedication

I would like to dedicate this work to my beloved father who suddenly passed away during the course of this doctoral study.

Acknowledgements

I would like to express my deepest and sincere appreciation to my supervisor Professor Øivind Kure and Professor Paal Engelstad for giving me the opportunity to be their student. Without their patience, guidance, encouragement and understanding, my accomplishment would never been possible, but rather an illusion.

I am thankful to my employer, The Norwegian Defence Research Establishment (FFI), for the willingness and generosity in allowing me the opportunity to enrich my knowledge as well as improving my skill as a researcher.

I would also like to express my gratitude to my former project manager at FFI, Professor Knut Øvsthus at Bergen University College, for his arrangement such that I could begin with the PhD study. He has given me many good advices and taught me many useful things for my thesis work.

I would like to thank Dr. Erlend Larsen who has assisted me, all from the simplest thing to the most difficult technical matters. I have really learnt and enjoyed much during these years working close with him. I would also like to thank Dr. Lars Landmark for many good discussions and ideas.

My sincere appreciation to all friends and colleagues at Unik for giving me a joyful and memorable time.

Finally I would like to thank my mother, my brother and sister, my wife and daughter for their sacrifice, understanding, support and encouragement during these years.

Kjeller, October 15, 2011

Vinh Pham

List of Publications

The thesis is based on the following five papers, referred to in the text by letters (A-E). Paper A is written in cooperation with Erlend Larsen and should be regarded as equally shared primary authorship with the author of this thesis. The author of this thesis is the principal contributor of paper B to E. The co-authors provided with invaluable contributions in terms of ideas, comments and corrections in all above papers. The author of this thesis has contributed to papers F to I as a discussion partner and partly in the code implementation in paper I.

PAPER A: V. Pham, E. Larsen, K. Øvsthus, P. Engelstad and Ø. Kure, "Rerouting Time and Queueing in Proactive Ad Hoc Networks," In proceedings of the International Performance Computing and Communications Conference 2007 (IPCCC 2007), New Orleans, USA, April 11-13, 2007, pp. 160-169.

PAPER B: V. Pham, E. Larsen, Ø. Kure and P. E. Engelstad, "Routing of Internal MANET Traffic over External Networks," Mobile Information Systems Journal, iiWAS/MoMM special issue, Volume 5, Number 3, 2009

PAPER C: V. Pham, E. Larsen, P. E. Engelstad and Ø. Kure, "Performance Analysis of Gateway Load Balancing in Ad Hoc Networks with Random Topologies," Proceedings of The 7th ACM International Symposium on Mobility Management and Wireless Access (Mobiwac09), Tenerife, Canary Islands October 26-30, 2009

PAPER D: V. Pham, E. Larsen, Ø. Kure and P. Engelstad, "Gateway Load Balancing in Future Tactical Networks", IEEE Military Communications Conference 2010 (MILCOM 2010), San Jose, CA, USA, October 31 - November 3, 2010

PAPER E: V. Pham, E. Larsen, Q. Le-Trung, P. Engelstad and Ø. Kure, "A Radio Load Based Gateway Load Balancing Scheme with Admission

Control," Proceedings of the International Symposium on Wireless and Pervasive Computing (ISWPC 2011), Hong Kong, China, February 23-25, 2011.

Related papers:

PAPER F: E. Larsen, V. Pham, P. Engelstad, Ø. Kure, "Gateways and Capacity in Ad Hoc Networks," In proceedings of the International Conference on Advances in Human-oriented and Personalized Mechanisms, Technologies, and Services 2008, (I-CENTRIC 2008), Sliema, Malta, October 26-31, 2008, pp. 390-399, ISBN: 978-0-7695-3371-1

PAPER G: E. Larsen, L. Landmark, V. Pham, Ø. Kure and P. E. Engelstad, "Routing with Transmission Buffer Zones in MANETs," In proceedings of the IEEE International Symposium on a World of Wireless Mobile and Multimedia Networks 2009 (WoWMoM 2009), Kos, Greece, June 15-18, 2009.

PAPER H: E. Larsen, L. Landmark, V. Pham, P. E. Engelstad, Ø. Kure, "Preemption Mechanisms for Push-to-Talk in Ad Hoc Networks," The 34th IEEE Conference on Local Computer Networks 2009 (LCN 2009), Zürich, Switzerland, October 20-23, 2009

PAPER I: E. Larsen, L. Landmark, V. Pham, P. E. Engelstad, Ø. Kure, "Optimized Group Communication for Tactical Military Networks," IEEE Military Communications Conference 2010 (MILCOM 2010), San Jose, CA, USA, October 31–November 4, 2010.

Contents

List of Figures

Abbreviations

AC Admission Control

ACK Acknowledgment

AODV Ad hoc On-Demand Distance Vector Routing

AP Access Point

CBR Constant Bit Rate

CDMA Code Division Multiple Access

CDS Connected Dominating Set

CEDAR Core Extraction Distributed Ad Hoc Routing

CSMA Carrier Sense Multiple Access

CSMA/CA Carrier Sense Multiple Access with Collision Avoidance

CSMA/CD Carrier Sense Multiple Access with Collision Detection

CTS Clear To Send

CW Contention Window

DCF Distributed Coordination Function

DeHiGate Deployable High Capacity Gateway for Emergency Services

DIFS Distributed Inter-Frame Space

DREAM Distance Routing Effect Algorithm for Mobility

DSDV Destination-Sequenced Distance-Vector Routing

DSSS Direct-Sequence Spread Spectrum

DSR Dynamic Source Routing

DYMO Dynamic Manet On-Demand Protocol

EL Even Load

ETT Expected Transmission Time

ETX Expected Transmission Count

FDMA Frequency Division Multiple Access

FHSS Frequency Hopping Spread Spectrum

FIFO First In First Out

FTP File Transfer Protocol

GPS Global Positioning System

GW Gateway

HNA Host and Network Association

HOLSR Hierarchical Optimized Link State Routing

HSR Hierarchical State Routing

IETF Internet Engineering Task Force

LAR Location-Aided Routing

LANMAR Landmark Ad Hoc Routing

MAC Medium Access Control

MACA Multiple Access with Collision Avoidance

MANET Mobile Ad hoc Network

MID Multiple Interface Declaration

MIP Mobile IP

MPR Multi Point Relay

NAM Network Animator

NHDP Neighbor Hood Discovery Protocol

ns-2 Network Simulator

OFDM Orthogonal Frequency-Division Multiplexing

OLSR Optimized Link State Routing

OSI Open Systems Interconnection

OSPF Open Shortest Path First

PCF Point Coordination Function

PDA Personal Digital Assistant

PRNG Pseudo Random Number Generator

QoS Quality of Service

QUAD Quality of Service in Ad-hoc Networks

RERR Route Error

RLAC Radio Load Based Load Balancing with Admission Control

RLLB Radio Load Based Load Balancing

RREP Route Reply

RREQ Route Request

RTS Request To Send

RTT Round Trip Time

SIFS Short Inter-Frame Space

SLB Simple Load Balancing

SNR Signal to Noise Ratio

SP Shortest Path

TBRPF Topology Broadcast based on Reverse-Path Forwarding

TC Topology Control

TCP Transmission Control Protocol

TDMA Time Division Multiple Access

UDP User Datagram Protocol

VoIP Voice over IP

Wi-Fi Wireless Fidelity

WLAN Wireless LAN

WMN Wireless Mesh Network

WSN Wireless Sensor Network

ZRP Zone Routing Protocol

Part I

Introduction

Chapter 1

Introduction

1.1 Background

Communication systems for emergency and rescue operations are evolving from single-hop, voice-only systems, to more sophisticated broadband multihop communication networks, supporting a diversity of services including voice, data, and video communications. This enhancement in functionality and capacity enables efficient exchange of critical information, increases situational awareness and allows for rapid response in emergency and rescue operations.

The technology offered by ad hoc networks is envisioned to be one of the key components in the realization of such future communication networks. An ad hoc network has a number of advantageous features. It can serve as an extension to wired communication infrastructure as in last mile communication, or it can also function as an independent infrastructureless network. The key feature in an ad hoc network is the capability of mobile nodes to collaborate and to build a network without the need of any predeployed infrastructure. All communication are thus entirely based on wireless links, which again facilitate rapid deployment of the network, and in a more cost effective way. Furthermore, the network can automatically and quickly adapt to the dynamics in an emergency and rescue operation, i.e. nodes are allowed to be mobile, join and leave the network at any time.

Although ad hoc networking is envisioned as a promising technology for future emergency and rescue communication systems, there are yet many challenges that need to be solved. These challenges are related to factors such as the lack of a centralized administration, node mobility, capacity, scalability, and interference. They can adversely affect the overall performance of the network as well as the

experienced Quality of Service (QoS) of the end users. Hence, before it is possible to adopt the ad hoc networking technology in such a communication system, solutions must be developed to overcome these challenges.

1.2 Motivation and Objectives

The work in this thesis addresses the challenges related to *node mobility* and *load balancing*. The aim is to develop solutions in order to improve the overall performance in ad hoc networks.

Typically, in an emergency and rescue scenario, rescue teams and personnel will usually be in constant motion when performing their task. Due to this reason, it is therefore important that the underlying communication network supports mobility. However, node mobility is the source to many of the challenges that an ad hoc network has to face. First of all, mobility may result in frequent link breaks and packet loss. Second, it can even lead to loss of connectivity. While it is very difficult to guarantee that the connectivity is maintained at all time in a dynamic and mobile environment, it is possible to minimize the packet loss due to link breaks. When ongoing data traffic is interrupted by a link break, the routing protocol is responsible for rerouting the traffic via alternative paths in order to circumvent the broken link. However, this process of rerouting traffic usually takes a certain amount of latency, which is referred to as the *rerouting time*. Minimizing the rerouting time is essential in order to reduce packet loss and improve network performance.

Load balancing refers to the process of distributing traffic load more evenly in the network in order to minimize congestion and to optimize the usage of network resources. This is especially important in wireless ad hoc networks since the effective capacity with respect to throughput in such networks is considerably lower than in wired networks. Nevertheless, a more load-balanced network can help to extend the network lifetime, especially when considering that each node in the network has only a limited battery capacity. However, due to the interfering nature in wireless ad hoc networks, it is therefore questionable whether it is feasible to perform load balancing under such conditions. Hence the focus is to investigate the feasibility and the potential benefits of performing load balancing in ad hoc networks. More specifically, we consider load balancing for two different scenarios, i.e. load balancing for *intradomain* and *interdomain* traffic. Load balancing for intradomain traffic considers the potential of routing part of the local traffic over a backbone network (such as the backbone in a Wireless Mesh Network (WMN)) in order to relief the traffic load in the ad hoc network. On the other hand, load balancing for interdomain traffic considers the potential for distributing interdomain

traffic among multiple gateways in order to avoid congestion at the gateways and maximizing the capacity for interdomain traffic.

To summarize, the focus in this thesis is to investigate the issues related to mobility and load balancing as described above. In this context, a number of questions arise that need to be answered:

Mobility:

- What are the factors that affect the rerouting time in proactive routing protocols?
- Why does the rerouting time in many cases considerably exceeds the time needed to detect a link break?
- How can we minimize the rerouting time?

Load balancing:

- What is the feasibility and the potential benefit of performing load balancing in wireless ad hoc networks?
- What are the factors that affect the performance of load balancing?
- How can we perform load balancing in an efficient way?

1.3 Research Methodology

Research may be defined as the search for knowledge or a systematic investigation. Different methods or approaches can be applied in the research depending of the field of concern. In this section we will discuss the research methods or approaches that are common in the field of ad hoc networking. More importantly, we will discuss the research methods that we have used in our investigation and the search for the answers to our questions.

1.3.1 Theoretical Analysis

The method of *theoretical analysis* can be used to acquire fundamental knowledge of the issues, mechanisms or system being investigated. In this approach, mathematical or statistical models are often used to describe a phenomenon, or to estimate the expected quantity of certain parameters. We have for example in paper A derived a model to estimate the rerouting time. However, during the work in this thesis, we experienced that many of the issues or scenarios being studied are too

complicated to be solved by using a theoretical approach. This is especially true in scenarios where there are many nodes, and if these nodes are in addition mobile. Due to this reason, most of our work is therefore based on computer simulations that will be discussed in the following section.

1.3.2 Simulations

In cases where the issue or scenario being investigated is too complex, using the method of performing *simulations* may be easier. This approach has a number of advantages:

1. Complex scenarios with many nodes and various traffic and mobility patterns, can be set up within a short amount of time in a simulator. The time required to perform a simulation depends on many factors such as the computer power, the number of nodes, the desired simulation time, the traffic and mobility pattern. In any case, performing investigations on complex scenarios can be conducted much faster and with lower cost than performing similar investigations with real life experiments.
2. Any desired simulation data is easily accessible from the trace file, including packet transmissions, receptions and collisions at any nodes. The availability of these data makes it much easier to explore and analyze simulation results.
3. The fact that simulation results are reproducible (by rerunning the same simulation with the exact same random number seed) makes it possible to thoroughly study a particular event of interest or a chain of events, through code tracing. In contrast, in real world experiments, it is usually very difficult to exactly recreate a particular event.
4. Using for example Linux shell scripts, it is possible to automate the simulation process in order to study a large number of different topologies, or the impact of varying simulation parameters.

1.3.3 Real Life Experiments

The third method is conducting *real life experiments*, which is the most accurate method to evaluate the behavior of the scenario or the system being studied. This approach allows the system to be exposed to all physical constraints and effects that prevail in the real world, and should therefore be used when possible. In connection with the investigation of the rerouting time in paper A, we initially conducted

a number of real life experiments in order to measure the rerouting time. However, we experienced that retrieving all necessary data, such as the amount of queued packets, was very difficult due to the unavailability of source code for the wireless network interface's firmware and drivers. Additionally, conducting real life experiments in practice is often very challenging and time consuming, especially when the network scenario consists of many nodes. Due to these reasons, all subsequent work in paper B to E related to load balancing, are mainly based on simulations and the analyses of the simulation results.

1.3.4 Research Method in this Thesis

Method
Throughout the work in this thesis we have used a combination of the three methods discussed above. However, due to the complexity of the scenarios that we investigated, much of the work is therefore based on simulations using the Network Simulator (ns-2) simulator [3] version 2.28-2.33. ns-2 is a discrete event network simulator that began as a variant of the REAL network simulator in 1989 and has evolved substantially over the past few years. It is one of the most popular simulators in the networking research community due to the concept of open source, the rich availability of online documentation, and the active newsgroups. From the large community of users, it is easier to get support to the challenges that one may encounter.

Code Validation/Verification
Various algorithms/mechanisms for improving the network performance have been developed in this thesis, including the radio load measurement algorithm, admission control mechanism, various load balancing schemes, and the adaptive retry limit mechanism for improving the rerouting time. These algorithms/mechanisms are implemented in ns-2, either at the network or the link layer. Moreover, we have also implemented a number of tools used in the analysis of the simulation data. In order to assure that the algorithms and analysis tools fulfill their intended purposes, the implemented codes have been through comprehensive testing and debugging. For the network algorithms/mechanisms, we have used a variety of simple to more complex network models to verify the desired response or behavior.

Scenarios
Throughout the work in this thesis, simulations are performed on a large number of different topology configurations, including both simple static string and grid topologies, to the more complex random static and mobile topologies. The scenarios are created on the basis that they are simple and helped to underline the

strengths and weaknesses of the studied mechanisms. The simple topologies are used to gain knowledge about the fundamental mechanisms that prevail in wireless multihop networks such as the rerouting time, interference, and congestions. On the other hand, the complex random topologies are used to explore the behavior and performance of the proposed algorithms/mechanisms in scenarios close to "real networks". Furthermore, we used a large number of topologies in our simulations with the purpose to achieve statistically more reliable results. The result may therefore serve as an estimate on the expected performance of the algorithms/mechanisms.

Propagation Model
For all simulations in this thesis, the two-ray ground propagation model is used. The two-ray ground model assumes that the received signal strength is the sum of the direct line of sight path and the path of one reflection from the ground. A common restriction of this model and similar models is the fact that they do not allow to specify obstacles in the environment. In addition, the simulator allows that only one propagation model can be used for the entire duration of the simulation, meaning that the spatial and temporal variations cannot be modeled. We have intentionally chosen to use this simple model in order to keep the test environment as simple as possible. The purpose is to isolate the simulation scenarios from fluctuating and temporary disturbances at the physical layer. Our focus is to study, gain knowledge, and provide improving solutions at the network or link layer, such as those related to rerouting time or load balancing. Furthermore, since the two-ray ground model is one of the most used propagation model, it is therefore a natural choice to also use it in our work. This enables comparison with results from other work.

Traffic Patterns
Only Constant Bit Rate (CBR) traffic over User Datagram Protocol (UDP) is used in the simulations, which is the common transport protocol in research related to ad hoc networks. We did not use Transmission Control Protocol (TCP), although it is the prevalent transport protocol in the Internet, due to several reasons. First of all, the work in [4] shows that the performance of TCP in wireless multihop networks is degraded using various Medium Access Control (MAC)-sublayer protocol such as Carrier Sense Multiple Access (CSMA) and Multiple Access with Collision Avoidance (MACA). Second, when using TCP over IEEE 802.11, there are problems such as throughput oscillation and severe unfairness, as demonstrated in [5]. Third, TCP does not differentiate between congestion-related packet drops and transmission failures at link layer. TCP treats all packet losses as an indication of network congestion and activates the internal congestion control mechanism. Consequently, this will result in a degradation in the throughput [6]. Due to the

above drawbacks of TCP, we therefore omitted using it in order to avoid unforeseen impacts on the simulation results. Since the main focus of this thesis is to study the rerouting time and the performance of load balancing, and not the performance of TCP, it is therefore naturally and reasonably to isolate these issues from the potentially disturbing effects of the TCP protocol.

Measurement Parameters
When evaluating the performance of the proposed algorithms, the throughput is used as the main measurement parameter. Even though not presented in the contributed papers, measurement on other parameters were also conducted during the investigations, including packet delay, jitter and different types of packet loss. However, throughput is chosen as the main measurement parameter since it is the most fundamental and important parameter, determining the capacity of the network in transporting data traffic. Especially in the work related to load balancing, the throughput is a more important parameter. Delay and jitter, on the other hand are more important to certain types of applications such as Voice over IP (VoIP), while other applications such as File Transfer Protocol (FTP) and e-mail, are more or less insensitive to these parameters.

Data Processing
The result of each simulation, can either be analyzed using the output trace file or using the Network Animator (NAM) [7] visualization tool. However, we realized early that in order to perform in-depth analyses, it is essential to carefully analyze the output trace files. Usually these files are very large which require a substantial amount of time to inspect and analyze. To ease this task it was therefore necessary to develop a number of tools in Python and Linux shell scripts to parse the trace files and to extract data of interest. The extracted data is then used to make various graphs for further analyses. We also developed different visualization tools such as the *congestion maps* in paper D. The NAM visualization tool was mainly used to verify scenarios and traffic patterns.

Limitations
Although using simulations as a tool to perform studies of complex issues or systems are convenient and efficient, the disadvantage of this approach is the inaccuracy. This is due to the fact that it is very difficult to accurately model the physical world, and the models that are used in most simulators are only an approximation and simplification of it. For example, in ns-2, the transmission range of a omnidirectional wireless interface is represented by a perfect circle with radius r. Radio communication is perfectly received within this range, while no signal can be received at all beyond it. On the other hand, real radios usually have a non-uniform and non-circular radiation pattern [8] [9], with spatial and temporal signal fluctuation that may cause rapid changes in network connectivity and transmission

range. The work in [10] has previously showed that the radio propagation models that are often used in the research of ad hoc networks are inaccurate and may have negative impact on the simulation results. The inaccuracies stem from the usage of simple assumptions in these models, such as the terrain is flat, all radios have equal range, the signal strength is a simple function of distance, and the links are always symmetrical. Furthermore, we also discovered an inaccuracy in the IEEE 802.11 MAC-sublayer implementation of ns-2: while receiving a packet that is sent over the wireless medium, the implemented MAC-sublayer model can only account for interfering signal from one single concurrent transmission, even though there may be multiple concurrent interfering transmissions. In a more accurate model, the resulting interfering signal should be the sum of all concurrent transmissions. Therefore, this MAC-sublayer model may potentially overestimate the received signal strength and consequently, a packet that should have been discarded due to severe interference may instead be successfully received.

Despite the limitations of the simulator in modelling the real world, we still believe it is a good and appropriate tool in performing research. Even though simulation results may not be 100% exact, they can still provide an indication of the behavior characteristics to the system being studied. The work in [11] has demonstrated that with proper adjustment of simulation parameters in ns-2, the accuracy may be quite good with respect to packet delivery ratio and connectivity graphs, but less good with respect to packet delay.

Finally, it is important to be aware that simulation setup and execution comprehend many pitfalls, as reported in [12]. These pitfalls are related to issues such as model validation and verification, correctly setting Pseudo Random Number Generator (PRNG) seed, scenario initialization, and performing statistical analyses. Avoiding these pitfalls are important to achieve more reliable simulation results. We have therefore, throughout the work in this thesis, strived to comply with the recommendations given in [12].

1.4 Paper Contributions

The contributions in this thesis are represented by five published papers, i.e. paper A to E, in peer-reviewed international conferences or journals. Paper A addresses mobility and rerouting time in proactive routing protocols. Paper B addresses the issue of transit routing or load balancing for intradomain traffic. Paper C to E address gateway load balancing or load balancing for interdomain traffic. An overview and summary of the contributed papers is shown in Figure 1.1, in chronological order.

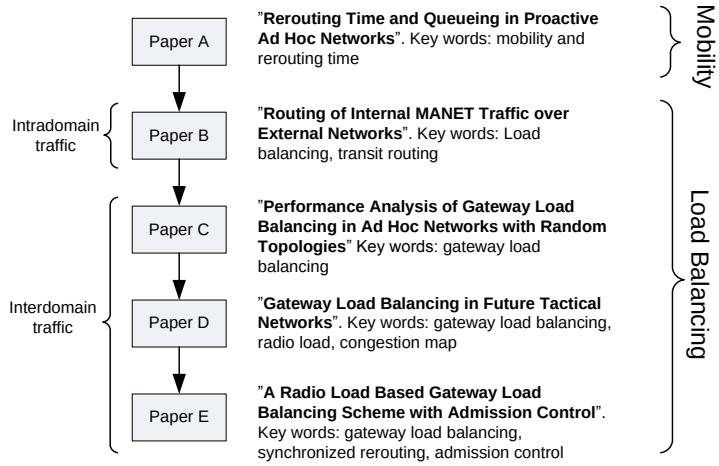


Figure 1.1: Outline of contributed papers.

I. Paper A investigates the factors that affect the rerouting time in proactive networks. Using Optimized Link State Routing (OLSR) as a reference routing protocol, the paper identifies the main reasons for why the rerouting time in many cases is considerably higher than the time actually required to detect a link break. Based on the results of our investigation, a simple model is derived to estimate the expected rerouting time. Furthermore, we also propose a solution that can reduce the rerouting time considerably.

II. Paper B investigates the feasibility of performing load balancing for intradomain traffic in ad hoc networks. The assumption is that there exists a high-capacity backbone network which can be used as a transit network for intradomain traffic. The idea is to route part of the intradomain traffic over this network in order to reduce the traffic load in the ad hoc network and to improve the throughput for certain source and destination pairs. Furthermore, the paper provides an analysis of scenarios in which it is advantageous to perform transit routing. As a result of this analysis, we also propose a cost metric algorithm which facilitates transit routing in ad hoc networks.

III. Paper C investigates the feasibility and the potential benefits of performing load balancing for interdomain traffic. In providing connectivity with external networks such as the global Internet, the gateway node plays an important role. Since all interdomain traffic has to traverse the gateway, there is a risk that this node will be congested and become a bottleneck node. To alleviate this problem, the common solution is to deploy multiple gateways in the network. The paper considers the scenario with two gateways, and ex-

plores the factors that affect the potential benefits of distributing outbound interdomain traffic between these gateways, including the offered load, the level of asymmetry, the gateway distance, the level of spatial reuse (or frequency reuse) and the shape and size of the network area. The study is based on extensive simulations with a large number of randomly generated topologies in order to provide an estimation of the expected average throughput enhancement by performing load balancing.

IV. Paper D is a continuation of the work in Paper C in which we try to find the answers for why the performance of gateway load balancing is considerably high for certain topologies, while it is very poor for others. The paper introduces the concept of *congestion map* which can be used as a tool to analyze how the specific layout of a topology may influence the performance of load balancing. Furthermore, the paper also demonstrates through the proposed RLLB load balancing scheme that it is possible to take advantage of the radio load information provided by the underlying IEEE 802.11 MAC-sublayer for the purpose of performing load balancing.

V. Paper E is a continuation and an improvement of the work in Paper D. The work in this paper addresses the problem of *synchronized rerouting* when load balancing is performed distributedly. This is the main reason why the RLLB load balancing scheme in Paper D performs well in topologies with higher level of asymmetry, but on the other hand, performs poorly when the asymmetry is low. To solve this problem, the RLAC scheme is proposed, which introduces the concept of probability based load balancing. The idea is to let the probability for a local node to select one gateway as default gateway, be a function of parameters such as the difference in hop count and the difference in gateway and bottleneck radio load. A further improvement in the RLAC scheme is that it also performs admission control based on the same radio load information as used in load balancing. While Paper C and D only consider static topologies, Paper E investigates the potential performance enhancement of load balancing in both static and mobile topologies.

1.5 Thesis Outline

This thesis is organized in two main parts. Part I provides an introduction to the work in this thesis and relevant background knowledge. The intention is to make it easier to the reader to have a clearer understanding of the work herein. Part II is the collection of contributed papers in peer-reviewed international conferences or journals.

Part I consists of Chapter 1-6, where Chapter 1 provides an overview of this thesis with respect to the motivation, the objectives, and the research methods used in this work. Chapter 2 gives an introduction to ad hoc networking which is the main research area in this thesis. Various types of ad hoc networks, i.e. Mobile Ad hoc Network (MANET), WMN and Wireless Sensor Network (WSN) are presented along with their typical or envisioned applications. Although ad hoc networking is a promising technology suitable in a number of applications, spanning from civilian to military scenarios, many challenges are yet to be solved. Chapter 2 therefore discusses the main challenges that are characteristic to wireless networks.

Chapter 3 provides a description of the IEEE 802.11 MAC-sublayer, which is by far the present most popular MAC protocol for ad hoc networks in the research community. The MAC protocol is responsible for the coordination of medium access, and enables single hop communication between adjacent nodes. A thorough insight into the inner workings of this protocol is essential for the understanding of the proposed solutions or mechanisms such as the *Adaptive Retry Limit* solution in paper A, and the *radio load metric* in paper D and E.

While the MAC protocol provides single hop communication, the routing protocol on the other hand, makes it possible to perform multihop communication. The routing protocol is the common language that enables the network formation and allows nodes inside the network to communicate with each other. MANET routing protocols differ from traditional routing protocols in which they are tailored for the wireless environment with lower bandwidth and higher level of dynamic. Chapter 4 provides an introduction to some of the most essential topics related to routing in MANETs, including an overview and classification of typical routing protocols for MANETs. A more detailed description on the reactive Ad hoc On-Demand Distance Vector Routing (AODV) and the proactive OLSR routing protocols is also given to illustrate the diversity in routing approaches. The OLSR routing protocol is especially important since it is used throughout the work in this thesis. Furthermore, we also discuss *link failure detection*, and *routing metrics* which are two important aspects related to the rerouting time in paper A and load balancing in paper B-E.

Chapter 5 provides an introduction to the topic of load balancing. A description is given on various types of load balancing in MANETs and the challenges faced when performing load balancing in a wireless environment. The main focus in this thesis is confined to the research of two types of load balancing, namely *transit routing* and *gateway load balancing* addressed in paper B-E. A description on *multipath load balancing* is also given for completeness. However, this type of load balancing is not considered in this thesis since several previous works have showed that the potential benefit of multipath load balancing is rather limited in

single channel wireless networks.

Finally, Chapter 6 provides an overall summary of the work in this thesis. In addition, a more detailed description of the contributions in each individual paper is also given. We round up with a conclusion of this thesis and give some suggestions for further research.

Chapter 2

Challenges in Ad hoc Networks

In recent years, the rapid development and growth of devices with networking capabilities has made the research in ad hoc networks more relevant than ever. The key to the increased popularity and proliferation is the availability of low cost devices and the possibility for rapid deployment. The latter reason is especially attractive in applications such as emergency rescue and military operations. Even though ad hoc networking is a promising technology, there are yet many challenges that need to be solved. In this chapter, we first provide an overview of different types of network technologies classified under the ad hoc network family (Section 2.1). Second, in Section 2.2 to Section 2.6, we will discuss the challenges that prevail in ad hoc networks. The motivation is to provide the reader a broader overview of the ad hoc technology and its typical applications. Furthermore, a better insight into the challenges that ad hoc networks have to face is important in order to bring forth innovative solutions or to be aware of the limitations.

2.1 Overview of Ad Hoc Networks

2.1.1 Mobile Ad Hoc Networks

The history of Mobile Ad hoc Network (MANET) can be traced back to the early 1970s when DARPA developed the PRNET (Packet Radio Networks) [13] [14]. This eventually evolved into the Survivable Adaptive Radio Networks (SURAN) program in the early 1980s [15]. The goal of these programs is to provide packet switched networking that can be used in mobile and hostile environments related to military operations.

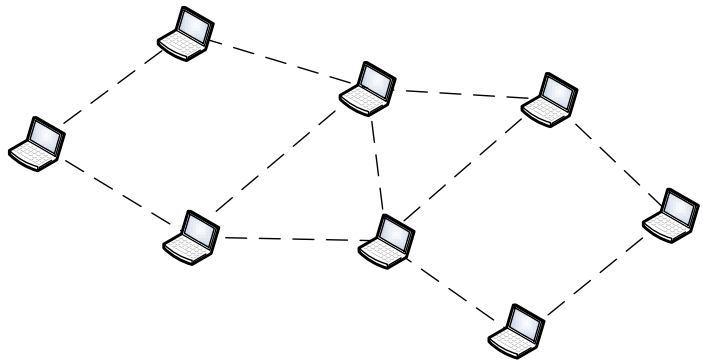


Figure 2.1: Example of a MANET.

A MANET [16] is a collection of mobile nodes, capable to form a network even without the existence of any pre-deployed fixed infrastructure, as illustrated in Figure 2.1. Communication is thus performed over wireless links, using omnidirectional wireless radio interfaces. A MANET is highly dynamic in the sense that the network is formed in a spontaneous and temporary manner. Nodes may randomly join and leave the network or move around. Due to this dynamic nature of the network, there is usually no centralized administration, but instead nodes equally and autonomously collaborate in a distributed manner to form a multihop network. This implies that a node both takes the roles as a host in an end-to-end data communication, or as a router to relay data on behalf of other hosts that may not be within direct transmission range of their destinations. Furthermore, a MANET can operate as a stand-alone network, or be integrated with external networks such as the global Internet through gateway nodes. This is demonstrated in many papers, including paper B-E.

2.1.2 Wireless Mesh Networks

A Wireless Mesh Network (WMN) [17] consists of two types of entities: mesh routers and mesh clients. Mesh routers are usually stationary or have minimal mobility, and they form an infrastructure or backbone for clients that connect to them. These routers are usually equipped with multiple wired/wireless interfaces, which can support various access technologies in addition to the most commonly used IEEE 802.11 technologies. The wireless interfaces can either be omnidirectional or directional antennas. A mesh router is usually not a host in an end-to-end data communication. Rather, it is merely a router responsible for relaying data on behalf of other hosts. Additionally, a mesh router may possess gateway functionality

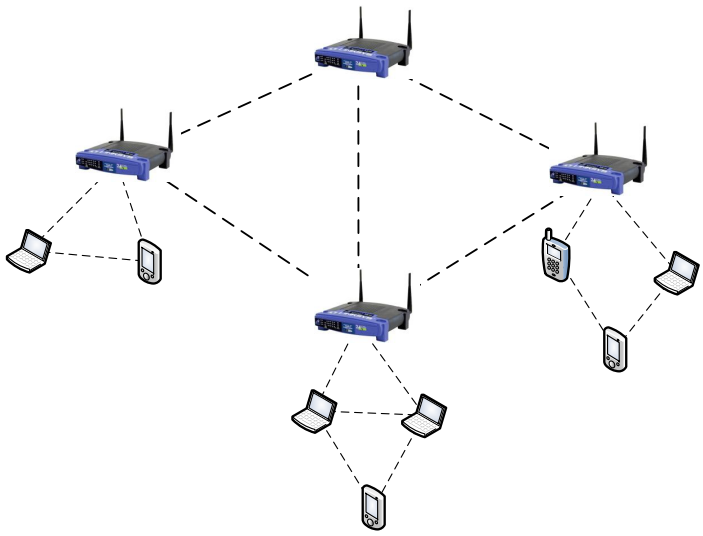


Figure 2.2: Example of a WMN.

that enables integration with other external networks such as the global Internet, cellular networks, Wireless Fidelity (Wi-Fi) networks and so on.

Mesh clients are equivalent to MANET nodes, i.e. they are mobile, and function both as hosts or routers. But in contrast to mesh routers, mesh clients do not have gateway or bridge functions. In addition, mesh clients are usually equipped with only one single wireless interface, which most commonly is an omnidirectional antenna. Furthermore, mesh clients can be of various types of devices such as laptops, pocket PC, Personal Digital Assistant (PDA), phones and so on. An example of a WMN is shown in Figure 2.2, where 4 mesh routers form the backbone of the WMN, while a variety of heterogeneous nodes are the mesh clients.

In contrast to the spontaneous and unplanned characteristic of a MANET, a WMN is partly preplanned. This implies that mesh routers are often deployed in a planned manner to maximize coverage and to form a backbone that usually contains a number of redundant links for increased reliability and robustness. This backbone of stationary mesh routers provides a number of advantages compared to MANETs. First, it gives a certain level of stability and structure to the network. Second, it can alleviate the traffic load from mesh clients, i.e. much of the data traffic both, interdomain and intradomain, can be routed over backbone routers that are normally more powerful and have higher bandwidths. Third, while mesh routers have unlimited external power, mesh clients rely on limited battery power. Thus, by routing much of the data traffic over the backbone nodes, the lifetime of mesh

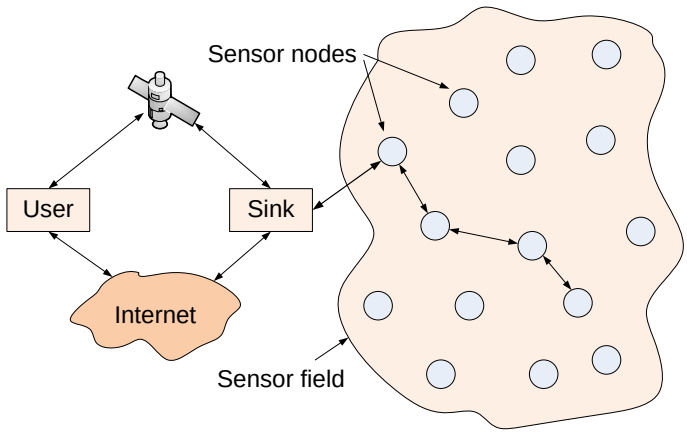


Figure 2.3: Example of a WSN.

clients may be prolonged. The objective of the work in paper B is to take advantage of these benefits, where it is demonstrated that the backbone infrastructure can be used to alleviate traffic load in the ad hoc subnet and to improve the performance of intradomain traffic.

The advantages discussed above make WMNs easier to implement than MANETs. Therefore, an increasing number of experimental mesh network implementations have been deployed in recent years, including the Roofnet [18] experiment of the Massachusetts Institute of Technology (MIT), the BWN-Mesh [19] WMN testbed of the Broadband and Wireless Network (BWN) Lab at Georgia Institute of Technology, and the Magnets [20] project of Deutsche Telekom Laboratories (DTL).

2.1.3 Wireless Sensor Networks

Wireless Sensor Network (WSN) [21] is another type of ad hoc network consisting of wireless sensor nodes that cooperatively monitor physical or environmental parameters such as temperature, pressure, sound, or pollutants. The development of WSN was initially motivated by military applications such as battlefield surveillance and monitoring [22], detection of attack by weapons of mass destruction [23], such as chemical, biological or nuclear weapons. However, the advantages of WSNs over traditional networks resulted in many other potential applications both in industrial and civilian applications, including disaster area monitoring [24], healthcare applications [25], industrial process monitoring and control, environment and habitat monitoring, home automation, and traffic con-

trol [26] [27]. Furthermore, WSNs are especially suitable in inaccessible environments such as volcanoes or at the sea bottom.

A typical example of a WSN is shown in Figure 2.3. The sensor nodes are tiny devices equipped with one or more sensors, a small microcontroller, and a radio transceiver. While the sensors in a node are responsible for measuring physical parameters such as those mentioned above, the processing capacity of the microcontroller allows the nodes to perform local computation on the sensed data. The radio transceiver enables these nodes to exchange data between neighboring nodes or over multihop to the sink node, which is the point of aggregation of the sensed data. The sink node is usually a more powerful node compared to the sensor nodes, and possesses higher capacity in terms of processing power, storage and battery power. In addition, it can also be equipped with a high bandwidth radio link to transmit the collected data, either over the global Internet or via a satellite, to a remote end user.

Sensor nodes in a WSN are often deployed in a large number, randomly and densely distributed over the sensor field with minimal planning. One can for example throw of drop sensor nodes from an aircraft to cover a certain area of interest. Similar to a MANET, nodes in a WSN autonomously collaborate in a distributed manner to form a multihop network without the need of any pre-deployed infrastructure. However, there are a number of differences between these types of networks. First of all the number of nodes in a WSN can be several orders of magnitude larger than the number of nodes in a MANET. Second, sensor nodes are more prone to failure and energy drain, and their battery sources are usually not replaceable or rechargeable. Third, a WSN is data-centric, meaning that the queries in a sensor network are addressed to nodes which have data satisfying some conditions. On the other hand, MANETs are address-centric, with queries addressed to particular nodes specified by their unique address. Most routing protocols used in MANETs cannot be directly ported to WSNs because of the limitations in memory, power, and processing capabilities in the sensor nodes. Besides, the generally non-scalable nature of MANET protocols is incapable to handle the high number of nodes in WSNs.

2.2 Network Connectivity

The topology in an ad hoc network is defined by the set of wireless links that exist in the network. This is again closely related to the relative placement of nodes and the range of their radio transmitters. In ad hoc networks, nodes are usually deployed in an ad hoc manner, without pre-planning. Consequently, the placement

of nodes may be regarded as a random process from which the network topology emerges.

An important aspect in this process is the node density. The previous work in [28] [29] showed that there is a clear relationship between node density and network connectivity. They used the theory of percolation to show that there is a cut-off point in node density, called the critical density. If the density is below this critical point, then there is a risk for network partitioning, where the network topology is divided into smaller unconnected subnetworks. On the other hand, if the density is above this critical point, the network is more likely to be unpartitioned, and there is connectivity between the majority of nodes in the network. However, if the node density is just above the critical point, then the resulting topology may be quite sparse in terms of connectivity. Consequently a node must rely on a few links only, in order to preserve connectivity with neighboring nodes or the network. Hence, a link break may have severe impact on the network connectivity in a sparse network. In contrast, a dense network has usually many redundant links and paths between the majority of the node pairs, and a link break does not affect the network connectivity to any extent. The redundancy provides a number of alternative paths to circumvent a broken link. From this, it is apparent that in sparse networks, it is more challenging for the routing protocol to dynamically adapt to changes in topology caused by link failures, than in dense networks. Hence, the node density directly affects the ability of the routing protocol in adapting to topology changes, and at the same time maintaining connectivity. When setting up scenarios for evaluating proposed algorithms/mechanism in paper B-E, care was therefore taken that a suitable number of nodes is deployed in order to ensure connectivity and redundancy.

Furthermore, the node density has meaning only when treated relative to the nodes' radio transmission range. If the radio range is reduced while the number of nodes per unit area remains the same, then the connectivity in the network is also reduced. This implies that variations in node density as well as the radio range, are both affecting the network connectivity. The variations in radio range are for example caused by random variations in the environment such as topography or weather condition. Variations in available power in each node may also be a second reason. This can for example be induced intentionally in order to save battery power or to reduce radio interference between nearby nodes [30] [31].

Another issue that affects network connectivity is the *communication gray zone* problem [32], in which unicast data packets cannot be exchanged even though link sensing with broadcast control messages indicates neighbor reachability. This problem is rooted in the difference in transmission range between broadcast and unicast data packets. In IEEE 802.11, a broadcast packet is always transmitted at

the basic data rate, while a unicast packet is normally transmitted at higher rates. This is due to the fact that broadcasting is more unreliable than unicasting, since it is not protected by the retransmission mechanism at the link layer as in the case of unicasting. Therefore broadcast packets are transmitted at the lowest data rate to increase reliability. However this also increases the radio range of broadcast packets. This difference in transmission range between a broadcast and a unicast packet is the main reason for communication gray zones to occur, resulting in establishment of links that are potentially unusable for unicasting data packets. A possible solution to this problem is to use Signal to Noise Ratio (SNR) as a measure to differentiate and discard "weak" control packets.

2.3 Node Mobility

In a MANET, nodes move around in a arbitrary manner. The presence of node mobility causes frequent link breaks and formation of new links in the network. As a result, the network topology may continuously change. The challenge that arises when the topology is changing, is the difficulty in keeping the routing table up-to-date to correctly reflect the actual view of the topology. This is due to the fact that routing protocols generally need a certain amount of time to detect link breaks. For example, the work in paper A shows that proactive routing protocols can use up to 6 seconds to detect a broken link (with default Hello intervals of 2 seconds), in addition to the time needed to commence rerouting. Consequently, the perceived topology of the routing protocol will usually lag behind compared to the actual network topology.

An important parameter is the average node velocity, which determines how fast the network topology changes over time. The faster the velocity is, the more difficult it is for the routing protocol to keep track with the changes. This is one of the major challenges that we experienced in paper E, i.e. as the velocity increases, it is correspondingly more difficult for the routing protocol to perform load balancing. One way to improve this is to reduce control traffic interval as the node velocity increases. However, this solution entails a significant increase in the amount of control traffic overhead.

Even though it is possible to keep up with the changes in the topology, the routing protocol cannot predict how the topology will change in the near future. Packets that are forwarded on the basis of the current view of the topology may still be discarded en-route to the destination due to unforeseen changes in the topology. Furthermore, as a consequence of frequent link breaks, the effective capacity in a mobile network is usually lower than in a static topology.

2.4 Unidirectional Links

Most research related to ad hoc networks is based on the simplifying assumptions that all wireless links in the network are bidirectional (also called symmetrical): if node *a* can hear node *b*, then node *b* can also hear node *a*. However, in the real world, this is not always true. Unidirectional links (also called asymmetrical) do exist, and may occur due to various reasons. First, heterogeneity of receiver and transmitter hardware may lead to differences in radiation patterns and radio range. Second, power control or topology control algorithms may be used to adapt the transmission power to the remaining energy reserve, or to reduce the level of interference in the network [33] [34] [35]. Third, unidirectional links may also result from interference due to concurrent transmissions. The level of interference may be different at node *a* and *b*, so that one of them cannot temporarily receive data from the other [36]. The negative impact on network performance due to the presence of unidirectional links is documented in various works [37] [38].

2.5 Network Capacity

The network capacity in terms of bandwidth or throughput is dependent on the applied underlying physical layer. Most work related to ad hoc networks are based on the IEEE 802.11 standards which can, in theory, provide a throughput up to 150 Mbps. Despite the high data rates specified by these standards, the capacity is in reality much lower, especially in multihop networks. This is due to the fact that wireless communication must share a common medium. Therefore, multiple concurrent transmissions may potentially result in interference and disruption, unless the transmitters are located far enough from each other. The latter condition provides spatial separation and allows concurrent transmissions to occur without destructively interfering each other. This is commonly referred to as *spatial reuse* [39]. Higher level of spatial reuse can increase the throughput in the network, but may however, also increase the probability for packet collisions, as shown in paper C. Hence, the trade-off between these two conflicting mechanisms needs to be considered.

The analysis in [40] showed that the total end-to-end capacity in multihop wireless networks is correlated with the number of nodes *n* in the network, and is roughly $O(\frac{n}{\sqrt{n}})$. This implies that the per-node throughput capacity is just $O(\frac{1}{\sqrt{n}})$, and approaches zero as the number of nodes increases. Thus, in order to ensure a reasonable amount of throughput capacity for each node, it is recommended to keep the network size to a small number of nodes. This shows that scalability in

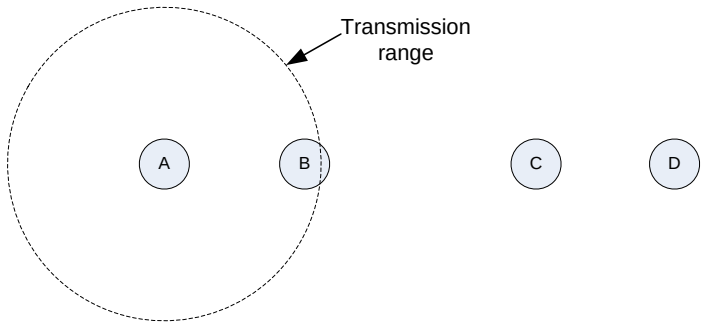


Figure 2.4: The hidden node and exposed node problems.

wireless multihop networks is not just an issue seen from the perspective of routing layers, but is also an issue with respect to the limitations in throughput capacity provided by the underlying link and physical layer.

Furthermore, the analysis in [41] showed that one of the reasons for limited capacity in ad hoc networks is due to the *unfairness* problem. They used a chain topology with one single traffic flow, where the leftmost node originated traffic to the rightmost node through hop-by-hop forwarding. It was observed that the originating node was allocated more access to the medium and therefore injected more packets into the network than subsequent nodes could forward. Consequently, packets were discarded at subsequent nodes. The reasons for this unfairness in medium access time are due to the decentralized medium access control, the binary exponential backoff scheme and the difference in the amount of competition experienced at each node, i.e. the outermost most nodes experienced less interference compared to the nodes in the centre of the chain.

2.6 Medium Access

Medium access is a difficult issue in ad hoc networks due to the dynamics in network topology and the lack of centralized control. Using Time Division Multiple Access (TDMA) or Frequency Division Multiple Access (FDMA) are rather complex since there is no centralized control as in cellular networks. Code Division Multiple Access (CDMA) is difficult to implement due to node mobility and the consequent need to keep track of the frequency-hopping patterns and/or spreading codes for nodes in the time-varying neighborhood. The contention based medium access scheme as in IEEE 802.11 is currently the favorite.

In IEEE 802.11, a node must basically sense the medium before commencing a transmission in order to avoid collision. However, this approach can only to a certain extent reduce the probability for interference and collisions. It can not entirely eliminate it. The well known phenomena of *hidden node* and *exposed node* are examples of problems that carrier sensing cannot handle. Consider Figure 2.4, where four nodes are placed along a line, and we assume circular and uniform radio range for all nodes. The phenomenon of hidden node [42] takes place when node C also sends traffic to D, while A is sending to B. This happens because node C, which is the hidden node to A, cannot sense the transmissions from A (and A cannot sense the transmissions from C), and therefore may commence sending traffic. Consequently, the result is interference and packet collisions at node B. During the work in paper C, we observed that this problem is especially severe when the traffic load in the network is high. A possible means to alleviate this is to increase the sensing range, which will result in lower probability for packet collision and improved performance. Alternatively, as will be discussed later, the RTS/CTS mechanism can also be used to alleviate the hidden node problem.

The exposed node problem is more or less opposite to the hidden node problem. Suppose B sends traffic to A. At the same time, C has also traffic to send to D. However, since C can hear the signals from B and interprets the medium as busy, C therefore defers sending traffic to D. In reality, C could send traffic to D without interfering with the transmission from B to A. Consequently, this "misinterpretation" results in non-optimized utilization of the medium and lower network performance.

Chapter 3

IEEE 802.11

IEEE 802.11 refers to the set of standards for Wireless LANs (WLANs). The original version of the standard (IEEE 802.11), was first released in 1997 and supported data rates of 1 and 2 Mbps. Later on new amendments were added, such as IEEE 802.11a, b, g and n supporting data rates up 150 Mbps. The IEEE 802.11 standards define both a Physical-layer and a MAC-sublayer (Figure 3.1), where the Physical-layer supports different modulation techniques such as FHSS, DSSS and OFDM, and operates at the 2.4 and 5 GHz frequency bands.

3.1 The MAC-sublayer

While the routing layer provides multihop communication, the MAC-sublayer provides single hop communication. IEEE 802.11 MAC layer is by far the most popular MAC protocol in ad hoc networks. The IEEE 802.11 MAC-sublayer implements the access control mechanism that enables nodes to access and share a common physical medium. The standard defines two types of operation modes, Distributed Coordination Function (DCF) and Point Coordination Function (PCF), where DCF is the basic mode of operation for ad hoc networks, while PCF is an optional operation mode that is suitable for infrastructure-based networks.

DCF is a distributed and contention-based medium access method that is based on Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) combined with a random backoff procedure. CSMA protocols are well-known in the industry, where the most popular variant is Carrier Sense Multiple Access with Collision Detection (CSMA/CD) that is used in the Ethernet or wired LAN. In DCF, when a node has a data packet to send, it must first sense the medium to determine whether

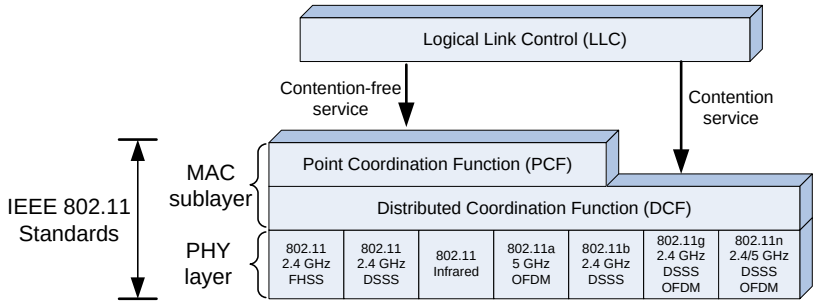


Figure 3.1: The IEEE 802.11 Standards.

it is idle or busy. If the medium is sensed idle for a time equal to Distributed Inter-Frame Space (DIFS), the transmission of the data packet may proceed. Upon receiving the data packet, the receiving node must wait for a Short Inter-Frame Space (SIFS) before it sends a MAC-sublayer Acknowledgment (ACK) back to the sender node. When the sender node receives the ACK, the transmission is completed, and a new contention period begins. To contend for the medium, each node must generate a random backoff time from the Contention Window (CW). This is completed by drawing a random number k from a uniform distribution in the interval [0,CW], where CWmin $\leq$ CW $\leq$ CWmax. The initial value of CW is equal to CWmin. The actual backoff time is equal to the random number multiplied by the slot time, i.e. k·SlotTime, where the value for the slottime as well as CWmin and CWmax are dependent on the Physical-layer type. Each time the medium is idle for a period of DIFS, the backoff phase is entered where the backoff time is gradually decremented slot by slot. If the medium becomes busy during the backoff phase, the backoff time count-down is suspended. The effect of this backoff procedure is, if multiple nodes are contending for the medium, then the node that has generated the smallest backoff time will win the contention. The node may then begin to transmit if there is a pending packet. The main purpose of the backoff time combined with the "sense before transmit" approach, is to reduce the probability for packet collisions at the point of time where collision is most likely to occur, i.e. just after the medium becomes idle following a busy medium, is when the probability for collision is highest. This is because multiple nodes may have waited for the medium to become available again. Since a node must go through a backoff procedure after having transmitted a packet, the medium access mechanism also provides long term fairness to access the medium.

The MAC-sublayer provides a certain level of reliable unicast packet transmission through the usage of explicit acknowledgement with an ACK from the receiver to

the sender. The lack of such an ACK after an ACKTimeout interval, may indicate to the sender that an error has occurred. The error can indistinguishably be a result of a collision either to the data packet or the ACK message, or it can also be a result of a temporary link failure or a persistent link break. Either way, each time a transmission is failed, a retransmission procedure is invoked. In this process, the data packet is scheduled for retransmission, in which a new backoff time is generated. However, in order to reduce the probability for collision, after each failed transmission, the CW is exponentially increased until the threshold CWmax is reached. For example, the CW can sequentially be increased (using the equation $2^i - 1$, where *i* is an integer) from 7, 15, 31, 63, 127, and 255, where CWmin=7 and CWmax=255. After a successful transmission, the CW is again reset to CWmin.

Furthermore, a retry counter is maintained to account for the number of retransmissions that the current packet has experienced. Each time a packet is retransmitted, the retry counter is increased by one. After a successful retransmission the retry counter is again reset to 0, while the CW is reset to CWmin. On the other hand, if the number of retransmission has reached a predefined threshold value, the packet is discarded, and the retry counter and the CW are reset. The MAC-sublayer is then ready to handle the next pending packet in the queue.

In order to determine the state of the medium and to avoid collisions, the standard provides two carrier sensing mechanisms, a physical and a virtual. The medium is considered as busy whenever either mechanisms indicate that the medium is busy, otherwise the medium is considered idle. The physical carrier sensing mechanism is provided by the underlying Physical-layer and is used to detect medium activity and avoid collisions at the sender node, but it cannot prevent collisions from occurring at the receiver node due to problems such as *hidden node*, as discussed in Section 2.6. In order to overcome this problem, the MAC-sublayer virtual carrier sensing mechanism provides an optional hand-shake mechanism, using Request To Send (RTS) and Clear To Send (CTS) control messages, to schedule a data packet transmission. Upon hearing one of these messages, either the RTS from the sender node or the CTS from the receiver node, nearby nodes can be made aware of the scheduled transmission and can thus defer any pending transmissions, even if they are either outside of the sender's or the receiver's radio range.

Even though the RTS/CTS mechanism may alleviate the probability for collisions, the disadvantage of using this mechanism is the potential for reduced throughput due to increased overhead. Besides, this mechanism only attributes to reducing the effect of hidden node. It cannot eliminate the problem if the distance between the sender and receiver is larger than 0.56 times the radio range, assuming a minimum SNR of 10 for successfully receiving a packet. The reason is that the power level needed for interrupting a transmission is much smaller than that of success-

fully delivering a packet as explained in [43]. Due to these reasons, the RTS/CTS mechanism has therefore not been used in our studies.

Chapter 4

Unicast Routing Protocols for MANETs

A routing protocol provides a set of rules and regulations for how nodes communicate with each other. A major part of the work in this thesis is concerned about enhancing the functionality of the routing protocol in order to optimize network performance. This chapter provides an introduction to some of the most essential topics related to routing in MANET. First, an overview and classification of typical routing protocols for MANETs are given in Section 4.1. Section 4.2 and 4.3 provide a more detailed description of two selected routing protocols, namely the reactive AODV and the proactive OLSR protocol. These protocols are among the most cited, and variants of these concepts can be found in many other routing protocols. The description of OLSR is however more important since it has been used throughout the work in this thesis. Furthermore, Section 4.4 discusses an important aspect related to routing protocols, i.e. the various mechanisms used by routing protocols to detect link breaks. This topic is especially important in terms of the work in paper A. Finally, Section 4.5 provides an overview of common routing metrics in MANETs, which are a vital component in performing QoS routing or load balancing.

4.1 Overview and Classification of MANET Routing Protocols

To enable communication within an ad hoc network, a routing protocol is required to establish routes between source and destination pairs. A routing protocol may be

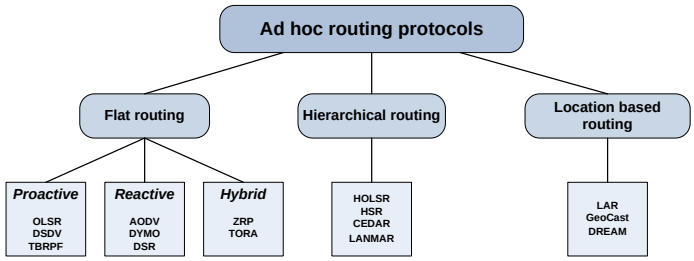


Figure 4.1: Classification of MANET routing protocols.

regarded as the language that nodes use to communicate with each other in order to exchange routing information about the topology. This implies that all nodes belonging to the same network must speak the same language, i.e. they must run the same routing protocol.

In the literature, a number of different ad hoc routing protocols have been proposed. A majority of these proposals originate from the Internet Engineering Task Force (IETF) working group for mobile wireless networks [44]. Due to the diversity in applications that is feasible with the ad hoc network technology, including military tactical networks, emergency rescue networks and community networks to name a few, each proposed routing protocol is therefore designed and optimized for a specific task or type of application. In many cases, the routing protocol is designed to optimize a specific parameter such as bandwidth, latency, mobility, scalability, battery consumption, etc.

Ad hoc routing protocols may be classified into three main categories [45] as shown in Figure 4.1. The first category is flat routing where all nodes are organized in one logical level. All nodes in the network have equal roles and importance with respect to route establishment and packet forwarding. Due to this reason, a flat address scheme is usually adopted to identify nodes in flat routing protocols. Flat routing may be further divided into three classes which include *proactive*, *reactive* and *hybrid* routing protocols.

In proactive routing protocols such as OLSR [46], DSDV [47], TBRPF [48], routes to all destinations in the network are established and maintained prior to when they are actually needed. This has the advantage that when a node needs to forward a packet to a destination in the network, the route is immediately available. Thus there is no delay in route discovery. Proactive routing protocols have additionally strong support for QoS, since QoS states or link quality information can be easily piggybacked with the routing messages. The process of keeping routes

and states up-to-date requires regular exchange and dissemination of neighbor and topology information. This however incurs a constant amount of overhead, consuming bandwidth, battery power and computation time, even though the routes are not always needed. If the network size is large or the level of mobility is high, the overhead in terms of control traffic and routing table storage may be of a substantial size. Hence, proactive routing protocols do not scale very well. On the other side, the overhead is not affected by an increase in the fraction of nodes acting as traffic sources, since routes to all destinations are anyway maintained by the routing protocol.

Reactive routing protocols (also known as *on demand* routing protocols) are another class of flat routing protocols, specifically developed for MANETs. Examples of frequently cited reactive routing protocols include AODV [49], DYMO [50] and DSR [51]. The focus of this class of routing protocols is to preserve network resources with respect to bandwidth, routing table storage and processing time. With the reactive approach, routes are neither maintained nor established until they are actually needed. The establishment of a route is usually initiated by flooding a Route Request (RREQ) control message, and eventually responded with a unicast Route Reply (RREP) message from either the destination node or an intermediate node with a valid route to the destination. Reactive routing protocols therefore incur a substantially lower constant overhead compared to proactive routing protocols and are thus also more scalable. On the other hand, the disadvantage is the route discovery latency that can be high if the destination node is many hops away.

In hybrid routing protocols, the advantages of both the proactive and reactive routing strategies are combined in order to make a compromise between control traffic overhead and route discovery latency. The idea is to utilize a proactive routing protocol in the local neighborhood, i.e. for destinations inside a predefined zone with radius of k hops from the local node. On the other hand, routes to destinations beyond the predefined zone are queried using a reactive routing protocol. Examples of hybrid routing protocols include Zone Routing Protocol (ZRP) [52] which was the first hybrid routing protocol introduced by Haas in 1997, and Core Extraction Distributed Ad Hoc Routing (CEDAR) [53].

When the network size is very large and beyond a certain threshold value, flat routing protocols become unsuitable due to the excessive amount of overhead in terms of control traffic as well as processing time. In such case, hierarchical routing protocols can be used as an alternative. Hierarchical routing protocols organize nodes into a hierarchy of clusters based on their relative proximity to one another. For each cluster, a cluster head is elected to be a local coordinator for transmissions within the cluster. Cluster heads at a lower level become members of the next higher level cluster. This process is performed recursively to provide a mul-

tilevel hierarchy of clusters. The motivation is to reduce the size of the routing table and thus achieve better scalability. In contrast to flat routing protocols, hierarchical routing protocols usually apply a hierarchical addressing scheme in which the identity of a local node can for example be the sequence of addresses of the associated cluster head nodes, from the top hierarchy to the node itself. Examples of hierarchical routing protocols include Hierarchical State Routing (HSR) [54], Hierarchical Optimized Link State Routing (HOLSR) [55] and Landmark Ad Hoc Routing (LANMAR) [56]

In location based routing protocols, the position information provided by for example Global Positioning System (GPS) [57] is exploited to perform directional packet forwarding. In contrast to for example flat routing protocols, establishment or maintenance of routes are not required. Based on the knowledge of the expected position to the destination node, a source node can forward data traffic directionally towards the destination. The advantage of location based routing protocols is lower overhead with respect to control traffic. The disadvantage is the dependency on position information, which is not guaranteed to be available everywhere such as inside a building. Examples of location based routing protocols include Location-Aided Routing (LAR) [58] and Distance Routing Effect Algorithm for Mobility (DREAM) [59].

4.2 AODV

AODV is one of the most cited and studied reactive routing protocol for MANETs, and has also been standardized by the IETF [60]. In AODV [49], when a node needs to send data traffic to a destination node that it has no route to, it then initiates a route discovery process by flooding a RREQ message. Upon receiving this message, an intermediate node refloods the message to its neighboring nodes. The process of reflooding the RREQ is repeated multiple times, covering an increasingly larger area, until either the destination or an intermediate node with a valid route to the requested destination is reached. During the route discovery process, each node that receives a RREQ message sets up a *reverse route* back to the originator of the received message, along the path that the message has traversed.

The destination node or an intermediate node with valid route to the destination will eventually receive the RREQ message. The node then responds with a RREP message containing the latest destination sequence number. This is one of the distinguishing features of AODV, i.e. the usage of the destination sequence number to ensure loop freedom. Duplicate RREQ messages, identified by their sequence number and originator address, that have taken alternative paths and are received at

a later point of time are discarded. This implies that only the first RREQ message received is processed and responded, based on the assumption that this message has traversed the best and shortest path, and therefore has arrived first. The RREP message is unicasted along the reverse route that was set up during the flooding of the RREQ message. Upon receiving the RREP message, an intermediate node sets up a *forward route* towards the destination node. The message is then forwarded to the next hop along the reverse route towards the source node. Eventually when the source node receives the RREP message, the path between source and destination is now established and ready for sending data.

During data transmission, if there is a link break anywhere along the established path, a Route Error (RERR) may be sent upstream to warn the source node. The source node can then reestablish a new route by initiating a new route discovery process. Alternatively, an intermediate node upstream to the link break may try to perform a local link repair by flooding a RREQ querying a new route to the destination node.

Being a reactive routing protocol, the advantage of AODV is lower control traffic overhead and routing table storage. Consequently it is more scalable than proactive routing protocols, since only needed routes are stored and maintained in the routing table. However, the reactive approach incurs a certain amount of delay in route discovery. Besides, the route discovery process often involves network-wide flooding of the route discovery message, which unfortunately incurs a substantial waste of scarce network resource, especially if the requested destination is only a few hops away from the initiator of the message. To alleviate this, the route discovery process is usually performed in a progressive manner where the search area is incrementally expanded until the destination node is found. However, this optimization, known as *expanding ring search*, may further increase the average latency of the route discovery, since multiple discovery attempts and timeouts may be needed before discovering a route to the destination node. Furthermore, due to the overhead in route discovery, reactive routing protocols do not scale very well, especially if there are many short lived traffic sources in the network and if the distance between source and destination pairs is far. The same also applies if the mobility in the network is high, since link breaks may result in frequent re-initiation of route discovery.

4.3 OLSR

The OLSR routing protocol is an example on proactive routing protocol for MANET. This routing protocol is brought forward by the MANET WG starting from 1998.

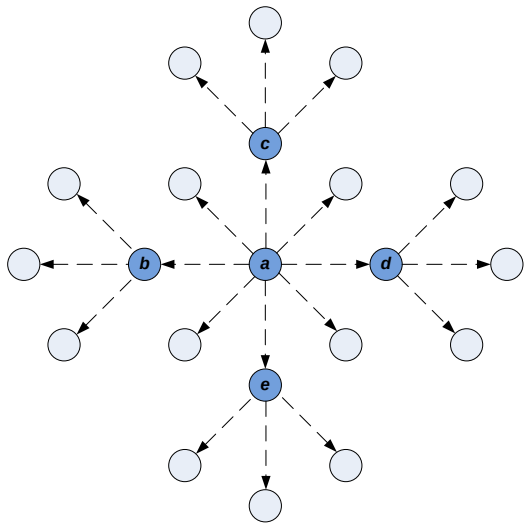


Figure 4.2: An example of MPR selection in OLSR.

It is inspired by an important class of routing algorithms for wired networks, namely link state routing, to which the widely used Internet routing protocol Open Shortest Path First (OSPF) [61] belongs to. OLSR is an optimization of the original link state algorithm and tailored to the requirements for MANETs. An OLSR version 2 [62] is currently being developed, implementing a more standardized packet format. Key differences are the flexibility and modular design using shared components: packet format packetbb, and neighborhood discovery protocol Neighbor Hood Discovery Protocol (NHDP). These components are being designed to be common key components among next generation IETF MANET protocols.

In OLSR, all nodes periodically broadcast Hello messages within their 1-hop neighborhood to perform link sensing and neighbor detection. To illustrate this process, consider the example in Figure 4.2. Assume that node *a* initially broadcasts an empty Hello message to announce its existence. When for example node *b* receives this message, it records that *a* is a asymmetric neighbor. This means that the link has only been verified to be functional in one direction, i.e. from *a* to *b*. Next, *b* broadcasts a Hello message announcing *a* as its asymmetric neighbor. When *a* receives this message, *b* is then considered as a symmetric node to *a*, since the link between *a* and *b* has been verified to be symmetric, i.e. the link is functional in both directions. Suppose that *a* has also received Hello messages from for example *c, d* and *e* announcing that they also have heard *a*'s initial Hello message and therefore declare *a* as their asymmetric neighbor. When *a* again broadcasts a new Hello message, *a* can now declare that *b, c, d, e* are symmetric neighbors.

Upon receiving this message, *b* now knows that *a* is a symmetric neighbor. In addition, *b* also knows about the 2-hops neighbors *c, d, e,* that can be reached via *a*. Thus, the exchange of Hello messages does not only allows a node to perform link sensing, but also enables it to obtain knowledge about the 2-hops neighborhood.

One of the most characteristic concepts in OLSR is the usage of Multi Point Relay (MPR) nodes to minimize control traffic overhead flooded in the network. Each node in the network selects an MPR set from among its 1-hop symmetric neighbors. This set is by default selected in such a way that the number of MPR nodes is kept as low as possible and at the same time satisfies the condition that all strict 2-hops neighbors are reachable via these MPR nodes. The essential point of the MPR selection process is the formation of a Connected Dominating Set (CDS) consisting of MPRs which can be used to perform network wide flooding in a much more efficient way. That is, when a control message is broadcasted throughout the whole network, only nodes selected as MPRs, for the node in which the message is received from, can forward the message. This technique of MPR flooding substantially reduces the message overhead compared to the classical flooding mechanism where all nodes are responsible for retransmitting broadcast packets. Referring to the example in Figure 4.2, it is sufficient for node *a* to select the nodes *b, c, d,* and *e* as its MPRs. When *a* performs a network wide broadcast, only these selected nodes need to forward the message from *a*. The transmissions from *a* and its MPRs are sufficient to disseminate the message to all nodes in the network.

While a Hello message is used to exchange local link state information and to perform neighbor detection in the local neighborhood, a Topology Control (TC) message is used to disseminate topology information to all nodes in the network. Only nodes selected as MPRs originate TC messages. Thus the overhead is further reduced since the number of control messages flooded in the network is minimized. In addition, an MPR by default only reports the links between itself and its MPR selectors (i.e. the nodes that have selected it as MPR) in the TC message it originates. Hence, the overhead is reduced even more since only partial link state is reported. This implies that from a certain node's point of view, the network topology beyond the 2-hops neighborhood consists only of the set of links between the MPRs and their MPR-selectors. Consequently, the constructed logical topology has fewer redundant links, but is still fully connected. The drawback is that it is less resilient to link failures.

The neighbor and topology information are updated periodically through the exchange of Hello and TC messages, and enables each node to compute an up-to-date routing table to all known destinations in the network. The computation is performed using Dijkstra's shortest path algorithm, and the routes are thus optimized with respect to the hop count. The advantage of this metric is that the

number of transmissions needed to send a packet from a source to a destination is minimized, and is thus optimal with respect to the usage of network resources, i.e. bandwidth and energy. On the other hand, the shortest path to a given destination is not always the best path when it comes to other parameters such as throughput, end-to-end delay or reliability. Hence, the work in paper B to E is therefore concerned about the potential of using different types of routing metrics in order to achieve load balancing and higher performance in terms of throughput.

In addition to the message types discussed above, OLSR also defines other types of messages, i.e. Host and Network Association (HNA) and Multiple Interface Declaration (MID) messages. The HNA message can be used by a gateway node to announce about its connectivity to external network domains such as the global Internet. This is for example used in Paper B, where the access points and gateway nodes periodically disseminate HNA messages to announce their presence and connectivity to external networks. The MID message format is used to support multiple interfaces. The additional information provided by the MID message is needed to map different interface addresses of a node to its main address.

Being a proactive routing protocol, OLSR has the advantages and disadvantage that are common in this family of routing protocols and mentioned earlier in the introduction of this section. However, OLSR has gained much attention and popularity since its appearance, due to its simplicity and interesting features such as MPR flooding and the ability to detect bidirectional links. Consequently, several OLSR implementations exist for use under different platforms such as Windows, Mac OS and Linux [63], [64]. Furthermore, OLSR has been chosen and used in the real-life olsrexperiment.net experiment in Berlin [65].

4.4 Link Failure Detection

In a MANET, the topology of the network usually changes over time either due to node mobility or because nodes join or leave the network. In addition, packet loss due to interference or temporal signal fluctuations may also affect the perceived view of the topology. One of the biggest challenges of a routing protocol is to maintain an up-to-date view of the network topology and to quickly detect and react to changes in the topology. This is important and necessary in order to correctly route data traffic. Otherwise there is a potential risk for wasting scarce resources and suffering from packet loss if the traffic is routed incorrectly. Generally, there are three approaches to detect link failures as discussed below:

Hello messages
The most common approach is based on regular exchange of Hello messages, which is actually used in both reactive and proactive routing protocols such as AODV and OLSR. The assumption behind this approach is that the reception of such "polling" packets indicates link availability with the originator of the Hello messages. This approach has proven to work well in wired networks which suffer from few packet losses and topology changes. In order to maintain a link between two neighboring nodes, it is required that these nodes must exchange Hello messages to each other at regular intervals. Failing to receive three successive Hello messages from a neighbor is interpreted as a sign of link failure to that neighbor. This approach is utilized in order to minimize the potential risk for erroneously invalidating a link due to temporal loss or link fluctuations. The disadvantage is that it takes a longer time to detect a link break. This issue is closely related to the work in paper A, in which it is shown how this latency in detecting the link break combined with other factors (such as the length of the link layer queue, and the retransmission mechanism at the MAC-sublayer), may result in a rerouting time that considerably exceeds the time used to detect the link break.

Link layer feedback
The link layer feedback can be alternatively used to detect a link break. When the MAC-sublayer fails to transmit a packet to a next hop neighbor node and after a number of unsuccessful retransmissions, the packet is discarded. A callback is then made to the upper layer to explicitly notify the routing protocol about the failed transmission. This notification from the MAC-sublayer indicates that the link to the neighbor node is broken, and thus allows for a much faster detection time than in the Hello message approach. However, the disadvantage of this approach is that temporal link fluctuations can be misinterpreted as a persistent link break. The work in [66] shows that the link layer feedback approach works better than the Hello message approach at low network load. However, if the network load is high, the amount of incorrect link failure detections increases dramatically and results in lower throughput performance.

Implicit acknowledgement
The third approach is implicit acknowledgement, where a node, after a packet transmission to the next hop on the route, continues to listen to the channel in order to overhear whether the next hop node forwards the packet further or not. Absence of such a forwarding within a predefined time interval, indicates that the transmission to this neighboring node has failed and the link is therefore probably broken, unless the next hop node is the destination node itself. The drawback of this approach is the requirement for the wireless network interfaces to support operation in promiscuous mode, which is extremely energy expensive. Due to this

reason the implicit acknowledgement approach has not gained wide attention in the ad hoc network research.

4.5 Routing Metrics

Routing metrics are criterions or algorithms used by a router to make routing decisions. The most common metric is the shortest path metric, which solely rely on the hop count in performing routing. In the following subsections we will provide an overview of other routing metrics common in MANETs. Among these, the radio load metric is the most relevant which is utilized both in paper D and E.

4.5.1 Per hop Round Trip Time (RTT)

The RTT metric is based on measuring the round trip delay between neighboring nodes using unicast probe packets [67]. To measure the RTT, a node periodically sends a probe packet containing a timestamp to each of its neighbors. Upon receiving a probe packet, each neighbor node immediately responds with a probe-ack packet echoing the timestamp. This enables the sender of the probe packet to calculate the round trip time to each of its neighbors. The exponentially weighted moving average method is used to avoid rapid fluctuations by smoothing out the RTT measurement. Based on per-hop RTT measurement, a routing protocol selects a routing path with the least sum of RTTs of all links on the path.

The RTT metric is designed to measure several aspects related to the quality of a link. First of all, if the link between two neighboring nodes is busy due to traffic load, then either the probe packet or the probe-ack will be subjected to queueing delays, resulting in increased RTT delay. Second, if nodes in the vicinity are transmitting, then the probe packet or the probe-ack will experience delay due to channel contention, which again will result in higher RTT delay. Third, if the link between two nodes is lossy, then either the probe packet or the probe-ack is likely to experience packet loss. In such cases, the retransmission mechanism at the IEEE 802.11 MAC-sublayer will attempt to retransmit a number of times in order to correctly deliver the packet. However, this will take some time which will result in increased RTT delay.

To summarize, the RTT metric is capable to determine the quality of a link with respect to traffic load, queueing delay, the packet loss ratio, and the level of contention in the surrounding neighborhood. Hence, based on this metric, the routing protocol can avoid using highly loaded or lossy links. The disadvantage of this

metric is however the increased overhead, since every pair of nodes are required to regularly probe each other. Second, since RTT is a load-dependent metric, which implies that it is rather sensitive to traffic load and queueing delays, it may therefore lead to route instability (or route flapping). This is a well known problem, and is also called *self-interference* by the authors in [68]. If a separate queue is assigned to the probe packets, then it is possible to accurately measure the link quality but in return cannot reflect the traffic load. Finally, RTT does not explicitly take link data rate into account, due to the small size of probe packets. Larger probe packets could be used to achieve this, but would at the same time increase the overhead.

4.5.2 Per-hop Packet Pair Delay

The packet pair delay is a well known technique in wired networks for measuring link quality [69]. A node periodically sends two back-to-back probe packets to each of its neighboring nodes. The first probe is small and the second is large. Upon receiving the probe packet pair, the neighboring node measures the delay between the receipt of the first and the second packet. The delay is then reported back to the sending node. This delay is a measure of the link quality in terms of loss rate, bandwidth and traffic load in the vicinity of the sending node. The advantage of using the packet pair technique over RTT is that it does not suffer from distortion by queueing delays, since both probe packets will experience the same delay. In addition, using a larger second probe packet makes it possible to take the link bandwidth into account in the measurement. On the other hand, the disadvantage is higher overhead, since two packets are sent to each neighbor instead of one, and the second probe packet is larger. Furthermore, [70] reported that even this technique is not completely immune to the self-interference phenomenon. However, it is less severe than in the case of RTT.

4.5.3 Expected Transmission Count (ETX)

The ETX metric predicts the expected number of transmissions, including retransmissions, required to send a unicast packet over a link [71]. The prediction is based on measurements of the loss ratio of broadcast packets in both directions of a wireless link. To calculate the ETX value, each node regularly broadcast probe packets with an average period of T seconds. Neighboring nodes count the number of received probe packets within a time interval of W seconds. The loss ratio in

one direction is then:

$$L = 1 - (\frac{C}{W/T}) \tag{4.1}$$

where *C* is the number of probes actually received and *W/T* is the number of probes that should be received within the window *W*. In order to calculate the ETX, the loss ratios in both the forward and reverse directions are required. This is due to the fact that a successful unicast data transfer using the IEEE 802.11 standards, involves a transmission of the data packet and receiving a link-layer acknowledgement from the receiver. The ETX value for a link is therefore a product of the loss ratio in the forward L_f and reverse direction L_r

$$ETX = \frac{1}{(1 - L_f)(1 - L_r)} \tag{4.2}$$

The ETX value given above, represents only the expected number of transmission over a link, i.e. for one single hop. The ETX value for a route consisting of two or more hops is the sum of ETX values for each link along the route.

The advantage of the ETX compared to the RTT and packet pair technique is lower overhead since probe packet is broadcasted instead of being unicasted to each neighbor. Besides, it also takes into account the potential difference in link quality in the forward and reverse direction. Furthermore, ETX suffers little from self-interference since loss ratio is measured instead of delay.

The ETX metric also have some disadvantages. Probe packets are small and since they are broadcasted, the transmission is performed at the lowest rate. They may therefore not experience the same loss rate as data packets sent at higher rates. Moreover, this metric does not directly account for link load or available bandwidth and is therefore not very precise in capturing the characteristic of the link. A heavily loaded link may have very low loss rate, and two links with different bandwidth may have the same loss rate.

In [68], the ETX metric was compared to the hop count metric and the end-to-end delay metric. It was demonstrated that ETX gave significantly better performance in static networks, but was outperformed by the hop count metric in dynamic networks because the probe based technique was too slow to adapt to the changes in the network.

4.5.4 Expected Transmission Time (ETT)

The ETT metric estimates the expected transmission time required to send a data packet over a link. This metric is based on the ETX, and is also called bandwidth-adjusted ETX. The ETT metric is defined as follows:

$$ETT = ETX \cdot \frac{S}{B}, \tag{4.3}$$

where *S* is the average size on a data packet, and *B* is the raw bandwidth of the link. The ETX value is calculated as described in Section 4.5.3, while the bandwidth *B* may either be set to a fixed value, or alternatively the packet pair method may be used to estimate it. The latter method was for example used in [70] and [72] in their implementations of ETT.

Based on the ETT metric, a routing protocol select the routing path in which the sum of ETT value on the path is the least. This implies that the ETT metric attempts to minimize the expected air time that is consumed in successfully delivering a frame from the source to the end destination.

The weakness of the ETT metric is that it does not explicitly account for contention (i.e. backoff time waiting for the channel to be idle) due to traffic from nodes in the vicinity. This is because the raw bandwidth is used in the definition of ETT in Equation 4.3. One way to incorporate the impact of contention is to use available bandwidth instead for the raw bandwidth. However, according to the authors in [70], current techniques to measure available bandwidth assume a point-to-point, First In First Out (FIFO) queuing model for the link, which is not the case for wireless links. On the other hand, using available bandwidth will make the metric to be more load dependent, which again will result in route instability in the network.

4.5.5 Radio load metric

The radio load is a measure for how busy the medium around a node is. Unlike the RTT, Packet Pair, ETX and ETT metrics, the radio load metric does not depend on probe packets to estimate how busy the medium is. Instead, the measurement is performed in a passive manner, where the IEEE 802.11 MAC-sublayer continuously monitors medium activity around the local node through the physical and virtual carrier sensing mechanism and the internal transmission state, as discussed in Chapter 3. The medium around a node is considered as busy if either i) the node is in the transmitting or receiving state or ii) if the node senses a busy carrier with

signal strength higher than the carrier sensing threshold. Otherwise the medium is considered idle. A similar technique is also applied in [73], but the difference is that the idle time is measured instead of the effective busy time.

Here we define the radio load as the amount of time T_{busy} within a time window T_{window} where the local channel is monitored busy. To estimate the average radio load *L* the exponential moving average is used as follows:

$$L_{new} = \alpha \cdot L_{previous} + (1 - \alpha) \cdot \frac{T_{busy}}{T_{window}} \tag{4.4}$$

where α is the weighting factor defined as $\alpha \in$ [0,1].

Each node in the network is responsible for measuring the perceived local radio load and makes this information available to the upper routing layer. The primary advantage of this metric is as stated earlier that it does not rely on active probing. This implies that there is no overhead to consume scarce bandwidth resources and is therefore more scalable with respect to larger topologies. Besides, the radio load metric does not suffer from the problems with queuing delay, self-interference, as in the case of the RTT, Packet-pair, ETX or ETT metrics, since there is no probing.

The drawback of this solution is however, the violation of the reference architecture of the Open Systems Interconnection (OSI) model [74], since the solution depends on cross-layering. In our case this implies modification of the interface between the link-layer and the network-layer to support exchange of lower layer (link-layer) information to the upper layer (network-layer). The authors in [75] [76], have emphasized the importance of architecture and discussed the architectural problems that cross-layering can create if done without care. They also warned about the possibility of inadvertent performance degradation due to interaction between conflicting cross-layer design proposals. If many violations of the architecture accumulate over time, the original architecture can completely lose its meaning and adversely impact the longevity of the architecture. However, we believe that the modification to support the exchange of radio load information is not harmful, since the fundamental behavior of the link-layer is unchanged. The only difference is that more information is presented to the upper layer.

Chapter 5

Load balancing

5.1 General Description

Load balancing is a key component in traffic engineering, and refers to the process of distributing traffic load more evenly in the network. It is an vital mechanism in order to achieve more optimal usage of network resources and improved performance.

In MANET, without an intelligent scheme for routing network traffic, the traffic load in the network can easily become unevenly distributed. This may potentially result in congestion at local hotspots, severe packet loss and degradation in the network performance. Uneven load distribution is usually caused by uneven user demands or uneven node distribution, where the latter may be a consequence of the unplanned and mobile nature of MANET. Furthermore, certain nodes in the network are more vulnerable to become congested than others due to their location or assigned role. Nodes located in the centre of the network tend to be more congested than nodes in the periphery, either because the majority of packets have to traverse these central nodes or they have to contend with a higher number of neighboring nodes for the medium. Nodes having the role as gateways between network domains may be more congested since all interdomain traffic has to traverse through them. Avoiding congestion at such key nodes is critical in maintaining network connectivity and the services they provide.

To prevent uneven load distribution and congestion in the network, parts of the traffic load in a congested area has to be diverted to other areas that are less congested. In this process the routing protocol plays an important role. Traditional routing protocols usually select the shortest path between any source and desti-

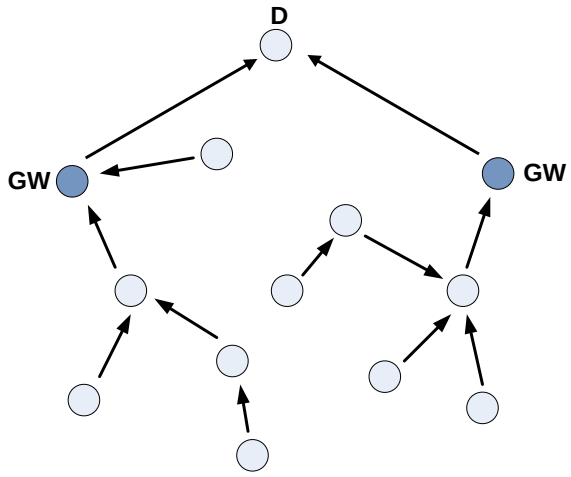


Figure 5.1: The concept of gateway load balancing

nation pairs. They do not consider the link quality of the path, in terms of the amount of available bandwidth, congestion, loss rate, and delay. Due to this shortcoming, they are without modifications incapable to perform load balancing, since load balancing in many cases involves selecting alternative longer paths in order to circumvent congested areas. However, the longer path length taken by the alternative paths increase the overall network resource usage, with respect to the number of transmissions and battery lifetime. Besides, end-to-end packet delivery delay may increase and throughput may decrease. Therefore trade-offs must be made between achieving load balancing and the potential increase in cost and decrease in throughput.

In this thesis, much of the focus has been devoted to the research in load balancing with the aim to improve *interdomain* and *intradomain* traffic. In the following sections, we will therefore provide an overview of the different load balancing types typical to MANET, and discuss the main issues and challenges related to each type.

5.2 Gateway Load Balancing

Gateway load balancing refers to the task of distributing interdomain traffic more evenly and intelligently between the gateways in order to achieve higher aggregated throughput, as illustrated in Figure 5.1. The prerequisite is that there are two or more gateways deployed in the network, providing connectivity to exter-

nal network domains such as the global Internet. Since all interdomain traffic has to traverse the gateway nodes, they are consequently more vulnerable to become congested. This why it is necessary to deploy multiple gateways in the network in order to increase the overall capacity and to alleviate the probability for congestion. In addition, it also provides redundancy and increased robustness. If one gateway encounters failure, there are still others that can serve. Furthermore, it can also lead to fairness improvement, i.e. with only one single gateway, different nodes enjoy different capacities depending on their proximity to the gateway. However, with multiple gateways, the average distance to the available gateways is the same for all nodes. This may therefore alleviate the unfairness problem [77].

Although the deployment of multiple gateways provides several advantages, however, it also introduces a number of issues and challenges that need to be addressed effectively in order to fully exploit these advantages. The authors in [77] provide a discussion on these challenges as discussed below, and emphasize the importance of performing load balancing between the gateways. Traditional shortest path routing protocols lacks this functionality and may cause a gateway to be overloaded, while others may be strongly underutilized, either due to uneven node distribution or user demands. Hence, without proper load balancing there is a potential risk for a degradation in the performance.

Characterization

Performing gateway load balancing is essentially to select a default gateway in which interdomain traffic can be forwarded to. In order to make correct decisions regarding which gateway to select as the default gateway, it is important to obtain statistics on characteristics related to the gateways and nonetheless the paths towards these gateways. This can either be one of the following characteristics or a combination of them: the hop count, loss probability, end-to-end delay, throughput or the traffic load. The metrics discussed in Section 4.5 are the common methods for obtaining these characteristics. What needs to be considered when using these metrics is the accuracy of the metric and the incurred overhead. Every kind of measurement is subject to some uncertainty, which can either come from the measurement instrument, the method being used, the environment, or other factors. In a distributed network, each node must individually carry out the measurement on its own and disseminate the measured information throughout the network. This is important, since all nodes in the network must have the same view about the condition of the network in order to distributedly and collaboratively perform load balancing in an efficient way. The dissemination of this information can either be done reactively of proactively. The common approach for reactive routing protocols is to gather the necessary information during the route discovery process. On the other hand, with proactive routing protocols, the information is usually dissem-

inated periodically and typically piggy-backed together with the routing messages. This is for example demonstrated in paper D and E. One of the challenges here is the unreliable nature of multihop wireless communication, which can potentially result in loss of vital information. Alternatively, it can also results in severe delay variations in the delivery of the information. These problems can be significant when the network load is high and when the dissemination is based on broadcasting, which is normally the case. Even with unicasting, it is not guaranteed that all nodes will receive the measured information. Thus the uncertainty in the dissemination can result in inconsistent view of the traffic load in the network. Consequently, this can lead to non optimal or erroneous decisions and adversely affect the efficiency of load balancing.

Gateway selection

Once the necessary statistics are obtained, the next issue to consider is which specific gateway to select as default gateway. For outbound traffic, it is up to the originating node to decide to which gateway the traffic is forwarded to. For inbound traffic, one possible solution is to let a *master gateway* as suggested in [77], to decide which *slave gateway* the traffic is forwarded to in order to achieve load balancing. The key parameters to consider when selecting default gateway are first of all the traffic load or available bandwidth at the gateways. Basically, the least loaded gateway should be favored, while the most loaded should be avoided. The second parameter to consider is the distance in terms of hop length from the local node to the gateways. In general, a shorter path should be favored since it can provide higher throughput, and has lower transmission cost and probability for end-to-end transmission failure. However, in cases where the nearest gateway is overloaded, it may be beneficial to route traffic to an alternative less loaded gateway, even though the distance is several hops longer away. The third parameter to consider is the characteristic of the paths towards the gateways. This may either be, the end-to-end delay, loss probability or bottleneck capacity. While the first parameter characterizes the condition at the gateways, the second and third parameters determine the characteristic of the paths towards the gateways. There are situations in which the first parameter favors one gateway while the second and third may favor another. Thus, the gateway selection algorithm must consider the trade-off between the parameters. The challenge is how to appropriately weight the parameters in order to achieve an overall increase in performance.

Route flapping

Normally, the gateway selection is performed distributedly where each node is responsible for selecting its own default gateway. Without coordination in the endeavor of distributing traffic more evenly, there is a potential risk for route flapping. In severe cases, a group of nodes may simultaneously and repeatedly reroute their

traffic back and forth between two neighboring gateways. This is referred as the *synchronized rerouting* problem and is discussed in more detail in paper E. The authors in [78] notice that it is very hard to maintain an absolute balance in the load distribution. Instead for a to aggressive approach in performing load balancing, care should be taken in order to avoid the ping pong effect. Besides, as the results in paper D show, it is not given that a perfectly even load distribution between the gateways will result in the most optimal performance. To alleviate the ping pong effect, [78] proposes a load balancing scheme that triggers gateway selection only when the measured load difference between two gateways exceeds a predefined threshold value. Alternatively, the authors in [79] propose a more centralized solution where a congested gateway may request one or more of its associated nodes to redirect their traffic to alternative less congested gateways.

Single vs Multiple Default Gateway
A node may select one or multiple gateways as its default gateways. The advantage of multiple default gateways is the possibility for distributing traffic from a node to several of its selected gateways. This allows for increased flexibility and more fine-grained granularity in performing load balancing. A node can for example direct a traffic flow to one gateway while another flow is directed to another gateway. For even finer granularity, a node can split up its traffic at the packet-level and route each packet individually towards the selected gateways. However, the challenge is how to allocate the correct amount of traffic from each individual node to each gateway. The authors in [80] argue that multiple default gateway architecture is very complex and difficult to realize in practice, and therefore the usual approach is to use a single default gateway architecture.

Packet reordering
Performing load balancing on multiple gateways can potentially cause packets to a single destination to traverse through different paths and gateways. This is especially true if the multiple default gateway architecture is used. The consequence is however that packets may arrive at the end destination in an out of order fashion. Hence, a packet reordering scheme is needed in order to ensure a FIFO delivery at the end destination. A possible solution to solve this is to use a *master-gateway* as mentioned earlier. The master gateway serves as an aggregation point or distribution center for interdomain traffic, and is responsible for collecting and reordering inbound/outbound packets such that they can be forwarded in the correct order.

The research in gateway load balancing is a central part in this thesis. The work in paper D and E is an effort to solve some of the issues and challenges discussed above in order to bring forth efficient schemes for performing gateway load balancing. In particular, the focus has been aimed at solving the main issues that are vital to the realization of such a scheme which include characterization, gateway selec-

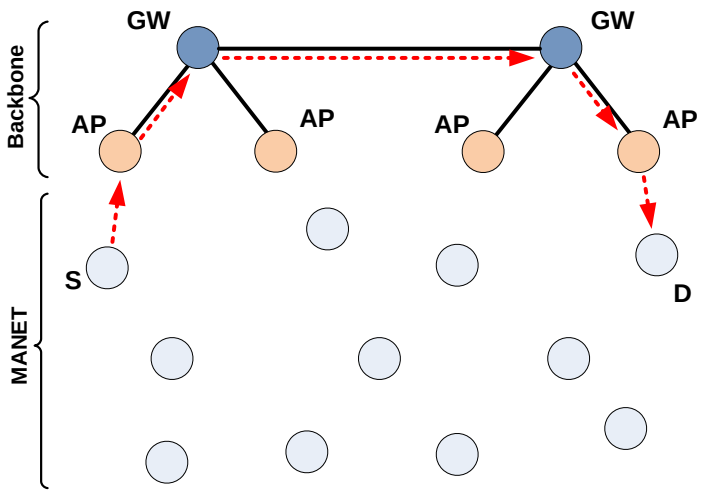


Figure 5.2: The concept of transit routing.

tion, and how to minimize the effect of route flapping. Due to time constraints, we have not paid much attention to issues such as packet reordering or multiple default gateway. Furthermore, besides of addressing the issues and challenges related to the design phase, much effort has also been invested in investigating the environmental factors that may have an impact on the performance of performing gateway load balancing. This is the focus of paper C, where it is shown that a number of factors such as gateway location, the level of offered load, the sensing range, and the specific layout of the topology are some of the important factors that may significantly affect the potential benefit of performing gateway load balancing.

5.3 Transit Routing

The previous section shows that the deployment of multiple gateways obviously has its advantages. In certain network architectures such as the proposals in [81] [82], multiple such key nodes are deployed in the network to form a backbone infrastructure. As illustrated in Figure 5.2, such a network is made up of two subnets, i.e. a backbone subnet and an MANET subnet that are connected together. The backbone subnet consists of gateway nodes and access points, interconnected by high capacity wired links (or wireless links). Normally this backbone infrastructure is exclusively used for interdomain traffic between nodes in the MANET subnet and destination nodes located in external domains. However, this backbone subnet can also be used as a transit network for data traffic between nodes in the

MANET as demonstrated in [83]. This has several advantages as discussed below:

- By routing part of the intradomain traffic over the high-capacity backbone subnet, the traffic load in the MANET subnet can be alleviated. Thus we may regard transit routing as a kind of load balancing for intradomain traffic.

- For certain source and destination pairs in the MANET subnet, e.g. in the case from S to D, transit routing makes it possible to achieve a considerably higher end-to-end throughput, since the wired backbone has much higher bandwidth than wireless links in the MANET.

- Transit routing also provides a higher probability for successful transmissions, since the wired links are much more reliable compared to wireless links.

- Wireless communication over multihop is often error prone and instable. By routing over the more reliable wired backbone, it is easier to maintain a more stable traffic stream between mobile nodes separated by many hops.

In order to support transit routing, there are a number of issues and challenges that need to be addressed. Many of these issues and challenges are common to the case of gateway load balancing, which include characterization, selection of default gateways/access points, traffic allocation, packet reordering etc. The main challenge is however to complement the routing protocol with the functionality for transit routing. Most MANET routing protocols such as OLSR or AODV are by default based on the shortest path metric, and as discussed previously, without modifications they are incapable of performing advanced tasks as transit routing, since it in many cases incur routing over longer path. To determine whether it is beneficial to perform transit routing or not for a certain source and destination pair, the routing protocol basically needs to consider the cost C_i for routing the traffic over the "ad hoc" path (i.e. internally within the MANET subnet), and the cost C_{ii} for the "wired path" (i.e. over the wired backbone subnet). The cost metric algorithm must be designed such that transit routing is favored only in situations when there is a potential benefit in terms of increased throughput and/or reduced traffic load in the MANET. The work in Paper B is an effort to address this issue, where the focus is to investigate the situations in which transit routing is beneficial or not, and thereby develop an appropriate cost metric algorithm in order to facilitate transit routing. A more detailed description of our work related to transit routing can be found in Section 6.3 and in Paper B.

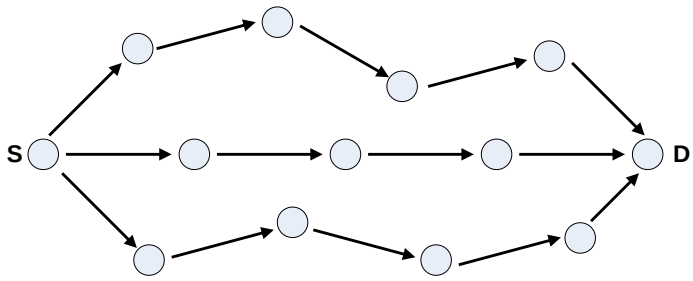


Figure 5.3: The concept of multipath load balancing

5.4 Multipath Load Balancing

Multipath routing is a routing approach that allows for load balancing of intradomain traffic, and therefore it is also called *multipath load balancing*. The concept is to distribute the traffic between a source and destination pair over multiple alternative disjoint/semi-disjoint paths as illustrated in Figure 5.3. Alternatively, multipath routing can also be used for other objectives such as to increase the reliability and confidentiality of data transmission, optimize energy consumption in the network, and to improve QoS in the network. A considerable number of previous work have investigated and proposed different schemes for multipath routing. The study in [84] provides an overview of the diversity of multipath routing protocols for MANETs. The majority of these proposals are based on reactive routing protocols such as AODV and Dynamic Source Routing (DSR), since the reactive approach provides an easier way to discover all possible disjoint paths between a source and destination pair.

Although the concept of multipath routing may contribute to improved performance, there are a number of issues and challenges that need to be considered as discussed in [85] and [84]. One of the issues is related to route discovery and route maintenance. These tasks are generally more costly to perform in terms of overhead and storage in multipath than in single path routing protocols. Furthermore, in sparse networks it can be difficult to find multiple disjoint paths (either node or link disjoint). Even in densed networks the number of node disjoint paths may be limited. Disjoint paths offer certain advantages over non-disjoint paths. For instance, non-disjoint paths are less resilient against link breaks and may have lower aggregate throughput. Another issue is related to route selection. If there are multiple paths between the source and the destination, the question is how many of these paths should be used for data transmission. One may either select all the paths or a subset of them, and the paths may be selected based on certain criteria

such as link quality, delay, and bandwidth in order to achieve optimization. Once the source node has selected the paths, it must also decide how to allocate traffic to these paths. Traffic allocation may be performed at different granularity. For instance, a per-connection granularity can be used to allocate all traffic from one connection to a single path. Alternatively, a more fine-grained per-packet granularity may be used to distribute packets from a connection amongst the selected paths. It is reported that per-packet granularity results in the best performance since it allows for fine tuning of the traffic distribution in the network [86]. However, the disadvantage of the per-packet granularity is the increased risk for packet reordering at the destination. The last issue is when to trigger route discovery. It can either be triggered each time one of the paths is broken or only when all paths are broken. The first may incur a considerable amount of control traffic overhead while the latter may result in performance degradation. A possible good compromise may be to initiate route discovery when a certain percentage of the paths are broken.

With respect to performing load balancing, previous work have proposed different multipath load balancing schemes in order to maximize throughput, and at the same time minimize packet delay and route failure. Multipath load balancing has proven to be efficient in wired networks [86] [87] [88] [89], however, the same effect is difficult to achieve in wireless networks. In contradiction to wired networks, the challenge with multipath load balancing in wireless networks is the interfering nature of the wireless medium. The interference that occurs among the paths limits the achievable gain in performance. The work in [1] refers to this interference as route coupling, and showed that this problem is especially severe in single channel networks. They showed that the performance gains provided by multipath load balancing is only negligible compared to single path routing. Route coupling is less severe in multi-channel networks, due to locally unique channel assignments. However, they showed through simulation results that route coupling still exists even in multi-channel networks. Furthermore, by routing over more spatially separated paths, the effect of route coupling is reduced since the level of spatial reuse is consequently higher. However, this often requires that the traffic is routed over longer paths in terms of hop count, causing more network resources to be consumed. The results in [2] also confirmed the conclusion above. They showed through analysis and simulations that the gain provided by multipath load balancing is negligible unless the traffic is distributed over a huge number of paths. Based on these reports, multipath load balancing has therefore not been considered in this thesis.

Chapter 6

Summary and Contributions

6.1 Summary of the Work

The work in this thesis addresses two selected issues in the context of ad hoc network for emergency and rescue operations, i.e. *mobility* and *load balancing*. The aim is to investigate the shortcomings of current solutions with respect to these issues, and bring forth new solutions to improve the performance of the network as a whole or for the individual nodes. Thus, our work can be regarded as an effort on the way to realize an emergency and rescue communication system based on the ad hoc network technology. While the work in paper A addresses the issue of mobility, paper B to E address the issue of performing load balancing.

With respect to the issue of mobility, the focus is to investigate the factors that affect the rerouting time in proactive routing protocols. That is, the time duration needed to reroute and restore a broken communication path due to node mobility. The aim is to provide solutions for minimizing the rerouting time such that the packet loss can be minimized and the performance in the network is maximized. As previously discussed in Section 4.4, the rerouting time can be reduced by reducing the link break detection time, for example by applying alternative methods for link break detection such as the Fast-OLSR scheme or the link layer feedback mechanism. However, the drawback is that these solutions either incur increased control traffic overhead or the potential for erroneously declaring a link as invalid. The work in paper A shows that besides the impact of the link break detection mechanism, other factors such as the queue length, the input packet rate and the retransmission limit at the MAC-sublayer can affect the rerouting time significantly. To solve the rerouting time with respect to these factors, either the queue length

or the number of retransmissions can be reduced. The problem with reducing the queue length is the potential risk for packet loss due to buffer overflow at higher packet rates. Therefore, paper A proposes a solution based on *Adaptive retry limit*, where the retransmission mechanism at the MAC-sublayer is gradually decreased in the event of a link break. The reason for this is to minimize the number of transmissions wasted on stale packets, i.e. packets with invalid next hop address. Simulation results show that the proposed solution is very effective. In fact, as long as the data rate into the queue is safely below the capacity of the MAC, the solution eliminates the queueing problem associated with the rerouting time.

With respect to the issue of load balancing, the focus is to investigate and explore the feasibility as well as the potential benefits of performing load balancing in MANETs. The aim is to bring forth solutions that can optimize the usage of network resources and to improve the network performance in terms of increased throughput.

The work in paper B addresses load balancing for intradomain traffic. The assumption behind this work is that there exists a wired high capacity backbone subnet part of a MANET. Traditionally, such a backbone subnet is exclusively used to provide Internet connectivity to wireless nodes and to extend coverage area. However, the work in paper B demonstrates that it is also possible to exploit the capacity of this backbone subnet to alleviate the load in the MANET. This is achieved by routing part of the intradomain traffic, i.e. traffic between nodes in a MANET, over the backbone subnet. We refer to this kind of load balancing as transit routing. Transit routing does not only allow for load balancing, but for certain source and destination pairs, the throughput can also be considerably increased. Paper B thus proposes a cost metric algorithm in order to facilitate transit routing. This algorithm is designed to commence transit routing only when appropriate. This means that when there is a performance gain in terms of throughput by using the alternative path through the backbone subnet, the cost metric algorithm will favor this path. Simulation results show that by using the concept of transit routing, it is possible to enhance the throughput by 50% on the simulated topologies.

The work in paper C to E addresses load balancing for interdomain traffic. The assumption is that there exist multiple gateways in the network that provide connectivity to external network domains, such as the global Internet. The advantage of having multiple gateways in the network is among others the increased capacity for interdomain traffic. To exploit and optimize the usage of the increased capacity, gateway load balancing needs to be performed in order to distribute interdomain traffic between the gateways more evenly. This is a rather challenging task especially when talking about wireless networks. Thus the work in paper C investigates the feasibility of performing gateway load balancing and explores the factors that

affect the potential gain of it. It is shown that a number of factors can affect the efficiency of load balancing as listed below:

- Level of asymmetry
- Offered load
- Level of spatial reuse
- Sensing range
- Shape and size of the network
- Location of gateways

However, these factors alone cannot explain why the performance of load balancing is high for certain topologies while it is poor for others. Obviously, the specific layout of the topology is also an important factor.

The work in Paper D therefore focuses on investigating the importance of the layout of a topology. Using the *congestion maps*, a number of interesting characteristics are discovered, which contributes to explain why the layout of the topology has significant impact on the performance of load balancing. Based on these results, two different load balancing schemes are proposed in Paper D and E. These schemes demonstrate that radio load information can be used to perform load balancing in MANETs. The load balancing scheme in Paper D is based on a deterministic gateway selection algorithm, and is proven to perform well for topologies with higher asymmetry level (with respect to node distribution and traffic load). However, for moderate asymmetry level, simulation results show that the solution is inefficient. This is due to what we refer to as the *synchronized rerouting problem*, and is in fact a consequence of the distributed nature of MANETs. Thus the focus in paper E is to solve this problem by applying a randomized gateway selection approach instead of a deterministic approach as is previously done in paper D. In addition to performing load balancing, the proposed scheme in paper E also performs admission control in order to prevent the network load in reaching a critical high level. Simulation results show that the new scheme is indeed more efficient than the scheme in paper D.

The work in all the above papers is tightly related to the preceding chapters. The IEEE 802.11 MAC-sublayer and the OLSR routing protocol presented in Chapter 3 and Chapter 4 represent the two fundamental technologies that are used throughout the work in this thesis. IEEE 802.11 is the dominating technology in the research related to ad hoc networks. OLSR is one of the most popular proactive routing protocols for MANET, and it is a natural choice in our research due to several reasons. First, an implementation of OLSR exists for the ns-2 simulator. Second, the

advantage of using a proactive routing protocol is that it can quickly and dynamically adjust or rebalance traffic load to the changing conditions in the network. Third, routing metric parameters can be easily integrated with the routing protocol in order to achieve QoS routing.

Finally, the proposed solutions in our papers involve modifications or the integration of new mechanisms into these protocols. The *Adaptive retry limit* solution in paper A is an example of such a modification to the MAC-sublayer in order to reduce the rerouting time. In addition, paper D and E demonstrate how the MAC-sublayer can be modified to provide radio load information to the routing layer. Similarly, the work in paper B, D and E, show that with new functionality, it is possible to increase network performance through transit routing and gateway load balancing.

6.2 Contribution of paper A: Rerouting Time and Queuing in Proactive Ad Hoc Networks

One of the features that characterize a MANET is that nodes are allowed to be mobile. Due to mobility, established links may be broken, and new links may be formed with new neighbors. This process of link breaks and formation of new links happens frequently if the level of mobility is high. While transmitting data, a sudden link break will interrupt the forwarding of packets to the intended receiver. The routing protocol is designed to find alternative routes in these situations. However, this rerouting takes times, and the latency of the rerouting is referred as the rerouting time. Paper A investigates the factors that affect the rerouting time in proactive routing protocols.

6.2.1 Related Work

The work in [90] has previously investigated and compared various neighbor sensing approaches for OLSR: the original Hello based link sensing of OLSR, Fast-OLSR and OLSR with link layer feedback (OLSR-LL). They notice that during a link break caused by mobility, the accumulation of stale packets (i.e. packets with invalid next hop addresses) in the interface queue is one of the reasons leading to the deterioration of the performance. However, they do not provide any further analysis on why this accumulation occurs and the factors that are behind this incident.

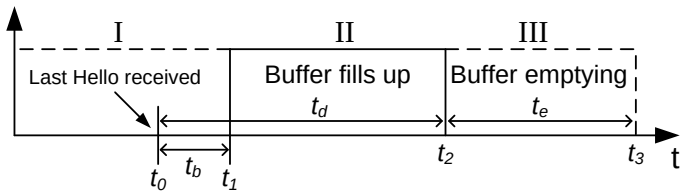


Figure 6.1: The queue build up during a link break

The work in [91] proposes the ingress queueing mechanism that shows to be an effective solution to the rerouting problem. The idea of this solution is to delay the process of looking up for the next hop address for an outgoing packet until the very last moment, i.e. the look up is performed just in advance to when the actual transmission takes place. The main advantage of this solution is that if a preceding packet is discarded due to a link break, then no further packets with the same next hop address are inserted into the queue.

6.2.2 Contributions

The main contributions of paper A is showing that the rerouting time in proactive routing protocols is affected by a number of factors such as the size of interface queue, the traffic load and the underlying retransmission mechanism at the MAC-sublayer. These factors can cause the rerouting time to significantly exceed the time needed to detect a link break. To solve this problem, we propose a solution that resides in the MAC-sublayer, and is thus independent of the routing protocol.

In proactive routing protocols such as OLSR and OSPF MDR, link sensing and neighborhood detection are performed by the exchange of Hello messages. When a node ceases to hear Hello messages from an established neighbor for a given period of time, then the link to this neighbor is assumed as broken. Given a Hello interval of 2 seconds (the default value in OLSR), a link break should normally be detected after 4-6 seconds. However, it is observed through simulations as well as with real life experiments that the rerouting time in many cases can be much higher, i.e. 20-40 seconds.

To explain why this happens, consider the illustration in Figure 6.1. During a data transmission, a link break occurs at t_1, however, this is not detected by the routing protocol until at t_2. Meanwhile, stale packets (i.e. packets destined to the downstream node upon which the link is broken) are still being inserted into the interface queue, and resulting to an accumulation or build up of stale packets in

the queue. This accumulation between t_1 and t_2 depends on a number of factors including the packet rate R_{in} and R_{out}, i.e. the packet rate into and out of the queue. The latter is mainly influenced by the preconfigured retry counter threshold at the MAC-sublayer, the packet size, the backoff delay, and the contention level around the sender node. Thus, the accumulation occurs when $R_{in} > R_{out}$, and the higher R_{in} is relative to R_{out}, then the faster will this accumulation occur. In the worst case the queue can be completely filled up with stale packets, causing other packets with valid routing information to be discarded due to buffer overflow. Furthermore, at t_2, since the routing protocol is now aware of the broken link, stale packets are not anymore inserted into the queue. In the interval between t_2 and t_3, the accumulated stale packets in the queue are now gradually emptied from the queue. However, this process may take a considerable amount of time, since each packet must be transmitted and retransmitted a number of times before it is discarded. While in this stage, the accumulated stale packets may block for any succeeding packets with valid routing information from being handled. This last until all the stale packets are eventually discarded at t_3.

From the above analysis, we define the rerouting time as the time starting from the last Hello received from the downstream node, to the time when the first packet rerouted via an alternative node is received by the downstream node. This corresponds to the interval from t_0 to t_3. Using the illustration in Figure 6.1 we can derive a model for the expected rerouting time as given in the equation below:

$$t_{rerouting} = t_d + \frac{1}{R_{out}} \cdot min[(t_d - t_b)(R_{in} - R_{out}), B] \tag{6.1}$$

In equation Equation 6.1, t_d is the link break detection time which by default is 6 seconds. t_b is the time interval between the last Hello from the downstream node that is received by the upstream node, and the actual time of the link break. *B* is the queue size in number of packets. Thus the rerouting time is equal to t_d, and the time needed to empty stale packets from the queue, represented by the second term. In the worst case when the queue is completely filled up with stale packets then the rerouting time equals to:

$$t_{rerouting\_max} = t_d + \frac{B}{R_{out}} \tag{6.2}$$

Otherwise, in the case when the rerouting time is larger than t_d but smaller than

$t_{rerouting\_max}$, we may write the rerouting time as:

$$t_{rerouting} = t_b + \frac{R_{in}}{R_{out}} \cdot (t_d - t_b) \tag{6.3}$$

The analysis above shows that especially two factors may significantly affect the rerouting time. The first one is the queue length, where a large queue may potentially result in a large amount of accumulated stale packets, and consequently a higher rerouting time. On the other hand, reducing the queue length to a smaller size may cause packets to be discarded during burst packet rates due to buffer overflow. Hence, adjusting the queue size to solve the rerouting time problem is probably not a good solution. The second factor is the retransmission mechanism at the MAC-sublayer. Even though this mechanism is very useful during temporary transmission errors caused for example by interference, however, with respect to persistent link breaks caused by mobility, this mechanism will decrease the output rate R_{out} and increase the rerouting time significantly. The higher the number of retransmission is, the more severe is the problem. Furthermore, not only will the retransmission mechanism result in increased rerouting time, but it will also incur a severe waste of scarce resources in transmitting stale packets. Thus to solve the rerouting time problem, we propose a solution called *Adaptive retry limit*. This solution is based on the modifications of the retransmission mechanism at the MAC-sublayer, and basically decrements the maximum retry value by one for each discarded packet destined to the next hop node upon which the link is broken. Through the reduction of the number of retransmissions, the rate R_{out} is prevented from decreasing when a link break occurs. Simulations and analysis show that the proposed solution eliminates the entire problem of increased rerouting time. Finally, comparisons of the results from the estimated and simulated rerouting time show that the model given in the above equations is a good approximation.

6.3 Contribution of paper B: Routing of Internal MANET Traffic over External Networks

The work in paper B is concerned about performing load balancing for intradomain traffic in a MANET. Assuming that there exists a wired high-capacity backbone subnet in the network, the traffic load in the MANET can be reduced by routing parts of the intradomain traffic over this backbone subnet. Thus, the paper demonstrates that such a backbone subnet can be used as a transit network for intradomain traffic in the MANET.

6.3.1 Related Work

In the literature, a number of proposals have suggested various network architectures for providing Internet connectivity in a MANET. For example, the work in [82] and [81] propose network architectures where a MANETs subnet can gain access to the global Internet through a wired backbone access subnet. The architecture is similar to the reference topology in Figure 6.2, and is basically composed of 3 types of network entities: gateway nodes (GW), access points (AP) and mobile nodes (MN). However, the work in [82] and [81] are mainly concerned about the design of the architecture in order to provide Internet connectivity to the MANET, and the handling of micro and macro mobility. While micro mobility is handled by the routing protocol, macro mobility is handled by Mobile IP (MIP) [92]. On the other hand, we have through the work in this paper chosen to focus on exploiting the resources of the high capacity backbone subnet to enhance the performance of intradomain traffic in the MANET subnet. The aim is to route part of the traffic between mobile nodes in the MANET over the backbone network to achieve higher throughput and to reduce the load in the MANET. This is referred to as transit routing. To the best of our knowledge, only the work in [83] has considered transit routing in ad hoc networks. Their proposal is based on a reactive approach, using the Dynamic Manet On-Demand Protocol (DYMO) routing protocol [50], while our proposal is instead based on a proactive approach, using the OLSR routing protocol. More importantly, in their proposal, they use only a simple metric for transit routing, i.e. transit routing is only allowed when it results in a reduction in the number of wireless transmissions. However, in our study, it is demonstrated that in many cases, even when transit routing results in more wireless transmissions compared to direct routing within the MANET, there is still a potential for increasing the throughput considerably.

6.3.2 Contributions

The main contribution of paper B is to demonstrate that transit routing over a wired backbone subnet in many cases can be advantageous. As previously discussed in Section 5.3, transit routing has the potential benefit of alleviating the load in the MANET subnet. In addition, it can also increase the throughput and reliablility for traffic between two nodes in the MANET. To illustrate this, consider the reference scenario in Figure 6.2. Node n_0 has traffic to send to node n_5. n_0 can either send to n_5 using the "ad hoc path" (i.e. from n_0 $\rightarrow$ n_1 $\rightarrow$ ... $\rightarrow$ n_5), which implies that the traffic is sent within the ad hoc subnet and using wireless links only. Alternatively, n_0 can send traffic to n_5 using the "wired path" (i.e. n_0 $\rightarrow$ A0 $\rightarrow$

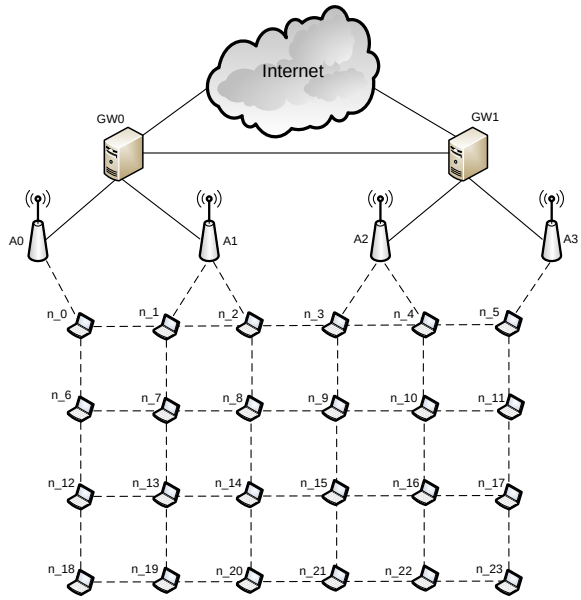


Figure 6.2: Example of a MANET interconnected with the global Internet through APs and GWs

... $\rightarrow$ A3 $\rightarrow$ n_5) which goes over the backbone subnet. A packet sent over the ad hoc path requires 5 successive wireless transmissions. On the other hand, sendig a packet over the wired path, requires "1+1" wireless transmissions (i.e. 2 non successive and distinct wireless transmissions), and 3 wired transmissions. Assuming that the bandwidth of wired links is much higher, it is obvious that using the wired path is much more advantageous than the ad hoc path. Besides, wired links do not suffer from self-interference problems as is the case with wireless links, and this is also the reason why the max throughput over wireless links decreases with increasing number of hops. In fact, simulation shows that using the wired path for the example above results in a throughput that is more than four times higher than the throughput of the ad hoc path. Thus the example shows that transit routing not only can reduce the load in the ad hoc subnet, but also improve the throughput significantly.

Based on extensive simulations and analyses, we have proposed a cost metric algorithm to facilitate transit routing. This cost metric algorithm is designed to commence transit routing only when it is beneficial for the performance, either for the MANET or for the individual nodes. For the further discussion, let us consider Figure 6.3, which is a simplification and generalization of the scenario in Figure 6.2. Assume that A is the source which has some data packets to send to B.

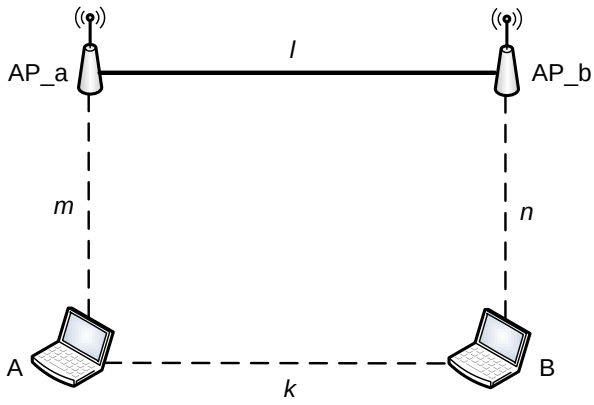


Figure 6.3: Transit routing scenario

The distance in number of hops from A to the nearest access point of A, which is AP_a, is *m* hops. Similarly, the distance from B to its nearest access point AP_b is *n* hops. The distance between A and B is *k* hops through the "ad hoc path". The distance from AP_a to AP_b through the backbone subnet is *l* hops. Node A can thus send its data packets either through the ad hoc path using *k* wireless hops, or alternatively it can send the traffic via the "wired path", i.e. through the backbone subnet and then to node B, using *m*+*n* wireless hops and *l* wired hops.

In order to determine whether it is beneficial to utilize the ad hoc or wired path, a cost is calculated for each path. In the following, a brief description of the cost metric algorithm is given. A more detailed description can be found in paper B. The cost C_i for the wireless path is given in Equation 6.4, which is equal to the number of wireless hops *k*. The cost C_{ii} for the wired path comes in two flavors, depending on whether the section *m* and *n* are interfering with each other or not. For the case without interference, the cost is given in Equation 6.5. On the other hand, for the case with interference, the cost is given Equation 6.6. *c* is a constant which accounts for the bandwidth ratio between the wired and the wireless link. The essential point with respect to the cost C_{ii} is that without interference, the max throughput of the wired path is constrained by the wireless section that contains most hops, i.e. max(*m,n*). On the other hand, with interference, the max throughput of wired path is lower and is determined by the sum of *m* and *n*. Transit routing is thus most optimal when section *m* and *n* are not interfering with each other such that the highest throughput can be achieved.

$$C_i = k \tag{6.4}$$

$$C_{ii(no\ interference)} = max(m, n) + c \tag{6.5}$$

$$C_{ii(interference)} = sum(m, n) + c \tag{6.6}$$

Having calculated the cost for both paths, the algorithm below decides which path that should be selected, i.e. the path with lowest cost (lines 4-7). The parameter *g* in line 1 is a preconfigured parameter that determines the "greediness" of the algorithm. For example, if we want to limit transit routing only to cases where the wired path, at most, incur one additional hop compared to the ad hoc path, i.e. when "*m+n*" $\leq$ *k+1*, then *g* is set to 1. The purpose of this parameter is to control the accepted amount of extra load on the ad hoc subnet in exchange of a higher throughput using the wired path.

```
1   if ((m+n)-k > g)        #g= 0,1,2
2       ad_hoc_path
3   else
4       if (C_ii < C_i)
5           wired_path
6       else
7           ad_hoc_path
```

The cost metric algorithm above is evaluated by simulations on three different random topologies. By using the transit routing, the results show that the average enhancement in throughput for all three simulations is approximately 50 %.

6.4 Contribution of paper C: Performance Analysis of Gateway Load Balancing in Ad Hoc Networks with Random Topologies

In MANETs or WMNs, a gateway is a node that provides connectivity to the outside world such as the global Internet. Since all interdomain traffic has to traverse the gateway, it is vulnerable to congestion and become a bottleneck. To alleviate this problem, the common solution is to deploy multiple gateways in the network. However, in order to take advantage of the increased capacity provided by multiple gateways, the routing protocol utilized must efficiently balance the traffic load among available gateways such that the network performance is optimized. This is referred to as gateway load balancing, and is previously discussed in Section 5.2.

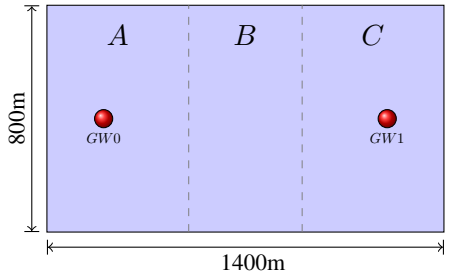


Figure 6.4: The reference network model.

6.4.1 Related Work

In the literature, a number of work have previously addressed the issue of gateway load balancing, and a corresponding number of load balancing schemes have been proposed [79] [93] [78] [94]. However, to the best of our knowledge, none have made a study under the condition that the topology is random. Very often, the topologies used in the evaluation are either very simple, consisting of a few nodes, or they are constructed in "unnatural" grid formations of various sizes. Hence, these topologies cannot be regarded as representative topologies from the real world. They are rather simplified examples to illustrate the feasibility of the concepts in a given setting. Consequently, it is questionable if these results are qualified to validate the actual performance of their proposals in a real and random setting.

6.4.2 Contributions

Due to the shortcomings discussed above, it is reasonable to question the feasibility of performing load balancing in a wireless and interfering environment. Furthermore, if feasible, to which extent does load balancing improve the network performance, and what are the factors that may set an upper limit for the performance that can be achieved? Thus, the contributions in paper C is concerned about finding the answers to these questions based on simulations with a large number of random topologies.

In this study, the default scenario used in most of the simulations is as shown in Figure 6.4. The network spans over an area of 1400 m by 800 m, and consists of two gateways (GW) and 50 randomly deployed nodes (not shown in the figure). The GWs are symmetrically placed at fixed locations inside the network area, and by default, the distance between the two GWs is 1000 m as shown in Figure 6.4. The aim is to study to which extent it is possible to improve the performance for

outbound traffic if we can perform load balancing between these two GWs. Furthermore, in order to investigate how each specific factor affects the performance of load balancing, without potential disturbing effects, all simulations in this study are therefore performed on static topologies. Even though there is no node mobility in the simulated topologies, it is anticipated that our results also give insight into the performance of scenarios with random node mobility, assuming that each simulated topology might represent a snap-shot of a topology with mobile nodes.

One important characteristic of a topology is the level of asymmetry with respect to node and load distribution. As shown later, this parameter is one of the main factors that affect the performance of load balancing. Assuming that each node has the same amount of outbound traffic, we may define the *asymmetry index AI* which is a measure of the degree of imbalance in load distribution between the two gateways, as follows:

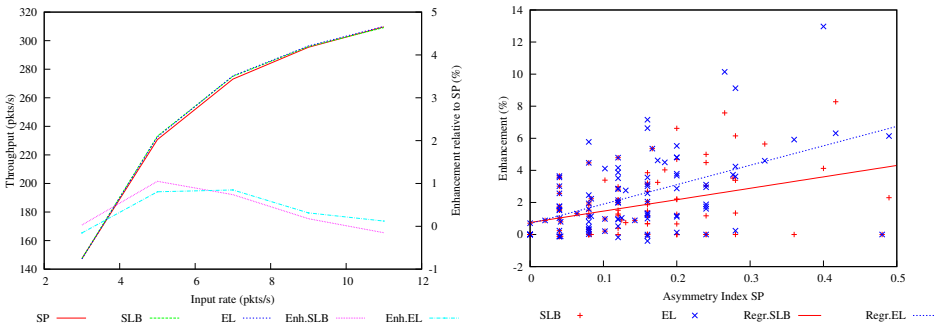
$$AI = \frac{abs(n_0 - n_1)}{n_0 + n_1} \tag{6.7}$$

where n_0 and n_1 are the number of traffic flows sent to *GW0* and *GW1* respectively. When the load distribution between the two gateways is perfectly balanced, then *AI*=0. In the worst case when all traffic is sent to one gateway, then *AI*=1. Furthermore, given that we use the Shortest Path (SP) metric, then *AI* may also represents the asymmetry in the topology, i.e. the asymmetry in node distribution relative to the gateways.

In our study, three different routing metrics are used in the simulations, as described below:

- The SP metric, also known as shortest hop count metric, is the default metric that is used in most MANET routing protocols. This metric basically select the nearest gateway as default gateway. If a node has the same hop count to two or more gateways that are the nearest, then a random gateway is selected as the default gateway.

- With the Simple Load Balancing (SLB) metric, nodes basically select the nearest gateway as default gateway. However, when a node has the same hop count to two gateways that are the nearest, then the least loaded gateway is selected as the default gateway. This metric is conservative in the sense that it does not allow a node to send traffic to alternative less congested gateways that are farther away. This is because a longer path consumes more resources in terms of bandwidth and battery power.



(a) Throughput vs. offered load

(b) Enhancement vs. AI

Figure 6.5: Simulation results of uniformly distributed topologies.

- With the Even Load (EL) metric, the network load is attempted to be distributed as evenly as possible between the gateways. In contrast to the SLB metric, a node can choose to forward its traffic to a more distant and less congested gateway in order to achieve load balancing. Compared to the SLB metric, the EL metric is much more aggressive in the way load balancing is performed, i.e. outbound traffic is more evenly distributed between the gateways, but at the cost of higher bandwidth and battery power consumption due to longer paths.

With these metrics, we can explore how the two load balancing approaches SLB and EL perform relative to the default SP metric. The result may give us an idea about the potential benefit of performing load balancing with respect to the aggregated throughput for outbound traffic. Even though the focus in this study is limited to outbound traffic only, we believe that the results obtained in this work is also applicable to the case with inbound traffic as previously discussed in Section 5.2.

Initially, simulations are performed on 100 topologies, where the nodes are randomly and uniformly deployed within the network area. We refer to these topologies as *uniformly distributed topologies*. Results from the simulations show that, surprisingly, the SP, SLB and EL metrics on the average, have almost the same performance as shown in Figure 6.5(a). This occurs even though the average AI=0.142, which is equivalent to 7 flows, or approximately 32 % in load difference between the two gateways. We believe that the coupling effect, i.e. interference and lack of spatial reuse, may be one of the reasons why SLB and EL do not improve the throughput compared to SP.

Figure 6.5(b) shows the peak throughput enhancement of SLB and EL relative to

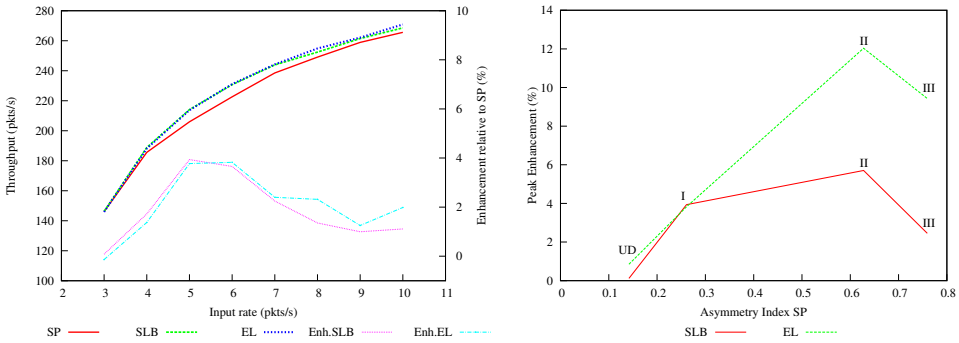
Table 6.1: Node Distribution Configurations

Topology Set	n_A	n_B	n_C
I	20	20	10
II	30	15	5
III	35	10	5

SP for each topology, where each mark in the figure refers to the simulation result of one specific topology using one specific metric, SLB or EL. The throughput enhancement is plotted as a function of the *AI*. From the results we see that the potential benefit of load balancing in a random setting is relatively limited. Of the 100 topologies, only two topologies result in a peak enhancement exceeding 10 %. On the other hand, if we consider the average peak enhancement for all topologies, the result is only around 1 % for both SLB and EL. However, the scattering of the results in Figure 6.5(b) indicates that the enhancement in throughput performance is strongly dependent on the layout of the specific topology. In addition, the linear regression lines for SLB and EL show that the enhancement in throughput increases with an increasing level of asymmetry. Hence, due to these observations, we relax our requirement of randomness, and generate new sets of topologies that are more asymmetric. We refer to these topologies as *asymmetric random topologies*. The aim is to determine under which conditions where it is optimal to perform load balancing.

The asymmetric random topologies are created by deploying the nodes asymmetrically such that, on the average, significantly more nodes are associated with *GW0* than with *GW1*. This is achieved by dividing the simulation area into 3 sections denoted as A, B, and C as shown in Figure 6.4, and then we randomly deploy for example 20, 20 and 10 nodes in section A, B and C respectively. Table 6.1 shows the node distribution for the three topology sets (TS I, II, and III) that are generated. Each topology set consists of 30 topologies.

Using these new sets of topologies, it is possible to statistically explore parameters that affect the potential for load balancing. Figure 6.6(a) shows the average throughput and performance enhancement for TS I. The enhancement in performance is significant only within a limited window along the offered load axis. At the lower limit of this window when the offered load is low, none of the gateways are congested and load balancing is unnecessary since there is no excessive load that needs to be migrated. This explains why the load balancing metrics have approximately the same performance as SP at low loads. In fact, utilizing an aggressive load balancing metric like EL will only result in poorer performance due



(a) Throughput vs. offered load

(b) Enhancement vs. asymmetry

Figure 6.6: Simulation results of asymmetric random topologies.

to the increased average path length, and consequently increased packet loss. On the other hand, when the offered load is high (i.e. in the upper limit of the window or above), only nodes closest to the gateways will be able to successfully send traffic while nodes farther away will suffer from very high packet loss. In this situation, load balancing will no longer be able to enhance the throughput, since it is only the nodes closest to the gateway that contributes to the throughput anyway.

Furthermore, simulations are also performed on TS II and III, in order to explore how the various levels of asymmetry affect on performance of load balancing. Figure 6.6(b) compares the average peak enhancement for both the uniformly distributed and the 3 asymmetric topology sets. The results indicate that with increasing asymmetry, the benefit of load balancing is indeed greater, and at the best, the performance is approximately 12% for the case of TS II. However, the results also indicate that above a certain level of asymmetry, the advantages of load balancing will decrease as in the case of TS III. This is due to the fact that with a very high level of asymmetry, the chance for partitioning is also considerably higher. Furthermore, we also see that at higher levels of asymmetry, there is a stronger need for a more aggressive load balancing approach, such as the EL metric. Thus, for the topology set II and III, EL appears to yield a higher enhancement compared to SLB. On the other hand, at lower asymmetry level, a conservative approach such as the SLB metric may be more appropriate. This is because both SLB and EL have approximately the same performance, but the cost in terms of network resources is less with SLB compared to EL.

Furthermore, although not shown in this summary, the work in paper C also shows that other factors including gateway distance, the level of spatial reuse (or frequency reuse), the shape and size of the network area are also factors that can

have an impact on the performance of load balancing. However, all the factors discussed and mentioned above cannot alone explain why for some topologies, the performance of load balancing is very good, i.e. up to 45% enhancement, while for others it is very poor. Indeed, the specific layout of the topology is another important factor that has a major impact on the performance.

6.5 Contribution of paper D: Gateway Load Balancing in Future Tactical Networks

Paper D is a continuation of the study in paper C. From the results obtained in paper C, we were left with the question of why the performance of gateway load balancing is considerably high for certain topologies, while it is very poor for others. The results in paper C indicate that the specific layout of a topology is likely to play a crucial role on the efficiency of load balancing. The question is what are the differences that make a topology more suitable for load balancing than others? Thus the motivation behind this work is related to the answer of this question, and the aim is to gain more knowledge in order to bring forth more efficient solutions.

6.5.1 Related Work

Although a number of previous work have proposed different schemes for gateway load balancing [78] [79] [94] [95] [80], very few provide an analysis of the factors that may affect the performance of load balancing. The results in paper C show that load balancing is affected by a number of factors, including the offered load, gateway locations, sensing range, and the level of asymmetry. However, these factors alone cannot explain why for some topologies, the performance of load balancing is very good, while for others it is very poor. Unfortunately, the explanation to this question cannot be found in any previous works. To our best knowledge, only the work in [77] provides some discussions on how the topology layout may affect different aspects such as capacity, fairness and resiliency. However, their study is not based on random topologies and neither do they provide an answer to the question above.

Furthermore, many proposed load schemes are based on a variety of metrics such as, queue length, the number of active flows, RTT and ETT. As far as we know, none have tried to use the radio load metric for the purpose of performing load balancing. Thus, the proposed scheme in this paper demonstrates that the radio load

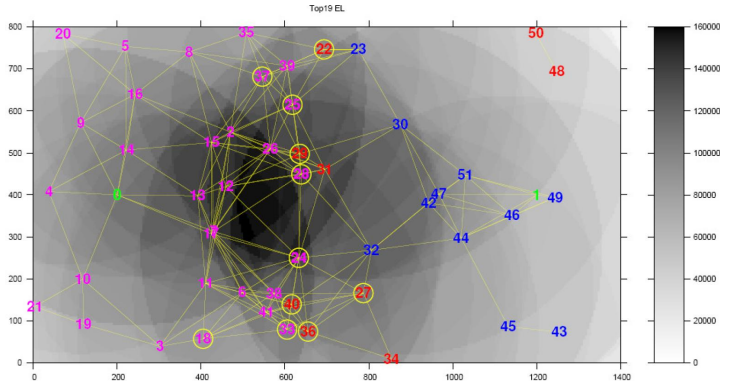


Figure 6.7: An example of congestion map. The dark areas represent the most congested areas in the network

metric can also be used for the purpose of performing load balancing. Compared with other metrics such as the number of active flows, the radio load is a more accurate metric, since a flow may vary greatly in traffic rate. Compared to RTT and ETT, the radio load metric has lower overhead since it does not require active probing of the links.

6.5.2 Contributions

The main contribution in paper D is showing why the specific layout of a topology may significantly influence the performance of load balancing. In this study, among the simulation results from a set of 30 topologies, the topologies with the best and worst performance are selected for a more detailed analysis. The aim is to abstract the main differences between these topologies that have such an importance on the feasibility and performance of load balancing. For the analysis, we use the *congestion maps* as a tool to visualize the accumulated congestion in the network. The congestion maps are created based on data from simulations, where for each CBR packet transmitted, we increase the background color gradient of the area corresponding to the carrier sensing range of the sender node by 1. We have for simplicity omitted the control packets of the routing protocol and the IEEE 802.11 MAC-layer ACKs in the creation of the congestion maps, since they represent only a minor portion of the total traffic load in the network, both in terms of number of packets and packet size. An example of such a congestion map is shown in Figure 6.7, where the nodes are shown as numbers. The two gateways are denoted as number 0 and 1.

By comparing the maps of the best and worst topologies, a number of important aspects related to load balancing are discovered:

- The congestion maps show the areas of the network that, over time, are the most congested. This is represented by the dark areas in the figure. In contrary to what is commonly believed, the areas around the gateways are not necessary the most congested. The congestion maps show that the area in the centre of the network (i.e. between the two gateways), is actually the most congested. This is due to the reason that the centre area is within the sensing range of a majority of nodes in the network.

- The congestion in the centre area explains why load balancing in wireless environments is often very difficult. In many cases, the congested area represents an obstacle or barrier, preventing traffic load to be efficiently diverted from a congested gateway to another less congested gateway.

- The essential point that determines whether load balancing is efficient or not, is where the nodes that actually reroute the traffic load are located relative to the congested area. In Figure 6.7, the most congested gateway is *GW0*, since more nodes are associated with *GW0* than *GW1*. Thus in order to reduce the load imbalance, part of the traffic load needs to be diverted to *GW1*. The nodes that actually perform rerouting are encircled, and as can be seen, these nodes are located on the correct side of the congested area, i.e. to the right of it. This is very advantageous, since the traffic can be efficiently diverted away from the congested area instead of crossing it. Due to this reason, the throughput enhancement for this topology is about 20 %.

One of the important aspects of the above analysis is that in order to efficiently perform load balancing, not only the gateway loads need to be considered, but it is also important to consider the properties of the paths towards the gateways with respect to the hop distance and the bottleneck capacity. While the gateway loads give us an indication about which gateway more traffic should be rerouted to, the properties of the paths on the other hand, give us an idea about which path we should select in order to avoid crossing the congested area. Based on these observations, a new load balancing scheme is developed. The aim is to verify the correctness of our observations, and second, to provide a better solution for gateway load balancing. We call this new scheme as the Radio Load Based Load Balancing (RLLB) scheme, which relies on the radio load information in making routing decisions. The calculation of the radio load is performed at the MAC-sublayer as previously discussed in Section 4.5 and made available to the routing protocol. Thus the RLLB scheme is a cross-layer based solution, and it is designed to take the above properties into account. This implies that rerouting of traffic is

performed only when appropriate, i.e. when the route to the alternative gateway is not crossing the most congested area. In contradiction to our proposal, many previous proposals do not consider all the above properties when performing load balancing. In most proposals, the load balancing algorithms are only based on one single parameter, i.e. the condition at the gateways.

Furthermore, in our proposal, the RLLB scheme is integrated with the OLSR routing protocol. This has several advantages. First, the calculation of the bottleneck radio load can be easily performed using the Dijkstra's algorithm that is part of the function for routing table calculation. Second, the radio load information can be integrated with the OLSR's control message, i.e. the TC message, in order to take advantage of the optimized flooding mechanism. Third, by using a proactive routing protocol, load balancing can be performed more dynamically in order to adapt to the changes in the network.

Simulation results show that the RLLB metric results in better performance than the SP metric. The average peak enhancement for 30 random topologies is almost 12%. Furthermore, the results also show that RLLB provides a better performance than both the SLB and EL metric in paper C by around 5 % (average peak enhancement). This is due to the fact that the SLB metric is too conservative in performing load balancing. Hence, in many cases, it is incapable to reduce the load imbalance sufficiently. The EL metric is too aggressive in reducing the load imbalance, and may therefore consume too much network resources, which again results in poorer performance. On the other hand, the RLLB metric is designed to adapt the amount of load to divert according to the layout of the topology and the condition in the network, and is thus more optimal than the other metrics.

6.6 Contribution of paper E: A Radio Load Based Load Balancing Scheme with Admission Control

The work in paper E is a continuation of the work in paper D. Previous simulation results show that the proposed RLLB scheme in paper D performs well in static topologies with higher levels of asymmetry. However, the performance is not optimal for topologies with lower levels of asymmetry. This is due to the *synchronized rerouting* problem that may occur in distributed systems. Paper E addresses this problem and proposes a new gateway load balancing scheme to resolve it. In addition to performing gateway load balancing, the new scheme also performs admission control (AC) in order to prevent the network load from reaching a critical high level. While the work in paper D only focuses on static topologies, paper E addresses both static and mobile topologies.

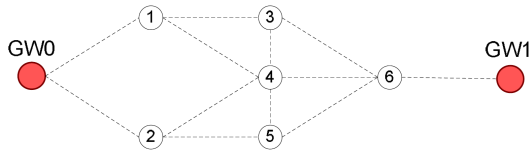


Figure 6.8: Example of topology that is subjected to the synchronized rerouting problem

6.6.1 Related Work

Although there are proposed a number of different schemes for load balancing, but very little attention is given to the challenges that may arise when performing load balancing in a wireless and distributed system. To the best of our knowledge, only a few papers address some of the challenges. The problem of route flapping is previously noticed in [96], and the work in [78] provides a solution for mitigating route flapping based on hysteresis. However, as far as we know, none of these works address the issue of synchronized rerouting.

As stated above, our proposed scheme jointly performs load balancing and admission control. A similar approach is also applied in [97], where the authors propose an admission control scheme with load balancing functionality. In their proposal, the primary goal of their solution is to perform admission control, while load balancing is the secondary goal, i.e. a flow is granted access to the least loaded path if there are multiple paths satisfying the requested bandwidth. On the other hand, the primary goal of our proposal is performing gateway load balancing. Admission control is just a secondary mechanism, with the aim to prevent excessive traffic from entering the network when the network load is high.

6.6.2 Contributions

The main contribution in paper E is showing why the *synchronized rerouting* problem may occur when performing load balancing in a distributed system. To solve this problem, a new load balancing scheme is proposed using a randomized instead of a deterministic gateway selection approach. This new scheme which we call Radio Load Based Load Balancing with Admission Control (RLAC) jointly performs load balancing and admission control.

Since a MANET is a distributed system where there is no centralized administration, each node in the network must take routing decisions on their own and independent of each other. In the context of performing gateway load balancing,

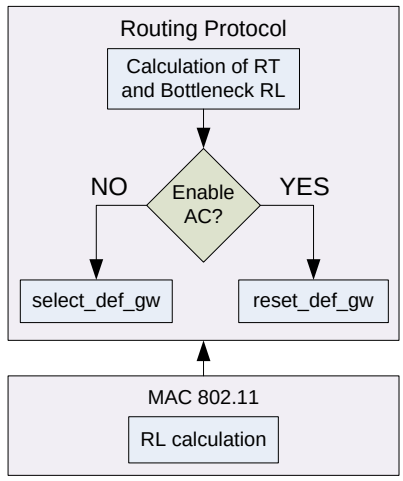


Figure 6.9: RLAC Architecture

the lack of coordination between nodes regarding to which gateway data packets shall be forwarded to, may result in non optimal load distribution and route flapping. To illustrate this, let us consider the scenario in Figure 6.8. Suppose all nodes originate traffic of equal rate to their default gateway. Thus, node 1 and 2 originate traffic to *GW0*, since this is the nearest gateway. Likewise, node 6 selects *GW1* as the default gateway upon which traffic is originated to. At this moment *GW0* is more loaded than *GW1*. Next, node 3, 4 and 5 now have upstream traffic to send. These nodes are aware that *GW0* is more congested than *GW1*, and therefore choose to send traffic to *GW1*. This will eventually result in that *GW1* becomes more congested than *GW0*. When node 3, 4 and 5 are aware of this condition, they will try to rebalance the traffic load between the two gateways, by rerouting their traffic to *GW0*, and once again, causing *GW0* to be more congested than *GW1*. This situation of ping pong effect where a group of nodes simultaneously and repeatedly reroute their traffic back and forth from one gateway to another, is referred to as the synchronized rerouting problem.

Thus to solve the above issue, a new scheme called RLAC is proposed. The architecture of this scheme is illustrated in Figure 6.9. Similar to the RLLB scheme in paper D, RLAC is a cross-layer based scheme, and adapted for proactive routing protocols such as OLSR. One of the fundamental building block of the RLAC scheme, is the radio load information (RL) provided by the underlying MAC-layer. This information is used by the routing protocol to perform the tasks of: i) load balancing (by determining the optimal default gateway), and ii) admission con-

trol (AC). During the calculation of the routing table (RT), the bottleneck radio load to each destination is also calculated using Dijkstra's algorithm. Once this is completed, the scheme determines whether to enable/disable AC on the local node. If AC is not enabled, the function *select_def_gw* is called where the process of default gateway selection is performed. On the other hand if AC is enabled, *reset_def_gw* is called to reset the default gateway.

One of the main differences of RLAC compared to RLLB lies in the gateway selection algorithm. In RLAC, a randomized instead of a deterministic gateway selection approach is used in order to solve the synchronized rerouting problem. The *select_def_gw* function calculates a probability P_0 for selecting *GW0* as default gateway as follows:

$$P_0 = \begin{cases} 0.5 - a \cdot \Delta B - b \cdot \Delta L - c \cdot \Delta h \\ 0 \quad , \quad P_0 < 0 \\ 1 \quad , \quad P_0 > 1 \end{cases} \tag{6.8}$$

where ΔB, ΔL and Δh are the differences in the bottleneck radio load, the gateway radio load and hop distance, respectively. The constants a, b and c are used to weight the above input parameters. The probability P_1 for selecting *GW1* as default gateway is simply:

$$P_1 = 1 - P_0 \tag{6.9}$$

Having calculated P_0 and P_1, a uniform random number $R \in [0, 1]$ is drawn. If $R < P_0$, the *GW0* is selected as the default gateway, otherwise, *GW1* is selected as the default gateway. If we take a look back to the example in Figure 6.8, since *GW0* is initially more congested than *GW1*, then $P_1 > P_0$. Suppose that the calculated probability P_0=1/3 and P_1=2/3. Then in long run, we may expect that 2/3 of the nodes 3, 4 and 5 will select *GW1* while 1/3 will select *GW0* as the default gateway. Thus, using a randomized gateway selection approach, the risk for synchronized rerouting to occur will be lower. To the best of our knowledge, no previous proposals utilize this approach in order to solve the rerouting problem. In fact, most proposals utilize the deterministic approach when performing gateway selection.

The task of the AC mechanism in RLAC is to prevent the load in the network from reaching a critical high level. This is especially important in wireless networks due to the interfering nature of the shared medium, where packet transmissions may be more vulnerable to collisions and loss when the network is overloaded. To prevent this we have implemented a simple AC scheme that works in the following

way: If *B0* and *B1* (bottleneck radio load along the path towards *GW0* and *GW1*) are higher than a given threshold *AC_UPPER*, seen from the point of view of a local node *n*, then AC is enabled on *n* if *n* is located more than *AC_MAX_HOPS* away from the nearest gateway. This implies that if the network load is high, AC is enabled on nodes that are located farther away from the gateways, in order to give priority to nodes closer to the gateways. This design choice may be justified by the fact that the cost of transmitting traffic destined to the gateways is higher for nodes located farther away than for nodes in the proximity of the gateways. Furthermore, when the network load is high, nodes that are located farther away from the gateways will most likely experience a very low packet delivery ratio. Enabling AC on these nodes is reasonable, since it does not matter whether no or only a few packets can reach the destination. Once Admission Control (AC) is enabled, only traffic originated by the local node is discarded, while transit traffic (i.e. traffic not originated by the local node) may still be forwarded.

While previous results show that the RLLB performs not so well for static topologies with lower asymmetry, simulations with the same set of topologies and using the RLAC metric, show that the throughput can in fact be enhanced by approximately 10 % relative to the SP metric. Furthermore, simulations with mobile topologies show that performing load balancing under mobility is extremely difficult. The reason for this includes frequent link breaks, the risk for network partitioning and the latency of the routing protocol in capturing the current topology of the network. Simulation results show that at low mobility levels, i.e. 1-2 m/s, the achieved enhancement relative to the SP metric is at best 4 %. At higher levels of mobility such as 10 m/s, the performance of load balancing is practically zero.

6.7 Concluding Remarks

The work in this thesis is an effort to bring forth solutions in order to improve the performance in a MANET. This is a small step on the way in realizing a future communication system for emergency and rescue operations. However, some of the suggested solutions have a potential for further exploration and optimization.

In paper A, although *Adaptive retry limit* is an effective solution in solving the rerouting problem, the reduction of the retry limit may potentially induce unfairness in the network due to less time in backoff. Thus a further investigation of this aspect is needed. Another way to solve the rerouting time problem is to dynamically adjust the queue length. When experiencing packet drops due to a link break, the queue length can for example be decreased in order to reduce the amount of stale packets inserted into the queue. Furthermore, considerations need also to be

taken for the potential of buffer overflow when reducing the queue size.

The work in paper B demonstrates that transit routing can be very beneficial in terms of increasing the throughput for certain source and destination pairs and reducing the load in the ad hoc subnet. Even though the work in this paper is focused on optimizing the throughput for one single traffic flow, simulations with background traffic also show that the proposed solution is capable to considerably enhance the throughput. However, in order to improve and make the solution more generic, one can for example to incorporate other metrics such as the radio load metric used in paper D and E. With the additional information on the traffic load distribution in the network, it is possible to derive an improved cost metric algorithm that also accounts for multiple traffic flows.

The work in paper C-E investigate factors that affect the feasibility of performing gateway load balancing, and propose two different load balancing schemes. However, the general assumption of these work is a network scenario with two gateways, in which gateway load balancing is performed on outbound traffic only. Although we believe that much of the concepts developed and results obtained through our work are applicable to a more generic scenario, further investigations should be performed on scenarios with multiple gateways and inbound traffic in order to validate and to potentially discover new aspects related to gateway load balancing.

Finally, even though the major part of our work assumes that the architecture of the emergency and rescue network is based on a mixture of MANET and WMN, it is however imaginable that future emergency and rescue communications systems will be a combination of all the three types of ad hoc networks. The idea is to take advantage of the technology provided by each type of networks in order to form a more efficient communication platform for emergency and rescue operations. While the MANET technology can provide mobility and flexibility, the backbone infrastructure of WMNs can increase reliability and robustness in the network. On the other hand, the technology in WSNs can provide the ability to monitor inaccessible or potentially dangerous areas for a variety of environmental parameters such as hazardous gases or radiations. The same technology may also be used to monitor the health conditions of rescue personnel at the emergency scene. Thus a study on the integration of these network architectures would be very interesting to carry out.

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Part II

Research papers

Paper A :

Rerouting Time and Queueing in Proactive Ad Hoc Networks

V. Pham, E. Larsen, K. Øvsthus, P. Engelstad and Ø. Kure

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Rerouting Time and Queueing in Proactive Ad Hoc Networks

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Abstract

In a MANET network where nodes move frequently, the probability of connectivity loss between nodes might be high, and communication sessions may easily loose connectivity during transmission. The routing protocol is designed to find alternative paths in these situations. This rerouting takes time, and the latency is referred to as the rerouting time. This paper investigates the rerouting time of proactive routing protocols and shows that the rerouting time is considerably affected by queueing. Simulations and analysis are conducted to explore the problem. Finally, we propose a MAC-layer solution that reduces the rerouting time problems due to queueing. Simulations and analysis show that the solution is so effective that it eliminates the entire problem in many situations.

1. Introduction

The research efforts in the field of ad hoc networking have been going on for many decades. Ad hoc networking enables communication directly between nodes, without the need for extra infrastructure. This makes it very suitable for military and rescue operations. The standardization of routing protocols has been undertaken by the Mobile Ad Hoc Networking (MANET) working group in IETF [1]. They are set to bring forward two protocols, one reactive and one proactive.

A common characteristic of ad hoc networks is that links may break due to changes in radio conditions, node mobility and other types of network dynamics. The routing protocol is designed to find alternative paths in these situations. The time period before new paths are found is referred to as the rerouting interval, and the duration of the rerouting interval is referred to as the rerouting time.

During the rerouting interval, stale routes exist over the link that has been broken. Rerouting can only take place after the routing protocol has detected that the link is broken. In fact, a significant part of the rerouting time is associated with the detection of the link break.

With proactive routing protocols, such as Optimized Link State Routing (OLSR) and Open Shortest Path First with MANET Designated Routers (OSPF-MDR), a link is maintained by the exchange of control packets. A link break is normally not detected until either a certain number of HELLO packets have been lost, or the lack of periodic updates results in a link timeout [2-4]. (Some implementations might let the link layer detect link breaks and signal this information to the routing protocol. Such cross-layer optimizations are outside scope of this paper. Here, we explore the common layered approach where HELLO packets are necessary for the detection of link breaks.)

With the default parameter settings of OLSR and OSPF-MDR, a link break should normally be detected after approximately 6 seconds. However, we conducted a series of lab experiments of OLSR [3] and OSPF-MDR [4] and observed rerouting times typically in the order of 20 - 40 seconds. Since the rerouting time depended on transmission rates of data traffic and on size of the transmission queues, we realized that the increased rerouting time in our experiment was mainly caused by the queueing of the data packets.

During the rerouting interval, the network layer at the node upstream to the broken link might try to forward data packets over the broken link. Instead, these packets are accumulated in the output queue. Due to the layered design, the link layer (L2) will keep trying to transmit the queued data traffic already designated to the broken link, even after the network layer (L3) has timed out the link. This does not only consume scarce radio resources. It also blocks the MAC layer. Thus the network layer is not able to announce that the link is broken, and the rerouting time increases correspondingly.

Finally, when all the stale data packets designated to the output queue have been dropped, the MAC layer is ready to transmit the link state announcement to

establish new routes throughout the network and to serve packets waiting in the output queue designated to reachable receivers.

In summary, the rerouting time due to link breaks depends on the time to carry out the following processes:

- Detection of a link break
- The emptying of all stale packets from the output queue
- Network-wide link-state announcement to establish new paths

While both link break detection and routing convergence have received considerable attention in the research community, surprisingly little focus has been directed to the effects of queueing. Indeed, the main contribution of this paper is to explore how queueing increases the rerouting time.

The rest of the paper is organized as follows. Section 2 gives background information on relevant technologies. In Section 3 we present the simulation setup, define the rerouting time, and show simulation results. Section 4 gives an analysis of the factors contributing to the rerouting time. Section 5 presents a proposed solution to the rerouting problem and in Section 6 we present some related work. Finally, in Section 7 the conclusion is presented and further work is sketched out.

2. Background

2.1. The MAC layer of IEEE 802.11

Today, IEEE 802.11 [5] is the most widely used wireless local area networking technology. The standard defines a Physical (PHY) layer and a Medium Access Control (MAC) sub-layer, where the latter supports two modes of operation, namely the Distributed Coordination Function (DCF) and the Point Coordination Function (PCF). Since DCF is the most common mode of operation, we focus only on DCF in this paper.

With DCF, the wireless stations (STAs) access the medium in a distributed way, using *carrier sense multiple access with collision avoidance* (CSMA/CA). With the basic access mechanism, each unicast DATA frame is acknowledged with an ACK frame. This is also known as the minimal frame exchange. (Multicast transmissions, however, are not followed by an ACK frame.) With the optional 4-way frame exchange, on the contrary, each DATA frame is preceded by an exchange of a *request to send (RTS) and a clear to send* (CTS) frame. The use of RTS/CTS is particularly useful to avoid collisions due to hidden terminals [6].

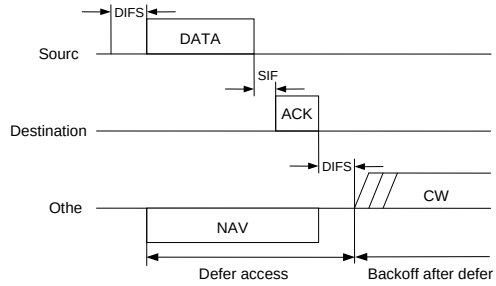

Fig. 1. Basic CSMA/CA.

The basic access mechanism with the minimal frame exchange is illustrated in Fig. 1. When a node has some data to transmit, it has to sense the medium to verify whether it is busy or idle. If the channel is idle for a time interval equal to a Distributed Inter-Frame Space (DIFS), the source node may begin data transmission. When the receiver has received the DATA frame, it waits for a time interval equal to a Short Inter-Frame Space (SIFS), and transmits an ACK back to the source node. While data is being transmitted, all other nodes must defer their channel access for a time interval equal to the Network Allocation Vector (NAV). This is a timer indicating the amount of time that the medium has been reserved for the current transmission. When the data transmission is finished and the NAV has expired, a new contention period is entered. Here, concurrent nodes with pending data traffic must contend for the medium. In this process, each contending node must choose a random time interval called *Backoff_timer*, selected from the contention window (CW) in the following way:

$$Backoff\_timer = rand\left[0, CW\right] \cdot slottime$$

where

$$CW = \left[CW_{\min}, CW_{\max}\right]$$

The value for the *slottime* is dependent on the PHY layer type. The backoff timer is decremented only after each time the medium is idle for a DIFS interval, and is frozen when the medium becomes busy. Eventually when the backoff timer of a node expires, it might transmit data. The main point of this medium access mechanism is to minimize the probability of a collision, i.e. of concurrent transmissions. Since a node must go through a backoff after having transmitted a

frame (also referred to as a *post-backoff*), the medium access mechanism also provides long term fairness to access the medium.

In a wireless environment where collision detection is hard or even impossible, a positive ACK from the receiver is used to confirm a successful transmission. The absence of such an ACK message indicates a collision, link failure or other reasons for an unsuccessful transmission. When this occurs, a retransmission is scheduled, and a new backoff value is chosen. However, in order to reduce the risk for consecutive collisions, after each unsuccessful transmission attempt, the CW is doubled until a predefined CW_{max} is reached.

There is a retry counter associated with the transmission of each frame, and the retry counter is incremented after each collision. After a successful retransmission, the CW is again reset to a predefined CW_{min}, and the retry counter is reset to null.

The maximum number of retransmissions for a frame is defined in the *dot11ShortRetryLimit* and *dot11LongRetryLimit* variables. The first variable is applicable for MAC frames transmitted with the minimal frame exchange (i.e. with length less than or equal to the *dot11RTSThreshold* parameter), while the latter is applicable to frames transmitted with RTS/CTS. For instance, each time a MAC frame of length less than or equal to the *dot11RTSThreshold* is transmitted, and it fails, the *short retry counter* is incremented. This will continue until there is a successful transmission or the counter has reached the dot11ShortRetryLimit and the packet is discarded. When this happens the short retry counter is reset to zero.

For simplicity, throughout the rest of this paper, we will use the term dot11ShortRetryLimit and "retry limit" interchangeably.

2.2. Queueing in the protocol stack

The unicast packets (multicast is considered out of scope) created by applications are passed down the protocol stack to TCP or UDP using the socket interface (Fig. 2). If the packet is a TCP packet, it may be queued to accommodate flow control. For UDP, and TCP eventually, the packet is passed down to L3 (i.e. the IP layer) for routing and designation of a next hop link layer address before passed down to the L2 (i.e. the link layer). There it is queued in the queue of the device driver until the buffer of the network interface is empty, and is then pulled onto the network interface. When the transmission medium is available, the packet is transmitted. If no ACK is received, the packet is assumed lost due to a collision, and the packet will be scheduled for retransmission.

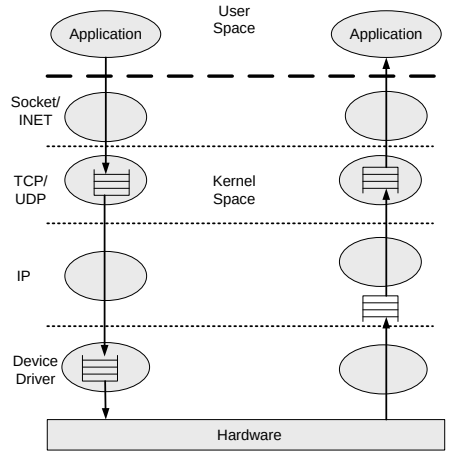

Fig. 2. Linux protocol stack [7].

When a packet is received at an interface, it is put in a backlog queue. Then L3 processes it, and either forwards it out on an interface or pushes it up the stack to UDP or TCP. TCP has a receive queue to serve flow control.

The L2 queue should be of a minimum size to allow traffic to be sent without loss from applications at a rate higher than the network capacity, as the network bandwidth can be variable due to fading, mobility, interference, contention etc.

Both Linux and the network simulator *ns-2* [8] implement a L2 queue for outgoing packets. In ns-2, using the CMU Monarchs wireless extensions, packets are queued in the interface priority queue (IFq). The network stack for a mobile node consists of a link layer (*LL*), an ARP module connected to LL, an interface priority queue, a MAC layer, a network interface, all connected to the channel. When a packet is created by the source application, the packet is queued in the IFq until all previous packets have been either sent or discarded.

2.3. Optimized Link State Routing Protocol (OLSR)

OLSR is a proactive routing protocol for ad hoc networks. The protocol is built around the notion of Multi Point Relay nodes (MPRs). The main purpose of MPRs is to create and forward link state messages. The MPRs are selected individually by each node in the

network in such a way that all nodes can reach their 2-hop neighbor nodes through an MPR.

The two most important message types in OLSR are the HELLO and the TC (Topology Control) messages:

1) HELLO Messages: Every node broadcasts HELLO messages periodically, to support link sensing, detection of neighbors and signaling of MPR selection. The recommended emission interval for HELLO messages is 2 seconds, and the holding time for neighbor information is 6 seconds. Thus a neighbor is considered lost 6 seconds after the last HELLO message received from the neighbor.

2) TC Messages: Based on the information collected through HELLO messages, link state (TC) messages are created and broadcasted throughout the network by each MPR. The recommended emission interval for TC messages is 5 seconds, and the holding time is 15 seconds.

3. Simulations

3.1. Description of the scenario

The scenario explored can be described as follows: Three nodes A, B and C form an ad hoc network where A sends traffic to C at a Constant Bit Rate (CBR). At the beginning, A and B stretch out the network. Then C moves past B, and loses connectivity with A until traffic from A is rerouted via B. In this scenario, C has always direct connectivity with B.

Although the scenario seems simple, it is realistic and sufficient to explore important aspects of the rerouting time. Note also that all nodes are within a two-hop distance of each other. This means that the dissemination of TC messages will not affect the rerouting time, and we are able to explore the rerouting time associated only with the detection of the link break and with the queueing effects.

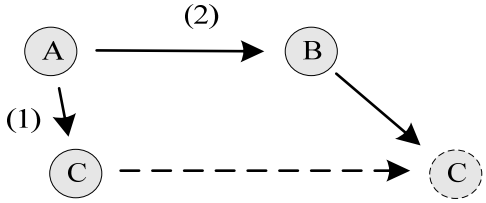

Fig. 3. Conceptual model for the simulations.

3.2. Simulation setup

All simulations were carried out using the network simulator ns-2. In the beginning, all three nodes A, B and C are in the immediate neighborhood of each other. Node A, which is the sender, sends UDP data packets of rate R_{in} packets per second directly to the receiver node C. (This flow is marked with "(1)" in Fig. 3.). While the data transmission is ongoing, node C moves away from node A. At a certain point where node A and C are no longer in the immediate neighborhood of each other, the connection between these two nodes is broken. In order to re-establish connectivity between node A and node C, node A has to reroute the traffic through node B. (This flow is marked with "(2)" in Fig. 3.). B forwards this traffic further on to node C.

In all simulations, the packet size was fixed at 1000 bytes. IEEE 802.11b [9] was used with the basic DCF mechanism (i.e. without RTS/CTS) and a nominal transmission rate of 11 Mbps. The RTS/CTS handshake mechanism is not necessary since there is no hidden node problem in our scenario. All nodes are inside each others sensing range.

Table 1. Simulation parameter settings.

Simulator	ns-2 version 2.30
Radio-propagation model	TwoRayGround
MAC type	802.11b
Interface queue type	FIFO with DropTail
Antenna model	OmniAntenna
Data rate	11 Mbps
Basic rate	1 Mbps
Packet Size IP	1000 Bytes
Movement speed of node C	3.3 m/s
OLSR HELLO_INTERVAL	2 seconds
OLSR REFRESH_INTERVAL	2 seconds
OLSR TC_INTERVAL	5 seconds
OLSR NEIGHB_HOLD_TIME	6 seconds
OLSR TOP_HOLD_TIME	15 seconds
OLSR DUP_HOLD_TIME	30 seconds

The implementation of OLSR by the University of Murcia was used as the proactive routing protocol for ns-2 [10]. In the OLSR configuration, the time interval between HELLO packets was set to 2 seconds, and the HELLO packets were given priority over data packets to avoid route instability. Furthermore, a link is considered down after the loss of 3 consecutive HELLO packets, leading to a detection time of link breaks of approximately 6 seconds:

$$HELLO\_INTERVAL = 2 seconds$$
$$NEIGHB\_HOLD\_TIME = 3 \cdot HELLO\_INTERVAL$$

Essential parameters used in the simulations setup are summarized in Table 1.

3.3. Definition of the rerouting time

In the simulations that were conducted, we mainly focused on measuring the rerouting time, i.e. the time duration from when the link between A and C is broken to the time when connectivity is re-established via the intermediate node B. However, our experience through many experiments - both in a real test-bed and in simulations - is that the rerouting time measured in this way will have a high degree of variance caused by random effects during rerouting. In order to minimize variance in the measurements, we have chosen to define the rerouting time $t_{reroute}$ as the time interval from the last HELLO message from node C received by node A before link break, to the moment where the connectivity is re-established, i.e. until the instant of time where the first UDP packet is received at C after the link break.

3.4. Simulation results

The results from the simulations for various retry limits (Fig. 4) show that a higher retry value gives a longer rerouting time. This is as expected, because each packet in the L2 queue is transmitted a number of times defined by this retry value. We also notice that the rerouting time is linearly proportional with the L2 queue size.

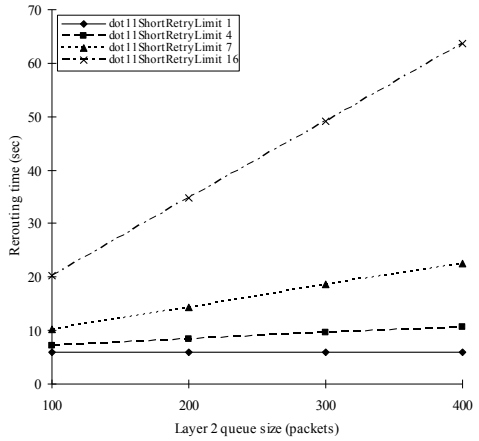

Fig. 4. Simulation results of rerouting time over layer 2 queue size.

Fig. 5 shows simulation results for the rerouting time as a function of the transmitted packet rate (marked as crosses, squares and triangles). Here, the retry limit is set to 7, and the queue size is set to 100, 400 and 1000 packets. The figure shows that for small packet rates, the rerouting time is at the minimum value of 6 seconds, which equals to the NEIGHB_HOLD_TIME. As the packet rate increases, the rerouting time also increases linearly up to a certain point where it suddenly stops to increase, and the rerouting time stabilizes at its maximum value. The maximum rerouting time depends on the queue size. For a queue size of 100, the maximum rerouting time is slightly more than 10 seconds. For a queue size of 400 it is nearly 23 seconds, while for a queue size of 1000 the maximum rerouting time lies around 47 seconds. With a queue size of 400, we see that at packet rates of 100 pkts/sec and over, the queue is filled at the time when rerouting takes place, and this results in a rerouting time converging on approximately 23 seconds.

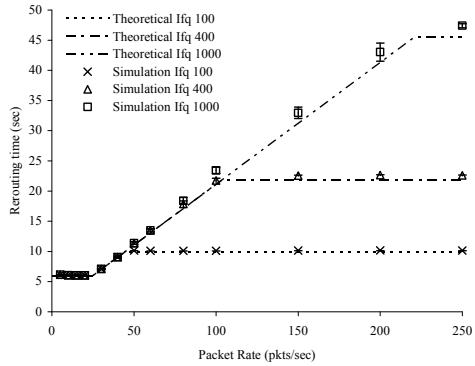

Fig. 5. Simulation results for rerouting time over packet rate for layer 2 queue size 100, 400 and 1000 packets, 7 MAC retries. (95% conf. int.)

It is also observed (Fig. 5) that at low rates (i.e. well below 20 pkts/sec) the rerouting time is flat at 6 seconds.

4. Analysis

4.1. Analysis of the problem

From the log file produced by ns-2 we can observe various incidents affecting the rerouting time. These incidents, which occur at node A, are illustrated in Fig. 6 and are explained below:

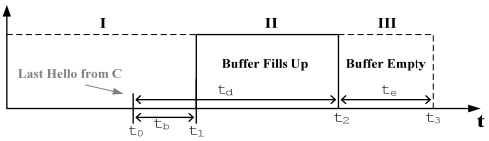

Fig. 6. Illustrating different incidents at node A's queue as function of time.

1) In region I, data packets are continuously inserted into the transmit queue at node A. At time t_0, the last HELLO message from C is received at A (short line in the figure). Then, after t_b seconds, the direct link between A and C is broken at t_1.

2) Although the link between A and C is broken, the routing protocol is still not aware of this, and therefore has not updated the routing table. As a result, "garbage" data packets with stale routing information continue to be put into the queue at node A.

3) In region II, i.e. $t_1 < t < t_2$, the queue at node A is being filled up. This happens since each garbage data packet in the queue at node A is retransmitted L times, where L is the retry limit. Because of all the retransmissions for each packet, the packet rate out of queue R_{out} will be reduced considerably. As long as the packet rate R_{in} into the queue is higher than R_{out}, the queue will be filled up. This will last for $t_d - t_b$ seconds, where t_d is the timeout value for the routing protocol's HELLO packets (which is equivalent with *NEIGHB_HOLD_TIME* in OLSR).

4) At t_2, garbage packets are no longer put into the queue. The routing protocol has now updated the routing tables. New data packets are instead correctly rerouted to B.

5) In region III, i.e. $t_2 < t < t_3$, the queue is being emptied for garbage packets. This will last for t_e seconds, depending on parameter values like R_{in}, L, packet size, queue size etc.

Note that packets are attempted transmitted and removed from the queue both in region II and region III. Thus, the queue will fill up in region II only if $R_{in} > R_{out}$. However, for the lowest packet rates, we will have $R_{in} < R_{out}$, and the queue will not be filled. In the latter case, both region II and region III will be non-existent, and the routing interval consists of only region I. This explains why the rerouting time is flat at 6 seconds for the lowest packet rates in Fig. 5.

In summary, the incidents in the time interval $t_0 < t < t_3$ are the main contributions to the rerouting time as defined above. It is also worth noting that in our test scenario, the delay from a TC message is irrelevant for the rerouting time. This is due to the fact that prior to the link break, node A will have node C in both its 1-hop and 2-hop neighbor sets. When A discovers a link break between A and C, A still has a route to C in its 2-hop neighbor table, i.e. through node B. Therefore, node A does not need to wait for any TC message from B in order to figure out how to reach C.

4.2. A model for the rerouting time

Based on the observations above we have derived a simple analytical model that can be used to predict the rerouting time. The derivation of the model is given below:

t_{difs}	DCF interframe space
t_{bo}	backoff time
t_{data}	delay for transmitting the data packet
t_{sifs}	short interframe space
t_{ack}	delay of acknowledge
T_e	slot time in IEEE 802.11
t_{RTS}	delay of a RTS packet
t_{CTS}	delay of a CTS packet
L	number of retries
B	queue size

1) According to the 802.11 standard, the delay of attempting to transmit a single packet over a broken link is

$$t_{packet} = T_c + t_{bo} \tag{1}$$

where T_c is the delay associated with the transmission attempt, and t_{bo} is the delay associated with the backoff. In our scenario, no ACK is received when the link is broken, and the transmission attempt is therefore perceived as a collision. However, according to the standard, a node must wait an ACKTimeout amount of time without receiving an ACK frame before concluding that the transmission failed. In our case, this ACKTimeout corresponds to the transmission of an ACK for a successfully transmitted frame. Thus, the delay associated with the transmission attempt, T_c, is equal to the delay associated with a successful transmission, T_s:

$$T_c = T_s = t_{difs} + t_{data} + t_{sifs} + t_{ack}. \tag{2a}$$

With the RTS/CTS mechanism, on the contrary:

$$T_c = t_{difs} + t_{RTS} + t_{CTS\_timout}. \tag{2b}$$

Prior to each packet transmission, a backoff time is uniformly chosen in the range (0, W_j -1). Here we

define W_j, where $j \in (0, m)$, as the contention window at "backoff stage" j, and m is the number of the maximum backoff stage. Let us also define L as the number of retries, and we can thus write the definition of the contention window as:

$$W_j = \begin{cases} 2^j W_0 & L \le m \\ 2^m W_0 & L > m \end{cases}. \quad (3)$$

Eq. (3) states that for the first transmission attempt, the contention window is W_0 which is equal to CW_{min}. (Note that this definition of the contention window is slightly different from the definition in Section 2. In fact, the IEEE 802.11 standard refers to W_j-1 as the contention window [5]. For convenience, we have defined the contention window differently in this paper.) After each unsuccessful transmission, the contention window is doubled, and the packet is attempt retransmitted. This will continue until we reach the maximum contention window $W_m = 2^m W_0 = CW_{max}$, where it remains for consecutive retransmission attempts. If a retransmission is successful after a number of retries, or the number of retransmission has reached the retry limit, the contention window is again reset to its initial backoff stage W_0.

In our scenario, when the link between A and C is broken, each "garbage" packet in the queue is retransmitted L times, and eventually is discarded because the maximum number of retries has reached.

The mean total delay for one single packet with L retries is then approximately:

$$t_{packet\_L} = (L+1) \cdot T_c + \sum t_{bo} \quad (4)$$

where

$$\sum t_{bo} = T_e \cdot \begin{cases} (\frac{W_0}{2}) \cdot \left[2^{L+1} - 1\right] - \frac{1}{2}(L+1) & L \le m \\ (\frac{W_0}{2}) \cdot \left[2^m \cdot (L - m + 2) - 1\right] - \frac{1}{2}(L+1) & L > m \end{cases} \quad (5)$$

which is the sum of the approximate mean backoff time. Here, T_e is the slot time. Note that T_e, W_0 and m are parameters that depend on the PHY-layer used. For 802.11b, T_e = 20 µs, W_0 = 32 and m = 5.

We have intentionally tried to keep the scenario as simple as possible, to derive a simplified model that is intuitive and easy to analyze. One of the simplifications made is the assumption that A is the only node trying to access the medium when the link is broken. Thus, during backoff the medium is always idle, and the duration of each backoff state is therefore T_e. It is not difficult to extend our analysis for the case when multiple nodes contend for the same medium. In [11], for example, Engelstad and Østerbø calculated the queueing delay by applying a Bianchi model that is extended to non-saturation conditions. Thus, extending our analysis is not hard to do, but draws attention away from the main objective of this paper. It is also considered out of scope due to space limitations, but might be addressed in a follow-on publication.

2) The packet rate R_{out} out of queue when each packet has to be retransmitted L times, is therefore:

$$R_{out} = \min\left[\frac{1}{t_{packet\_L}}, R_{in}\right] = \min\left[\frac{1}{(L+1) \cdot T_c + \sum t_{bo}}, R_{in}\right]. \quad (6)$$

3) The total rerouting time is:

$$t_{rerouting} = t_d + t_e, \quad (7)$$

where (depicted in Fig. 6):

$$t_e = \frac{1}{R_{out}} \cdot \min\left[(t_d - t_b) \cdot (R_{in} - R_{out}), B\right]. \quad (8)$$

4.3. Discussion

Eq. (7) equals the rerouting time as defined above, where only the most significant mechanisms contributing to the total delay of the rerouting time is considered. This delay is equal to $t_3 - t_0$ in Fig. 6. Here, we assume that the delay of transmitting one single packet through the alternative path, from A to B and then to C, is very small compared to t_d and t_e. This delay is therefore omitted in the equation.

The first term of the equation is a constant defined by the proactive ad hoc routing protocol configuration (this is equivalent to the NEIGHB_HOLD_TIME in OLSR). This value is also the absolute minimum rerouting time. The second term is variable, depending on parameters like R_{in}, the retry limit L, the queue size B, etc.

From Eq. (7) and Eq. (8) it is clear that there is a lower and an upper limit on the rerouting time. The lower limit occurs when $R_{in} = R_{out}$, in Eq. (8). Thus, for the lowest packet rates the rerouting time is equal to t_d, as we also observed in the simulations.

The upper limit occurs when the queue is filled and is constrained by the queue size B. Hence, for the highest packet rates (i.e. when $R_{in} > B/(t_d - t_b) + R_{out}$) the maximum rerouting time is:

$$t_{rerouting\_max} = t_d + \frac{B}{R_{out}}. \quad (9)$$

Furthermore, in the case when the rerouting time is larger than t_d and smaller than $t_{rerouting\_max}$, Eq. (7) yields:

$$t_{rerouting} = \frac{R_{in}}{R_{out}} \cdot (t_d - t_b) + t_b. \quad (10)$$

This reveals that the rerouting time is linear and proportional to R_{in} in this region.

Table 2. Comparison of the delay components of R_{out} and the resulting value for R_{out}. Values are given in milliseconds for the delay terms. R_{out} is given as packets per second.

L	0	1	2	3	4	5	6	7
$\sum t_{bo}$	0.31	0.94	2.21	4.76	9.87	20.1	30.3	40.56
$(L+1)T_s$	1.32	2.65	3.97	5.3	6.62	7.94	9.27	10.59
R_{out}	100	100	100	99.44	60.64	35.66	25.25	19.55

The packet rate out of the transmit queue R_{out} is also an important parameter for the rerouting time. A decreasing R_{out} means an increasing rerouting time. By inspecting Eq. (6), we see that the first term in the denominator is linearly proportional with L, while the second term is increasing exponentially with L [Eq. (5)]. This means that the second term will grow much faster than the first term, and therefore will be the dominating term when L is large. This is illustrated in Table 2 where only the results for the eight first retry values were calculated. Here, a packet size of 1000 bytes with a transmission rate of 11 Mbps was used to calculate the delay of t_{data} (in T_c) in Eq. (6). The rate in R_{in} was set to 100 packets per second.

The results from Table 2 show that for the given setting, R_{out} is rapidly decreasing for retry values above 4.

A plot of the estimated rerouting times based on Eq. (7) is shown for three different queue sizes (100, 400 and 1000 packets) as dashed curves in Fig. 5. The curves were calculated using a value of t_b = 0.9 seconds, which corresponds to the average t_b value observed in the simulation results shown in the figure. As the result shows, the estimated rerouting times are almost equal to the simulated results obtained from ns-2. This verifies that the derived formula is a good approximation for the expected rerouting time in the given scenario. We observe, however, that the simulations give a slightly higher rerouting time. This can be explained by the ARP request burst triggered by all packets sent to the Layer 2 in the time lapse from the route through B is chosen, until node B's MAC address is obtained. This behavior of ns-2 is a violation of the recommendations given in [12]. The ARP storm problem is bigger for higher packet rates and larger queue sizes, which can be observed in the figure.

5. Proposed solution

5.1. Adaptive retry limit

At the time the routing protocol becomes aware that the direct connection to the destination has been broken, the packets in the L2 queue no longer have a reachable link layer destination. These packets will be discarded only after being transmitted onto the medium for a number of times defined by the IEEE 802.11 dot11ShortRetryLimit. We argue that a solution to this problem should be implemented as a layered solution, to keep it as small and simple as possible. The link layer protocol will be able to detect the link break earlier than the routing protocol, so it is natural to implement a solution at the link layer. Our analysis shows that at the link layer it is the queue size and the retry limit that are the main contributors to the extended rerouting time. Reducing the queue size could be an option, but to have any effect, this reduction would have to be initiated as soon as the queue usage starts to grow. In this case it would be more efficient to keep the queue small at all times, instead of varying it, but this would restrain the flexibility of having a large queue.

Instead, we propose a solution to the accumulated queue time problem by introducing an adaptive retry limit into the IEEE 802.11 DCF MAC. For each successive packet with the same destination MAC address that is discarded due to reaching the retry limit, the retry limit is reduced by 1, until each packet is only attempted transmitted 1 time. If the original retry limit is 7, the retry limit is reduced to 0 after 7 consecutive packets are dropped due to reaching the retry limit. As soon as a packet is transmitted successfully, the retry limit is reset to its original value equal to that of the legacy IEEE 802.11 standard.

5.2. Discussion

It is very rare that many retry counter expirations occur directly following each other, unless something is wrong. To lower the retry limit gradually will probably not affect the functionality of the 802.11 MAC under normal network conditions. However, a problem with the adaptive retry limit solution is that it

might lead to an unfair resource distribution in terms of collision avoidance. The node sending packets that go unacknowledged will be able to contend for the medium with a high probability of a smaller backoff-counter than its peers. On the other hand, the emptying of stale packets from the queue takes place in a small period of time, and it is much more efficient to send a garbage packet only one time, than sending it multiple times. Another drawback is that the transmission attempts of garbage packets consume network resources. More complex solutions where the MAC layer discards packets without attempting to transmit them is certainly also possible. In summary, there are a number of variations of the proposed adaptive retry limit solution. The performance of a number of these variations in various networking scenarios will be detailed and discussed in a follow-on publication.

5.3. Implementation

To do the actual implementation in ns-2 we needed to introduce two new variables. The first of these new variables, *IDt,* keeps track of the destination of the last transmission attempt, and the second variable, called *PCnt*, counts the number of packets discarded because the retry counter has reached the retry limit.

Each time a packet is discarded because the retry counter has reached the retry limit, the *PCnt* is increased, until it reaches the value of the retry limit.

The *PCnt* is subtracted from the original retry limit, so that the effective retry limit gets lower and lower as the *PCnt* increases, until new packets are only transmitted once, and then discarded if not acknowledged by the receiving node.

If a packet is transmitted to a new destination, *PCnt* is set to 0 and *IDt* is updated. If the transmission was successful (indicated by a received ACK), the *PCnt* is set to 0.

5.4. Simulation results

In the simulation results of the adaptive retry limit solution (Fig. 7, with L2 queue size 400 packets and 7 MAC retries) we observe that with the proposed solution the rerouting time is kept at 6 seconds (which equals to NEIGHB_HOLD_TIME) until the packet rate exceeds 600 pkts/sec. At this packet rate the bit rate approaches the theoretical maximum throughput (TMT) of 5.03 Mbps (for 1000 bytes sized packets, and for a network with one sender, where backoff time has to be taken into account). When the packet rate is higher than TMT, the L2 queue gets filled also when the link between A and C is not broken. This is because R_{out} is smaller than R_{in} at all times.

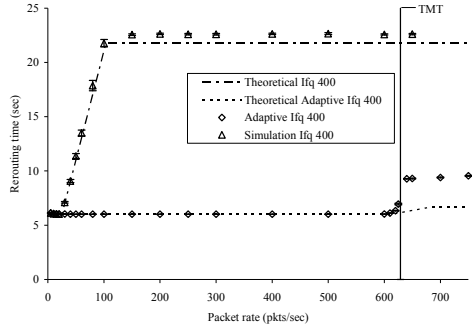

Fig. 7. Simulated results for solution with adaptive retry limit. (95% conf. int.)

Our solution prevents R_{out} from decreasing when the link between A and C is broken, by limiting the accumulated number of retransmissions. The results from the simulations have proven that the adaptive retry limit solution effectively eliminates the delay related to the queueing problem.

The proposed solution is a layered approach based at the link layer, but in some cases it would be more convenient to solve the problem at the IP-layer. This is left for further work.

6. Related work

An analysis of several neighbor sensing approaches is presented in [13]. The objective is to better be able to optimize performance in an OLSR network.

In [14], the OLSR routing protocol is evaluated through both simulations and experiments. Both route-flapping and control packet collisions are described, and solutions for these problems are proposed.

The tuning of routing protocol parameters in order to improve the end-to-end connectivity is studied in [15]. A performance metric called Routing Change Latency (RCL) is defined and analyzed. This metric is defined as “the time needed to determine a new route after a link failure”, but it also comprises a time lapse after the new route is discovered, until it is actually used. This time lapse is denoted as T_{new_route}. It is not explained, but observed to vary between 4.62 s and 8.86 s

7. Conclusions and further work

The rerouting time is an important performance measure in MANETs where node mobility is usually high, and connectivity between nodes may be disrupted frequently. For ongoing data traffic that suffers from link failures, it is highly desirable to reestablish connectivity through alternative paths as fast as possible. In this paper we have looked closer on a simple scenario where we have identified that queueing is among the main factors having considerable impact on the rerouting time.

The latency related to queueing is mainly affected by two parameters, namely the transmit queue size and the retry limit. A large transmit queue size may result in a too high amount of garbage packets with stale routing information being inserted into it. In addition, a high retry value may result in too many wasted retransmission attempts for these garbage packets. The combination of these factors might extend the rerouting time considerably.

We have derived a simple model that can be used to estimate the rerouting time. Comparisons of the estimated and simulated rerouting times have shown that the model is a good approximation. The analysis is used to explain how queueing might increase the rerouting time. In order to solve this problem, we have proposed a simple but very effective solution based on adaptive retry limit in the 802.11 DCF MAC. The queueing problem is resolved by decrementing the maximum retry value when successive packets for the same MAC destination are discarded due to expiration of the retry limit. The proposed solution was implemented and tested in simulations, and the results have shown how effective it can be. In fact, as long as the data rate into the queue is safely below the capacity of the MAC, the solution eliminates the queueing problem associated with the rerouting time.

Although the proposed solution seems to be very effective, there might be some problems associated with it. For example, the solution might lead to an unfair resource distribution in terms of collision avoidance. This needs to be explored in detail, and will be addressed by a follow-on publication.

It might also be possible to implement more complex solutions where the MAC layer discards packets without attempting to transmit them. Various variations of our solution will also be studied.

The proposed solution is a simple way to resolve queueing related delays. We believe there are other possibilities in solving the problem or improving the existing solution. A solution based on cross-layering, where L2 can send a notification up to L3, helping the routing protocol to detect link breaks much earlier is an exciting area. All this is also left to future works.

8. Acknowledgment

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Paper B :

Routing of Internal MANET Traffic over External Networks

V. Pham, E. Larsen, Ø. Kure, P. Engelstad,

Mobile Information Systems Journal, iiWAS/MoMM special issue, Volume 5, Number 3, 2009

Is not included due to copyright

Paper C :

Performance Analysis of Gateway Load Balancing in Ad Hoc Networks with Random Topologies

V. Pham, E. Larsen, P. E. Engelstad, Ø. Kure

In proceedings of The 7th ACM International Symposium on Mobility Management and Wireless Access (Mobiwac09), Tenerife, Canary Islands October 26-30, 2009

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Paper D :

Gateway Load Balancing in Future Tactical Networks

V. Pham, E. Larsen, Ø. Kure, P. Engelstad

In proceedings of The IEEE Military Communications Conference (MILCOM), San Jose, CA, USA, October 31–November 4, 2010.

Gateway Load Balancing in Future Tactical Networks

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***Abstract*—In future tactical networks, gateway nodes will have an important role in connecting different military communications platforms together to form a consolidated network. To increase capacity for upstream/downstream traffic as well as resiliency, more than one gateway should be deployed. In this context, performing gateway load balancing is vital in order to take full advantage of the resources available and thereby improve the performance. Previous work has shown that a number of factors such as the level of asymmetry, offered load, and gateway location may influence the performance of load balancing. However these parameters alone cannot explain why the performance of load balancing is high for certain topologies while it is very poor for others. Obviously, the specific layout of a topology also plays a crucial role on the efficiency of load balancing. We question what are the differences between topologies where load balancing is efficient from the topologies where it is inefficient? The work in this paper thus aims to find the answer to this question, and to explore the nature of performing load balancing in wireless multi-hop networks. Through the knowledge acquired we propose a *Radio load based Load Balancing* scheme (RLLB). Simulations of many randomly generated topologies show that the performance of RLLB is promising.**

I. INTRODUCTION

The advances in Information Technology have motivated for a transformation from a platform-centric towards a Network-centric warfare [1]. The key is more efficient information sharing and improved shared situational awareness facilitated by the underlying network communication infrastructure. In order to accommodate this vision, future tactical networks must support internetworking in multi-tiers networks and/or between heterogeneous communication platforms. An example is internetworking between troop Mobile Ad Hoc Networks (MANET) [2] and remote Tactical Operations Center via high capacity quasi-static backbone networks. Furthermore, internetworking between allied forces is also a highly desirable capability.

In this context, the gateway node plays an important role as a bridge between different networks domains. Since all upstream/downstream inter-domain traffic must traverse the gateway, this node will often become the bottleneck in the network. Furthermore, in a combat scenario, with a single gateway available, the network has a single point of failure and is vulnerable to loss of connectivity. Thus to improve the network in terms of capacity for inter-domain traffic as well as resiliency, more than one gateway should be deployed. However, in order to take full advantage of the increased capacity that comes with multiple gateways and achieve higher network performance, performing load balancing between gateways is vital.

In the literature, there are a number of proposals suggesting various load balancing schemes targeted for MANETs [3-6] and Wireless Mesh Networks (WMNs) [7-10]. These proposals are in general based on a variety of techniques for evaluating the network load, such as RTT [10], average queue length [9,11,12] and number of active flows [13,14]. Furthermore, load balancing is commonly classified into two categories: *multipath* and *gateway load balancing*.

In multipath load balancing [3,4], the traffic load between a source node and a destination/gateway is distributed among a set of alternative paths in order to maximize throughput performance and minimize the impact of route failure. However, [3,4] report that multi-path load balancing in single channel wireless networks only provides a negligible improvement in the performance due to route coupling among the alternative paths. Multipath load balancing is therefore not of interest in this paper.

On the other hand, with the gateway load balancing approach, the traffic load is attempt distributed between gateways in order to reduce the load imbalance and to maximize the total network throughput. Such gateway load balancing is considered to improve the network performance more effectively than multipath load balancing [7]. In this paper, we therefore focus only on gateway load balancing, and we will refer to it simply as "load balancing" in the remainder of the paper.

The previous work in [15] has shown that a number of parameters such as the level of asymmetry (in node and load distribution), offered load, gateway distance, carrier sensing range, may influence the performance of load balancing. However these parameters alone cannot explain why for certain topologies, load balancing may considerably improve the throughput, while for others, the improvement is very poor. This indicates that the specific layout of a topology is another decisive factor for whether load balancing is efficient or not. The question is what is the difference between topologies where load balancing is efficient from those that are not? The work in this paper is thus concerned with the answers to this question through the analysis of static topologies. We argue that using static topologies for this purpose are suitable, in which the direct relation between the specific layout and the performance of load balancing may be revealed. With mobile topologies, this would not be possible. However, the insight and knowledge gained through this analysis may serve as a building block in the work of performing load balancing in mobile topologies.

Through the knowledge learnt, we developed the *Radio-load based Load Balancing* scheme (RLLB) as a means to verify the results of the analysis. Simulation results show that the proposed scheme has a promising performance.

Finally, the study in this paper in based on IEEE 802.11 MAC- and PHY-layer [18] due to availability and secondly, we believe that the demand for high bandwidth capacity for supporting services such as video streaming is highly relevant

in future tactical networks. This will make high frequency, low range, and high bandwidth radio technologies similar to IEEE 802.11 or WiMAX [19] more attractive in future military tactical networks.

The rest of this paper is organized as follows. In Section II, background information and preliminary analysis are given. Section III presents the proposed load balancing scheme. An evaluation of RLLB is presented in Section IV. Finally, the conclusion of the paper is given in Section V.

II. Background and Preliminary Analysis

In order to uncover the reason why load balancing considerably improves the throughput for certain topologies, while there is no improvement at all for others, we initially conducted simulations on 30 static and randomly generated topologies. The general network model for the topologies is as shown in Fig. 1. Each topology consists of 50 nodes and 2 gateways, confined in an area of 1400 m x 800 m. The gateways are symmetrically deployed 1000 m apart. Furthermore, all nodes were configured to send CBR traffic of the same rate toward an appropriate gateway for 250 seconds.

We argue that using static topologies and CBR traffic is the best way to gain insight and understand how the specific topology layout may affect load balancing and the performance. If using mobile topologies and time-varying traffic flows such as TCP, the additional dynamic would increase the complexity and blur the picture.

For the purpose of the analysis we created *congestion maps* using the data from the simulations. Fig. 2 and 3 show two examples of such congestion maps. The topology in the first figure had the best results in terms of improvement in throughput while the topology in the second figure was one of the topologies with lowest improvement. The congestion maps were created as follows: for each CBR packet transmitted, we increase the background color gradient of the area corresponding to the carrier sensing range of the sender node by 1. We have for simplicity omitted the control packets of the routing protocol and the IEEE 802.11 MAC-layer ACKs in the creation of the congestion map, since they represent only a minor portion of the total traffic load in the network, both in terms of number of packets and packet size. In Fig. 2 and 3, the 2 gateways are assigned number 0 and 1, while the remaining sender nodes are numbered from 2 to 51. Furthermore, nodes that have a shorter hop distance to gateway 0 (*GW0*) are colored violet while nodes closer to gateway 1 (*GW1*) are colored blue. Nodes that have the same hop count to both gateways are colored red. From Fig. 2 and 3 we can draw the following observations:

1) When all traffic in the network is destined towards the gateways, it is intuitive to expect that the area around the gateways is the most congested. However, Fig. 2 and 3 show that the dark area near the centre of the network actually is the most congested area. This is because the centre area is within the sensing range of a majority of the nodes in the network. This means that a node located in the centre area is more exposed to interfering transmissions compared to nodes located in the periphery.
2) The congestion in the centre area of the network is the reason why the performance of load balancing is low for many topologies. In many cases, the congested area represents an obstacle or a barrier, preventing traffic load to be efficiently diverted to the less congested gateway. Furthermore, we observed that diverting traffic to one gateway or the other did not significantly change the congestion in the centre area. However, for nodes in the proximity of the gateway in which excess traffic is diverted away from, the level of congestion is alleviated.
3) For both topologies in Fig. 2 and 3 the distribution of nodes is asymmetric such that more nodes are associated with *GW0* compared to *GW1*. Thus for both cases, part of the load has to be diverted to *GW1* in order to reduce load imbalance. However, the reason why load balancing results in high improvement in throughput (20% improvement compared to shortest hop count metric) for the topology in Fig. 2 is because most of the nodes that performed the rerouting of the traffic (encircled) are located on the correct side of the congested area, i.e. to the right of the congested area. As a consequence, part of the excess traffic to *GW0* is efficiently diverted away from the congested area towards the less loaded *GW1,* resulting in higher aggregated throughput. On the other hand, the nodes that performed the rerouting of traffic in Fig. 3 are either located on the wrong side of the congested area, or located within the congested area. For the first case, rerouting traffic to *GW1* will only result in worse performance, since this implies that the traffic has to cross the congested area, i.e. traversing even more congested bottleneck nodes. For the latter case, routing the traffic to either *GW0* or *GW1* will probably not make any big difference, since the load balancing node itself is the most congested node. In this case, routing the traffic to the nearest gateway is perhaps the most optimal choice, since this does not require any additional resources. The reasons just discussed explain why the performance of load balancing for the topology in Fig. 3 is actually lower (-2%) compared to a shortest hop count metric.
4) Another reason why load balancing has a higher performance for the topology in Fig. 2 compared to the topology in Fig. 3 is because the level of asymmetry (both in terms of node distribution and load distribution) is higher for the first topology compared to the latter (In Fig. 2, 11 nodes participate in the rerouting of the traffic to *GW1*

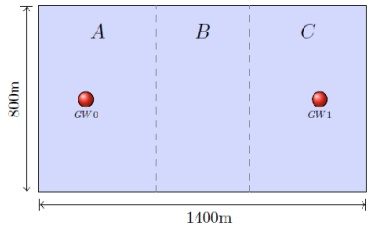

Fig. 1. The network model

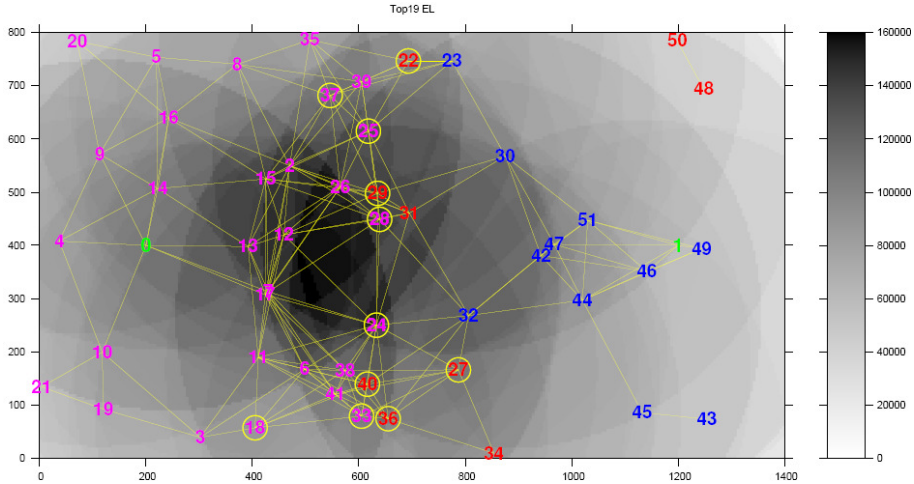

Fig. 2. Example of topology where the performance of load balancing is efficient (20%)

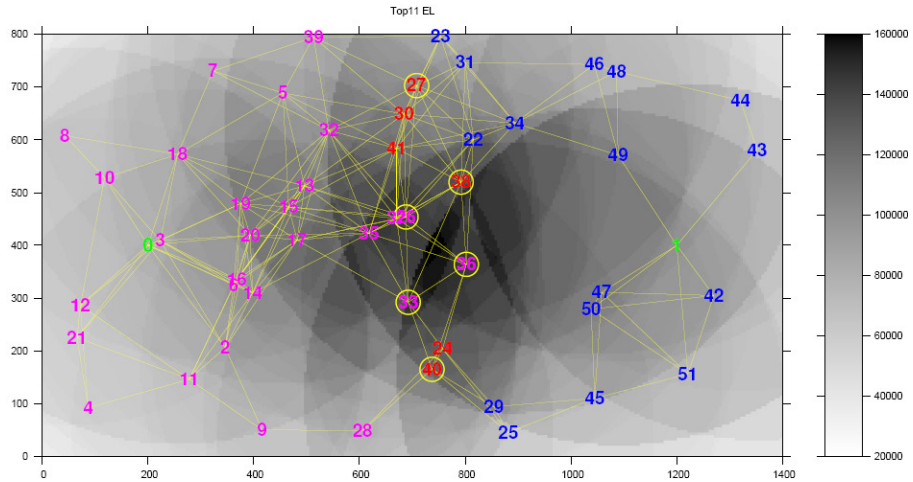

Fig. 3 Example of topology where the performance of load balancing is poor (-2%). Note that node 3 and node 26 are located side by side, and should not be mistakenly interpreted as node 326.

compared to 6 in Fig. 3). This conclusion is in accordance with the result in [15] where it is showed that with increasing asymmetry, the potential for improving the throughput is also higher.

III. Proposed Load Balancing Scheme

In order to achieve an optimized and higher aggregated throughput for the inter-domain traffic, e.g. traffic from a troop MANET to a quasi-static backbone network, it is obvious that the routing protocol must utilize a more intelligent metric instead of the traditional shortest path metric, when there are more than one gateway available. The routing protocol must be capable of performing load balancing by diverting traffic from an overloaded gateway to another under-utilized gateway and thereby improve the load distribution between gateways. However, designing an efficient load balancing mechanism for wireless multi-hop networks is a challenging task due to the interfering nature of the shared medium. In fact, the analysis above has shown that, for certain nodes, inappropriately commencing rerouting may result in a lower throughput. Therefore, the routing protocol must be able to decide when and for which node it is appropriate to commence the rerouting in order to avoid crossing the congested area situation described above. To address this, we propose the RLLB scheme. The scheme essentially consists of 3 functions: i) calculation of the radio load and dissemination. ii) calculation of the routing table and bottleneck radio load. iii) selection of the optimal default gateway. These functions are discussed in detail below. For the discussion we use the *Optimized Link State Routing*

Protocol (OLSR) [16] as a reference routing protocol. However we believe that the same idea is also applicable to similar proactive routing protocols.

A. Calculation of Radio Load and Dissemination

The radio load is a measure for how busy the medium around a node is. If the radio load is high, it indicates either that the local node is transmitting a large amount of traffic, and/or nodes within the sensing range of the local node are sending a high amount of traffic. We define the radio load as the amount of time T_{busy} within a time window T_{window} where the local channel is monitored busy. To estimate the average radio load L we use the exponential moving average as follows:

$$L_{new} = \alpha \cdot L_{previous} + (1-\alpha) \cdot \frac{T_{busy}}{T_{window}} \quad (1)$$

where α is the weighting factor defined as $\alpha \in [0,1]$.

Each node in the network monitors the channel and calculates the perceived local radio load. This information is made available to the routing protocol, which is then responsible for disseminating this information throughout the network. In this paper, the RLLB scheme is integrated with the proactive OLSR routing protocol, and in order to minimize control traffic overhead, the radio load information is therefore disseminated using a modified version of the TC message. Upon receiving TC messages, the radio load information is stored in a local repository for later use by RLLB.

B. Calculation of Routing Table and Bottleneck Radio Load

The calculation of the routing table is performed in the same way as in the ordinary OLSR implementation, i.e. based on Dijkstra's algorithm. However, during the calculation of the routing table, the bottleneck radio load B_i is also calculated for each destination D_i that is added into the routing table. B_i is defined as the highest observed value of radio load along the path from the local node to the destination D_i. In order to facilitate this, we introduce a new field *R_radio_load* in the routing table. This field stores the value of B_i for a given destination D_i in the *R_dest_addr* field.

C. Selecting the Default Gateway

Using the radio load information that is disseminated in the network, a local node may determine the gateway which is the most optimal and then select it as the default gateway.

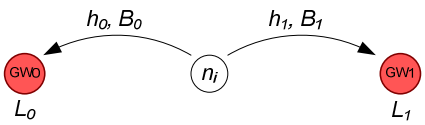

Fig. 4. Model for gateway selection

Consider the scenario in Fig. 4. A local node n_i is located h_0 and h_1 hops from gateway *GW0* and *GW1* respectively. The reported radio load at gateway *GW0* and *GW1* is L_0 and L_1. B_0 and B_1 are the bottleneck radio load to *GW0* and *GW1* respectively. The pseudo code below implements the selection of the optimal default gateway:

```
1   if (h0 == h1)
2     if (L0 < L1)
3       default_gw = GW0
4     else
5       default_gw = GW1
6   else if abs(h0 - h1) > MAX_HOPS
7     if (h0 < h1)
8       default_gw = GW0
9     else
10      default_gw = GW1
11  else if abs(h0 - h1) ≤ MAX_HOPS
12    if (B0-B1 > THRESHOLD) and default_gw=GW0
13      default_gw = GW1
14    if (B1-B0 > THRESHOLD) and default_gw=GW1.
15      default_gw = GW0.
```

Lines 1-5 ensure that the least congested gateway is selected as default gateway when the local node has the same hop distance to both gateways. Lines 6-10 restrict a node to select an alternative less congested gateway if this gateway is more than MAX_HOPS farther away than the nearest gateway. This restriction is necessary to minimize the excessive usage of network resources for the purpose of load balancing. Besides, a long path (in number of hops) in wireless networks implies reduced end to end bandwidth and increased packet loss probability, which in turn will reduce the throughput. Finally, lines 11-15 basically enforce selecting the gateway *GWi* with the lowest bottleneck radio load B_i as default gateway. In order to avoid frequent route flapping (also know as the ping pong effect), the selection of a new default gateway is only commenced if the bottleneck radio load of the new gateway is less than the current gateway by the THRESHOLD value. Note that the bottleneck radio load B_i is used as metric for comparison, in contrast to line 2 where the local radio load L_i is used. By using B_i instead of L_i we can prevent crossing the congested area situation to occur. For example, if $L_0 > L_1$ and $B_0 < B_1$, this means that even though *GW1* is less congested than *GW0,* selecting *GW1* as the default gateway will probably result in a lower throughput, since the bottleneck radio load of the path to *GW1* is higher than the path to *GW0*.

In the discussion above only two gateways were considered for simplicity. However this concept may be adapted to a more generic case with multiple gateways.

IV. Evaluation

A. Routing Metrics

The evaluation of the proposed load balancing scheme is performed by simulations in ns-2 [17] on a large number of randomly generated topologies. In addition to simulations with the proposed RLLB metric, each topology is also simulated with 3 other types of routing metrics (used in [15]) to facilitate comparison. These routing metrics are described below.

Table 1. Simulation parameters

Simulator	ns-2.33
Routing protocol	UM-OLSR 0.8.7
MAC/PHY-layer	IEEE 802.11
Packet size	512 Bytes
Interface Queue Size	50 Packets
Data rate (wireless)	2 Mbps
Data range	250 m
Carrier sensing range	550 m
Simulation time	300 sec
α	0.5
T_{window}	1 sec
MAX_HOPS	2
THRESHOLD	0.03

1) Shortest hop count metric (SP)

The SP metric basically selects the nearest gateway as the default gateway for inter-domain traffic. If a node has the same hop count to both gateways, i.e. h_0=h_1, where h_0 and h_1 are the hop distance to *GW0* and *GW1* respectively, then the default gateway is selected randomly for the traffic flow.

2) Simple load balancing metric (SLB),

With SLB, nodes also select the nearest gateway as their default gateway. However, if a node has the same hop count to both gateways i.e. h_0=h_1, then the least loaded gateway is selected as the default gateway. This metric is a light load balancing metric, since only a limited number of nodes are qualified to perform load balancing, i.e. the nodes that have same distance to both gateways. Furthermore, this metric may be regarded as conservative in the sense that it does not allow a node to send traffic to alternative less congested gateways that are farther away, and hence would have consumed more resources due to the additional hop length.

3) Even load metric (EL),

With the EL metric, the network load is attempted to be distributed as evenly as possible between the gateways. In contrast to the SLB metric, a node can choose to forward its traffic to a more distant and less congested gateway in order to achieve load balancing. Consequently, the EL metric usually consumes more network resources since the diversion of traffic often requires additional hops. However, the nodes that perform the rerouting of traffic are carefully selected such that the additional number of hops induced is minimized. For example, nodes that utilize one single additional hop have higher precedence to commence rerouting than nodes that utilize 2 additional hops. Similarly, nodes that utilize 2 additional hops have higher precedence than nodes that utilize 3 additional hops, and so on.

B. Simulation Results

For the evaluation we generated 30 asymmetric random topologies. Each topology is generated by randomly deploying 30, 15 and 5 nodes in section A, B and C of the network model shown in Fig. 1. The topologies are generated in this way to ensure a certain level of asymmetry in terms of node distribution relative to the 2 gateways, i.e. more nodes are by default associated with *GW0* than *GW1*. This is important in order to test the performance of the load balancing scheme, since without asymmetry load balancing is not necessary.

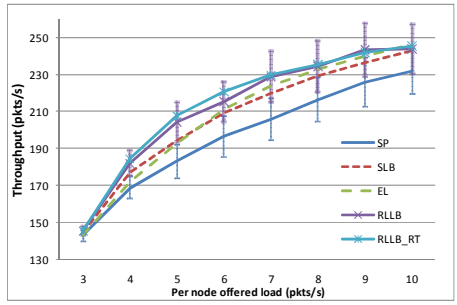

Fig. 5. Simulation result of 30 randomly generated topologies

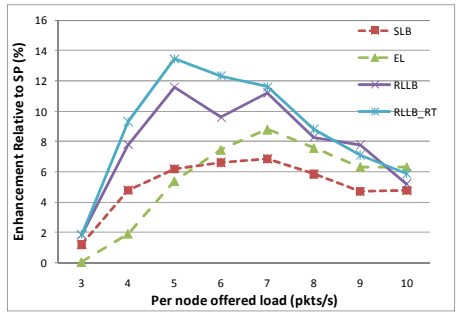

Fig. 6. Throughput enhancement in percent relative to SP

After deploying, all nodes remain at the same location, i.e. there is no mobility. Furthermore, in each simulation all 50 nodes are configured to start sending CBR traffic with packet size 512 Bytes, at t=30s. The duration of the simulation is 300 seconds, but only the result of the last 250 seconds is taken into account. Table 1 summarizes the simulation parameters.

The result in Fig. 5 shows that RLLB has a considerably better performance than SP in terms of average aggregated throughput (95% confidence interval). RLLB improves the throughput with up to 11.6% as shown in Fig. 6. Furthermore, RLLB has also higher performance than both the SLB and EL metrics, i.e. around 5 % enhancement at packet rate 5 pkts/s. The SLB metric is as previously described, a light load balancing scheme, which in many cases, is incapable to reduce the load imbalance sufficiently. Contrarily, the EL metric might be too aggressive in reducing the load imbalance, and consuming too much network resources due to longer paths. These reasons explain why SLB has higher performance than EL for rate 3-5 pkts/s, while EL has a higher performance than SLB for rate 6-10 pkts/s, as shown in Fig. 5 and 6.

The RLLB metric, on the other hand, is not constrained by the limitations of the SLB metric, i.e. only nodes that have the same distance to both gateways are allowed to reroute traffic. Neither does RLLB have to strive for zero load imbalance at any cost, as in the case of EL. The RLLB metric is designed to commence load balancing only when it is appropriate, i.e. using less congested alternative paths. One may say that RLLB lies in between the SLB and EL metrics in terms of how aggressive the load balancing is performed, and this explains why RLLB has higher performance than both SLB and EL.

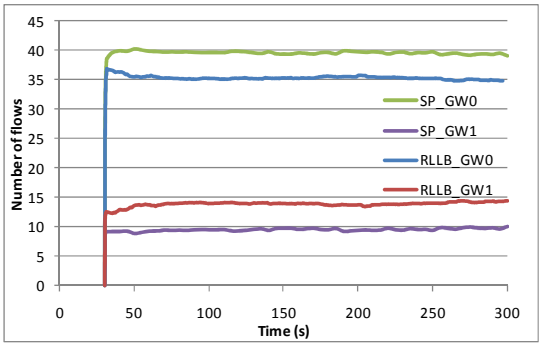

Fig. 7. Variations of the average flow distribution between *GW0* and *GW1* for RLLB at 8 pkts/s in per node offered load

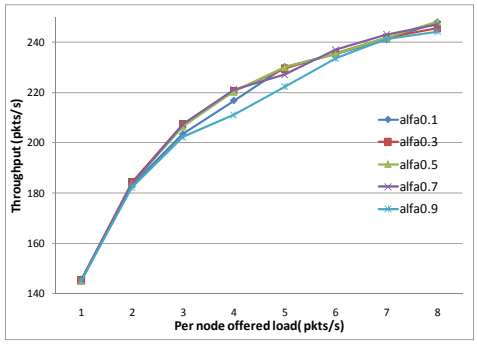

Fig. 8. Impact of varying α on the performance of RLLB

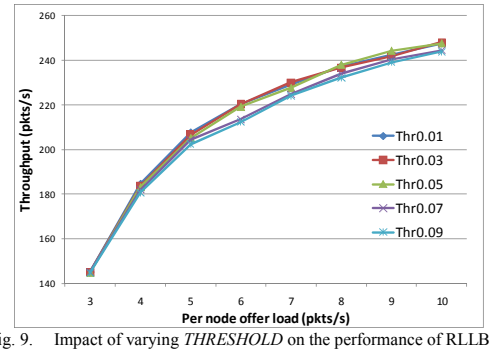

Fig. 9. Impact of varying *THRESHOLD* on the performance of RLLB

Furthermore, to evaluate how the unreliability in the dissemination of TC messages (and radio load) affects the performance of the RLLB metric, we also conducted simulations where real time radio load is used instead of TC based radio load. The result is denoted as RLLB_RT in Fig. 5 and 6, which shows that with real time radio load, the performance of RLLB_RT is slightly higher than RLLB. However, the difference is lower than we expected. We believe this is due to the fact that in static networks where the dynamic in terms of traffic distribution is low, the variations in the measured radio load over time are also low. Hence, using the periodically disseminated radio load information or using the real time radio load does not affect the performance significantly.

Fig. 7 shows how the average distribution of traffic flows between *GW0* and *GW1* (at 8 pkts in per node offered load) varies with time for the SP and RLLB metrics. From the figure, we firstly see that with SP, the load imbalance is high from the moment traffic is initiated (t=30s) till the end of the simulation (t=300s). The average traffic flow distribution between *GW0 and GW1* is approximately 39.72/9.61 in the time interval between t=50s to t=300s. On the other hand, with RLLB, the load imbalance is gradually reduced during the transient period from t=30s to t=50s. In the time interval between t=50s to t=300s, the traffic flow distribution for RLLB is approximately 35.39/13.94, i.e. lower load imbalance compared to SP. Note that in both of the above cases, the average total number of flows (49.33) is slightly lower than the number of sender nodes (50), due to disconnected nodes in certain topologies.

Secondly, when performing load balancing in static topologies with CBR traffic, one would in theory expect that after the initial transient period, route flapping should not occur. However, as shown in Fig. 7, there are small variations in the average flow distribution between *GW0* and *GW1* in the interval from t=50s to t=300s. The average number of rerouting in this interval is observed to be 10.6 for RLLB and 26 for SP. This is mainly due to the unreliability in the flooding mechanism of the routing protocol when the load is high, resulting in loss of control traffic and link breaks. Consequently, the affected nodes are forced to temporarily reroute their traffic until the broken links are restored. Furthermore, the unreliability in the flooding mechanism will become more severe with increasing load, resulting in even higher level of route flapping. This reason further explains why there is a higher level of route flapping with SP compared to RLLB, since with SP, the congestion in the network is higher due to higher load imbalance.

Additionally, we believe that the inherent uncertainty in the radio load estimation may also have a certain effect on the route flapping, i.e. the load balancing metric may mistakenly reroute traffic due to inaccurate radio load information.

C. *Impact of α and THRESHOLD*

By default the weighting factor α in (1) is 0.5. In order to investigate the impact of α on the performance of RLLB, we also conducted simulations with various values of α. Furthermore, real time radio load is used for all simulations in this subsection. The result in Fig. 8 shows that for α between 0.3 and 0.7 the performance of RLLB is virtually the same.

For the case when α=0.1, the performance of RLLB is slightly lower. This is probably due to the reason that the estimated radio load is more sensitive to changes in load distribution in the network. Consequently, there will be a higher level of fluctuations in the estimated radio load which again will result in an increased level of route flapping.

Oppositely, for high values of α such as α=0.9, the calculated radio load is less sensitive to changes in load

distribution, meaning that the radio load is too slow in capturing the changes. This again will adversely affect the RLLB metric, preventing it from performing load balancing in a timely manner. The result shows that using a too high value of α is more unfortunate than using a too low value.

The purpose of the THRESHOLD value is to control the level of route flapping. A low THRESHOLD will allow the routing protocol to freely perform rerouting when it is appropriate, but with the potential risk of a high degree of route flapping. On the other hand a high THRESHOLD, will reduce route flapping, but instead may prevent the routing protocol from performing the necessary load balancing. Hence, it is important to make a compromise when setting the value for the THRESHOLD. Fig. 9 shows the impact of the THRESHOLD on the performance. For THRESHOLDs between 0.01 and 0.05 the average throughput is approximately the same. For higher values of THRESHOLD (e.g., 0.07 and 0.09), the performance is lower due to the reason explained above.

V. CONCLUSIONS

In future tactical networks, the gateway nodes have the important role of connecting different network domains or platforms together, forming a consolidated network. We argue that there should be more than one gateway between two network domains in order to increase capacity and resiliency. However, in order to take full advantage of the increased capacity for inter-domain traffic, the routing protocol must be able to intelligently perform load balancing.

A common belief is that the gateways are the bottlenecks or most congested nodes in the network since all upstream or downstream traffic have to go through these nodes. However, we have through the work in this paper shown that this is not necessarily so. In fact for a network deployed with two gateways, the most congested area is actually located in the centre of the network, i.e. the area between the gateways. This explains why the efficiency of load balancing in many cases are very poor, since the congested area may act as an obstacle or barrier, preventing load to be effectively diverted to the alternative gateway with lower load.

Realizing this, we developed the RLLB load balancing scheme, which utilizes radio load information to make load balancing decisions. The RLLB metric is designed to improve the load distribution between gateways. Secondly the metric attempts to avoid routing traffic through the congested area such that the highest total network throughput can be achieved.

We have performed simulation on many randomly generated topologies using different load balancing metrics. The simulation results show that RLLB is more efficient than both SLB and EL in performing load balancing, i.e. approximately 5 % in throughput enhancement. Furthermore the throughput enhancement relative to the SP metric is on the average up to 11.6 %.

Finally, even though the study in this paper, for simplicity, is limited to the case with only two gateways, we believe the concept of the proposed load balancing scheme can also be adapted to scenarios with multiple gateways. We intend in future works to further investigate load balancing under more dynamic conditions, i.e. with mobile topologies and time-varying traffic load.

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Paper E :

A Radio Load Based Load Balancing Scheme with Admission Control

V. Pham, E. Larsen, Q. Le-Trung, P. Engelstad, Ø. Kure

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A Radio Load Based Load Balancing Scheme with Admission Control

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***Abstract*—In mobile ad hoc networks (MANETs) where there exist multiple Internet gateways, performing load balancing among available gateways is vital in order to take full advantage of the network capacity and improve the performance. However, performing load balancing in a wireless environment is very challenging due to the inherently interfering and unreliable nature that characterize wireless communication. Additionally, the dynamic in ad hoc networks due to node mobility makes it even more difficult. This paper explores the feasibility of performing load balancing in ad hoc networks both in static and mobile topologies, and proposes the RLAC scheme, that jointly performs the task of load balancing and admission control. RLAC relies on radio load information provided by the underlying MAC-layer to make routing decisions. Simulations with RLAC on a considerable number of random topologies, both static and mobile, show that the proposed scheme has a potential to improve performance with respect to the aggregated throughput.**

I. Introduction

In mobile ad hoc networks (MANETs), a gateway node is used to provide connectivity to external networks such as the global Internet. Since all traffic to and from the Internet has to traverse the gateway, it is therefore likely that this node will be congested and become a bottleneck, affecting the performance of the network. To alleviate this, the common solution is to deploy multiple gateways in the network. This does not only reduce the risk for congestion, but has also a number of advantages. First of all, the overall bandwidth capacity for Internet traffic is increased, both in terms of ingress and egress traffic. Secondly, multiple gateways will make the network more robust, i.e. if one gateway encounters failure, there are other gateways that can still provide connectivity to the global Internet. Thirdly, if multiple gateways are spread around in the network, the Internet traffic will also be naturally more distributed throughout the network even without any load balancing functionality.

With the traditional shortest path routing, a node will always send Internet traffic to the nearest gateway regardless of how congested this gateway is. Consequently, this will cause some gateways to be more congested than others, resulting in non optimal utilization of the network capacity. This shortcoming of the shortest path routing protocol therefore encourages the use of routing protocols with load balancing (LB) functionality, capable to control the load balance of traffic between the gateways more efficiently. However, performing LB in ad hoc networks is a very challenging task due to the inherently interfering and unreliable nature of wireless networks. The frequent changes in the topology due to node mobility will make it even more difficult, if not impossible, if the level of mobility is too high. We have previously shown in [1,2] that performing LB on static topologies may improve network performance. The question is, will LB give the same result in mobile topologies? Thus, this paper explores to which extent it is possible to improve network performance in static as well as mobile topologies with different levels of mobility.

In this paper, we propose the RLAC scheme that jointly performs LB and admission control (AC). We argue that AC is an important and necessary mechanism that will further improve the performance in addition to the benefits provided by LB. When the network load is high, AC can be used to reduce excess traffic from being inserted into the network and thus avoid even higher congestion and augmentation in packet loss rate. Furthermore, in cases where the congestion is very high along the path from a source to a destination node, the probability for successful transmissions of data traffic along this path is consequently very low. In such cases, AC can be used to deny traffic, that most likely would not reach its final destination anyway, from entering the network.

The rest of this paper is organized as follows. In Section II, we discuss related work and the background for this study. Section III describes the proposed load balancing scheme in detail. Section IV presents the evaluation of RLAC. Finally, the conclusion of the paper is given in Section V.

II. Related Work and Background

In the literature, there exists a number of proposals suggesting various LB scheme targeted for both MANETs [3,4,5,6] and Wireless Mesh Networks [7,8,9,10]. These proposals can be generally divided into two categories: *Multipath Load Balancing (MLB)* and *Gateway Load Balancing (GLB)*.

In *MLB* [3,4,5], the traffic load between a source node and a destination/gateway is distributed among a set of alternative paths in order to maximize performance and minimize the impact of route failure. However, [3,4] report that *MLB* in single channel wireless networks only provides a negligible improvement in the performance due to route coupling among the alternative paths. Thus *MLB* is not of interest in this paper.

On the other hand, in the *GLB* [1,2,6,7,8,9] approach, the traffic load is distributed among multiple gateways in order to maximize throughput performance and to reduce the load imbalance. *GLB* is considered to improve the network performance more effectively than *MLB* [7]. In this paper, we therefore focus only on *GLB*, and we will refer to it simply as LB in the remainder of the paper.

To perform LB, a variety of techniques is used for evaluating the network load, including RTT [10], average queue length [9,11,12] and number of active flows [13,14]. However, in our previous work [2], we showed that radio load information (RL) provided by the underlying MAC layer may also be used to perform load balancing. The advantage of using radio load is that active probing is not necessary like in other techniques such as RTT, and thereby avoid more overhead.

A number of important aspects related to performing LB in ad hoc networks, is also showed in [2]. Firstly, since all Internet traffic has to traverse through the gateways, it is intuitive to expect that the areas around the gateways are the most congested. However, using the *congestion map,* it is shown that the area near the centre of the network is actually the most congested. This is due to the fact that the centre area is within the sensing range of the majority of nodes in the network.

Secondly, the congested centre area explains why it is difficult, in many cases, to perform load balancing, since the congested area represents an obstacle or barrier, preventing traffic load to be diverted from a congested gateway to another less congested gateway. In such cases, redirecting traffic incurs that the traffic has to cross even more congested bottleneck nodes in the congested area, which would likely result in worse performance. The same reason also explains why the performance of LB is high for certain topologies while it is very poor for others.

To accommodate the issues above, the proposed Radio Load based Load Balancing (RLLB) scheme in [2] was therefore designed to incorporate radio load information in the process of performing LB. Based on the radio load, the scheme ensures that redirection of traffic is only commenced in appropriate situations, i.e. only in situations when the traffic does not have to traverse congested bottleneck nodes or areas. Simulations results show that the RLLB scheme performs well for static topologies with high level of asymmetry in terms of node distribution. However, the performance is not so optimal for topologies with lower level of asymmetry due to the problem of *synchronized rerouting*. We will explain this problem in more detail in the following section. Furthermore, LB in mobile topologies was not considered in the study.

The work in this paper is a continuation of our previous work, in which some missing parts are addressed, i.e. the issue of synchronized rerouting, the issue of LB in mobile topologies and the issue of admission control. The aim is to develop a more efficient and generic LB scheme for both static/mobile topologies and for topologies with varying level of asymmetry. We call this new scheme for RLAC, which jointly performs LB and AC. We believe that by combining these two mechanisms, it is possible to further improve the performance.

III. Proposed Load Balancing Scheme

The task of the LB mechanism in RLAC is to reduce load imbalance between the gateways and increase the aggregated capacity in terms of throughput for upstream traffic. In order to fulfil this task, each node in the network must carefully select its default gateway such that the load imbalance between gateways is as low as possible. Additionally, RLAC also has an AC mechanism to prevent excessive traffic from entering the network. In this section we will discuss the challenge related to performing LB in distributed systems, and then present the architecture and the main components of the RLAC scheme.

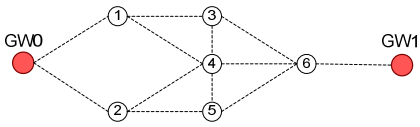

Figure 1. Scenario of synchronized rerouting

A. Synchronized Rerouting

In distributed systems where each node individually and independently selects its default gateway, the problem of *synchronized rerouting* may occur. This may be explained using the scenario in Figure 1. Suppose all nodes originate traffic of equal rate to their default gateway. Thus, node 1 and 2 originate traffic to *GW0*, since this is the nearest gateway. Likewise, node 6 selects *GW1* as the default gateway upon which traffic is originated to. At this moment *GW0* is therefore more loaded than *GW1*. Next, node 3, 4 and 5 have now upstream traffic to send. Suppose these nodes are aware that *GW0* is more congested than *GW1*, they will therefore choose to send traffic to *GW1*. This will eventually result in that *GW1* is more congested than *GW0*. When node 3, 4 and 5 are aware of this condition, they will try to reduce the load imbalance between the two gateways by rerouting their traffic to *GW0*, which will again cause *GW0* to be more congested than *GW1*. We refer to this situation of ping pong effect as *synchronized rerouting* where a group of nodes simultaneously and repeatedly reroute their traffic from one gateway to another.

B. Overview of RLAC Architecture

Figure 2 shows an illustration of the RLAC architecture. RLAC is a cross-layer based scheme intended for proactive routing protocols such as OLSR [15]. One of the fundamental building block of the RLAC scheme, is the radio load provided by the underlying MAC-layer. This information is used by the routing protocol to perform the tasks of: *i)* LB (by determining the optimal default gateway), and *ii)* AC (on nodes with high level of contention).

During the calculation of the routing table (RT), the bottleneck radio load (BN RL) to each destination is also calculated using the Dijkstra's algorithm. Once this is completed, the scheme determines whether to enable/disable AC on the local node. If AC is not enabled, the function *select_def_gw* is called where the process of default gateway selection is performed. On the other hand if AC is enabled, *reset_def_gw* is called to reset the default gateway.

The flow chart in Figure 3 shows how packet forwarding is handled by RLAC. If AC is enabled at the local node, and if

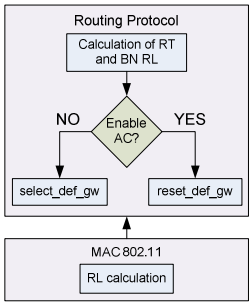

Figure 2. RLAC Architecture

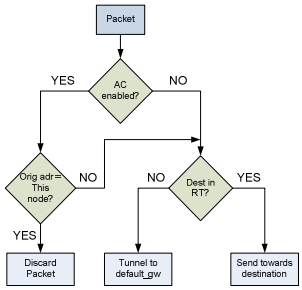

Figure 3. Flow chart of the internal packet flow

the local node is the originator of the packet, the packet is discarded in order to prevent new packets from entering the network. Otherwise, the local node may still forward transit packets to its final destination, even if AC is enabled. If AC is disabled, packets originated by the local node may either be tunneled towards the default gateway if the destination address belongs to the global Internet (i.e. the destination address is regarded as an external address if it does not exist in the RT), or forwarded to the destination node if the destination address is a local address.

C. Radio Load

The radio load is defined as the amount of time T_{busy} within a time window T_{window} where the local channel is monitored as busy. To estimate the average radio load L we use the exponential moving average as follows:

$$L_{new} = \alpha \cdot L_{previous} + (1 - \alpha)\frac{T_{busy}}{T_{window}} \quad (1)$$

where α is the weighting factor defined as $\alpha \in [0,1]$. A more detailed description of the radio load calculation and the dissemination process can be found in [2].

D. Metric for Gateway Selection

For the discussion of the gateway selection metric we use the model shown in Figure 4. Node n is located h_0 and h_1 hops from *GW0* and *GW1*. Furthermore, let L_0 and L_1 be the gateway radio load at *GW0* and *GW1*, and B_0 and B_1 be the bottleneck radio load of *GW0* and *GW1*. B_i is defined as the

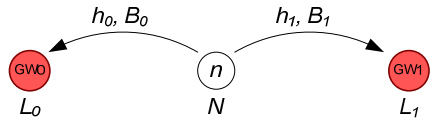

Figure 4. Model for gateway load balancing

highest observed radio load along the path from the local node n to *GWi*, excluding the radio load N of the local node and including L_i.

Let us define the probability P_0 for the local node to select *GW0* as default gateway as:

$$P_0 = \begin{cases} 0.5 - a \cdot \Delta B - b \cdot \Delta L - c \cdot \Delta h \\ 0 \quad , \quad P_0 < 0 \\ 1 \quad , \quad P_0 > 1 \end{cases} \quad (2)$$

where a, b and c are constants and

$$\Delta B = B_0 - B_1 \quad (3)$$
$$\Delta L = L_0 - L_1 \quad (4)$$
$$\Delta h = h_0 - h_1 \quad (5)$$

The probability for selecting *GW1* as default gateway is simply:

$$P_1 = 1 - P_0 \quad (6)$$

Thus the probability for selecting *GW0* as default gateway is a function of the bottleneck and gateway radio load as well as the hop count to each gateway. We argue that in order to determine the most optimal gateway in which traffic is forwarded to, it is necessary to take these parameters into account, since they all have a direct impact on the performance, in one way or another. While L_i determines how much traffic the gateway *GWi* can receive, the bottleneck B_i and the hop distance h_i is a measure for much traffic that can be transported along the path to *GWi*. In certain cases, e.g. when $B_0 > B_1$, $L_0 > L_1$ and $h_0 > h_1$, it is obvious that *GW1* should be selected as default gateway, and P_0 and P_1 should be 0 and 1, respectively. On the other hand, if $L_0 > L_1$, but $B_0 < B_1$, while $h_0 = h_1$, then is more ambiguous regarding which gateway that is best to select as default gateway. The purpose of the weighting constants a, b, and c is to make a compromise among these parameters, but finding the optimal values for these constants is not an easy task. However, the results from previous work [2] give us the following guidelines:

1. If $L_0 > L_1$, but on the other hand, $B_1 > B_0$, then forwarding traffic to *GW1* would probably result in poorer performance. This suggests that a should be given more weight than b ($a > b$).
2. Given that $L_0 > L_1$, it may still be beneficial to reroute traffic to *GW1* even though $h_1 > h_0$ by only a few hops. This suggests that b should be given more weight than c ($b > c$). Note that c serves both as a weighting and scaling constant, since Δh is a measure of the difference in hop count (integers), while ΔB and ΔL are the difference in radio load ranging from 0.0 to 1.0.

TABLE 1. SUMMARY OF VARIABLES

Variable name	Description	Default
a	Constant	10
b	Constant	1
c	Constant	0.4
AC_UPPER	Threshold for enabling AC	0.96
AC_LOWER	Threshold for disabling AC	0.92
AC_MAX_HOPS	The hop count threshold for enabling/disabling AC	2
N	Radio load of the local node	
Packet Size		512 Bytes
Queue Size	Size of the interface queue	50 pkts
Data Rate	Max data rate of wireless interface	2 Mbps
Data range	Max data transmission range	250 m

Based on these properties and in addition, using empirical data from many simulations, we found that setting $a=10$, $b=1$ and $c=0.4$ can provide good results with respect to performance. These values are not necessary the most optimal, rather using these values, we demonstrate that it is feasible to improve performance by using the proposed scheme.

After having calculated P_0 and P_1, a uniformly distributed random number $R \in [0.0, 1.0]$ is generated. If $R < P_0$, then *GW0* is selected as default gateway, otherwise if $R > P_0$, then *GW1* is selected as default gateway. By applying this probability based gateway selection approach, nodes 3, 4, and 5 from the scenario in Figure 1 will initially have a higher P_1 to select *GW1* as default gateway, but at the same time, there will still be a small probability P_0 that some of these nodes will select *GW0* as default gateway. Thus, this approach will ensure that on the average, the majority of nodes (that perform rerouting) will select the least congested gateway as default gateway, while a minority will select the more congested gateway as default gateway, and thereby preventing the problem of synchronized rerouting to occur.

E. Admission Control

Besides of performing LB, RLAC also performs AC. The task of the AC is to prevent the load in the network from reaching a critical high level. This is especially important in wireless networks due to the interfering nature of the shared medium, where packet transmissions may be more vulnerable to collisions and loss when the network is overloaded. In the worst case, severe link breaks may occur or the network may even cease to exist due to heavy loss of control traffic. To prevent this we have implemented a simple AC scheme that works in the following way: If B_0 and B_1 are higher than the threshold *AC_UPPER*, then AC is enabled on *n*, if *n* is located more than *AC_MAX_HOPS* away from the nearest gateway. This implies that if the network load is high, AC is enabled on nodes that are located farther away from the gateways, in order to give priority to nodes closer to the gateways. This design choice may be justified by the fact that the cost of transmitting traffic destined to the gateways is higher for nodes located farther away than for nodes in the proximity of the gateways. Furthermore, when the network load is high, nodes that are located farther away from the gateways will most likely experience a very low packet delivery ratio. Enabling AC on these nodes is reasonable, since it does not matter whether no or only a few packets can reach the destination. Once AC is enabled, only traffic originated by the local node is discarded, while transit traffic (i.e. traffic not originated by the local node) may still be forwarded.

If the network load is monitored to decrease such that B_0 or B_1 are lower than the threshold *AC_LOWER*, AC is disabled on *n*. Moreover, if node *n* has moved its location such that either h_0 or h_1 is equal to or lower than *AC_MAX_HOPS*, then AC is also disabled.

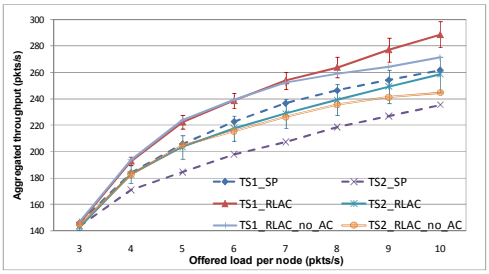

Figure 5. Average throughput of 30 static topologies

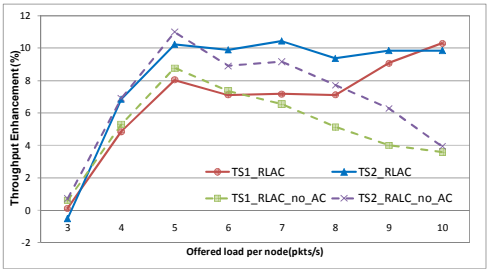

Figure 6. Throughput enhancement of RLAC (with and without AC) relative to SP.

IV. EVALUATION

In order to evaluate the performance of the proposed scheme, we implemented and integrated the RLAC scheme with UM-OLSR [16] and conducted a large number of simulations in ns-2 [17]. A summary of the parameters used in the implementation and the simulations is listed in Table 1.

Simulations are performed on a large number of randomly generated topologies, both static and mobile. The general topology is confined in a rectangular area of 1400 m × 800 m. Two stationary gateways, *GW0* and *GW1*, are deployed at location (200,400) and (1000,400), respectively. In addition, 50 nodes are randomly deployed. All 50 nodes originate CBR traffic with packet size 512 bytes, which are tunneled to either one of the gateways.

A. Static Topologies

For static topologies, the simulations are conducted on two topology sets with different levels of asymmetry in terms of node distribution. The generation of each topology set is conducted in the following way:

The rectangular that confines the simulation area is divided into three vertical sections A, B, and C. The intermediate

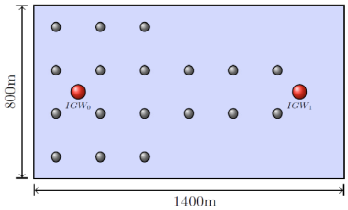

Figure 7. Mobile topology

border lines that separate the 3 sections are located at x=500 and x=900. For topology set I (TS1), 20, 20 and 10 nodes are randomly deployed in sections A, B and C, respectively. By deploying the 50 nodes in this fashion, it is ensured that the topologies are asymmetric in terms of node distribution (and traffic load), i.e. more nodes are located closer to *GW0* than to *GW1*. The asymmetry is necessary for the evaluation of the proposed scheme, since without it, there is no need to perform LB.

Similarly, for topology set II (TS2), 30, 15 and 5 nodes are randomly deployed in sections A, B, and C, respectively. Thus the level of asymmetry is even higher in TS2 than in TS1.
Each topology set consists of 30 randomly generated topologies. The simulation time for each topology is 300 s, where only the last 250 s is taken into account in the result.

Figure 5 shows the simulation results for both topology sets, where the y-axis represents the average aggregated throughput at the gateways in packets/s. The offered load per node is given in the x-axis. The result shows that as the offered load is higher than 3 pkts/s, the RLAC metric consistently has higher performance than the traditional shortest path metric (SP), for both TS1 and TS2. The same figure also shows that RLAC without the AC mechanism enabled (RLAC_no_AC) has lower performance than in the case where it is enabled.

The throughput enhancement of the RLAC metric relative to the SP metric is shown in Figure 6. The result shows that the improvement in throughput is approximately up to 8 and 10 percent for TS1 and TS2 respectively. This result is consistent with the result in [1], where it is showed that the enhancement in throughput provided by LB increases with increasing level of asymmetry. Furthermore, the figure also shows that without AC, the performance of RLAC gradually decreases for packet rates above 5 pkts/s. On the other hand, with AC, RLAC may still maintain approximately the same level of throughput enhancement even when the offered load is increased above 5 pkts/s.

B. Mobile Topologies

We also conducted simulations on mobile topologies in order to explore the feasibility of performing LB during mobility. Each topology is generated by deploying 18 mobile nodes at the initial locations as shown in Figure 7. The remaining 32 nodes are randomly deployed within the simulation area. This approach is used in order to generate mobile topologies that initially are asymmetric and unpartitioned. However, during the course of the simulation, node mobility may result in partitioning and change in the

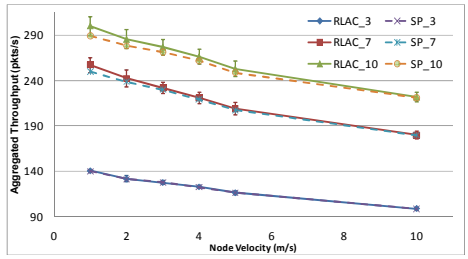

Figure 8. Aggregated throughput for mobile topologies with varying level of mobility (node velocity)

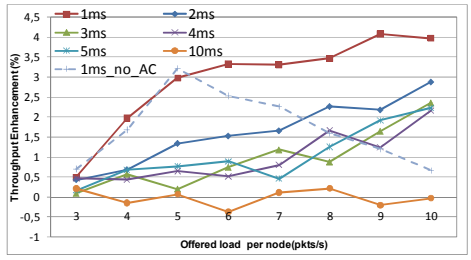

Figure 9. Throughput enhancement for mobile topologies

level of asymmetry. Furthermore, we used the random walk with reflection mobility model [18] where each node is basically moving with constant velocity v and direction for a duration of 20±5 s in travel time, before the direction is changed. The duration of each simulation is 600 s, and the result sampling is performed for the last 550 s.

Figure 8 shows the average aggregate throughput for 20 mobile topologies using the RLAC metric and SP metric, and plotted as function of node velocity. In the figure, only the results for packet rate 3, 7 and 10 pkts/s are shown. As expected, there is a close correlation between the level of mobility and the aggregated throughput, i.e. the higher the level of mobility is, the lower is the aggregated throughput.

Figure 9 shows the average throughput enhancement of the RLAC metric relative of the SP metric, and as can be seen, the level of mobility has a substantial impact on the beneficial of performing LB. The enhancement of RLAC is highest at 1 m/s and decreases as the node velocity is increased. At 10 m/s, the performance of RLAC is virtually equal to the SP metric. This result indicates that performing LB in mobile topologies is extremely challenging, especially when the mobility level is high. Even when the mobility level is low, e.g. 1 m/s, the advantage of performing LB is moderate, i.e. the enhancement in throughput is at best approximately 4 %. Comparing this result with the results in Figure 5 and 6, we see that the throughput enhancement at 1 m/s is less than 50 percent of what that can be achieved in static topologies.

Figure 9 also shows the performance enhancement of RLAC without AC enabled for the case with velocity equal to 1 m/s. The result shows the same tendency as in the case with static topologies, i.e. when the offered load is higher than 5 pkts/s, the contribution from the LB mechanism is decreasing while the contribution from the AC is increasing.

There are a number of reasons why the level of mobility has such a negative impact on the throughput as well as the beneficial of performing LB. First of all, link breaks occur more frequently with increasing level of mobility. This implies that the link life time will decrease with increasing mobility level, and consequently, the effective network capacity will become lower. Secondly, LB in many cases involves diverting traffic to a more distant gateway. This will unfortunately increase the probability for end to end transmission failure or packet loss due to the longer path distance that a packet must travel. Thirdly, in contrast to static topologies, partitioning is likely to occur in mobile topologies during the course of the simulation time. The probability for partitioning in the network increases as the mobility level is increased. When partitioning occurs such that the gateways are no longer connected, then performing LB is not feasible. Furthermore, with mobile topologies, it is difficult to maintain the same level of asymmetry as in the beginning (i.e. the initially induced asymmetry during generation of the topologies) for the entire duration of the simulation time. We observed that during the course of the simulation, the topologies either tend to be so asymmetric such that the network will eventually be partitioned, or oppositely the asymmetry is decreased to a minimum level. Either way, the condition for testing the performance of LB is non optimal.

Finally, the inherent latency of the routing protocol with respect to capturing the changes in the topology is another reason why the performance of LB and AC is low during mobility. In fact, the LB and AC mechanisms may even have an adverse impact on the performance when the routing information does not correctly reflect the actual view of the topology.

V. Conclusion

In a wireless network where there are multiple gateways, by not utilizing an appropriate routing scheme such that traffic load is distributed more evenly between available gateways, severe load imbalance may occur. This may potentially result in network instability as well as lower performance in terms of delay, throughput, packet loss etc. The focus in this paper is to investigate whether it is possible to achieve a higher performance in terms of throughput by applying a more intelligent routing protocol. The proposed radio load based RLAC metric is developed to meet this requirement, by jointly performing the task of LB and AC. Simulation results with static topologies show that the RLAC metric may improve the throughput up to around 10 % compared to the traditional SP metric. Furthermore, RLAC is capable to enhance the throughput even for mobile topologies. However, this is true only at low node velocity, i.e. 1-2 m/s. The achieved enhancement relative to the SP metric is around 4 %. There are a number of reasons why it is much more challenging to improve the throughput in mobile topologies than in static topologies. Frequent link breaks and the risk for network partitioning are some of the reasons that set an upper limit on the achieved enhancement in throughput. Another reason is the difficulty in maintaining the appropriate condition for performing LB with respect to the level of asymmetry and avoiding partitioning.

In this study we have only focused on performing LB for upstream traffic, but we believe that the same concept may also be applicable to downstream traffic as well. In order to accomplish this, LB functionality must also be implemented at the gateways in a similar fashion as it was implemented at the nodes for the case with upstream traffic.

Finally we have for simplicity limited this study to only consider LB with two gateways in the network. However, the results and insight gained in this study may form a basis for further development of more generic LB metrics applicable for scenarios with multiple gateways.

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