**LAPPEENRANTA UNIVERSITY OF TECHNOLOGY**
DEPARTMENT OF INFORMATION TECHNOLOGY

# WIRELESS APPLICATION TEST BED AND SIMULATIONS FOR BROADBAND MOBILE COMMUNICATIONS SYSTEMS

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**ABSTRACT**

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**Wireless Application Test Bed and Simulations for Broadband Mobile Communications Systems**

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The current development in broadband wireless communications has evoked the interest to utilize it professionally for public safety. Often in case of emergency the existing communication infrastructure is not available or the bandwidth is not adequate. Therefore the need for rapidly deployable and simple to operate wireless systems with broadband capabilities is risen.

The objective of this master's thesis is to investigate wireless ad hoc multi-hop networking for public safety and to implement a test bed for demonstrational and research purposes. Point-to-point and especially point-to-multipoint ad hoc multi-hop routing are investigated. The throughput performance, transmission power, and receiver sensitivity of the test bed are measured and then exported to a simulator to ensure consistency between the test bed and simulations. Then a set of required applications and application schemes for public safety are chosen and their performance with different routing schemes is measured with the test bed and the simulations. The results of the test bed and the simulations are compared and evaluated. Multicast multi-hop video was chosen as the primary application which will also be assessed in a real field trial.

**TIIVISTELMÄ**

Lappeenrannan teknillinen yliopisto

Tietotekniikan osasto

Pekka Parviainen

**Langaton sovellustestialusta ja simulointeja laajakaistaisia mobiileja tietoliikennejärjestelmiä varten**

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107 sivua, 30 kuvaa, 33 taulukkoa ja 4 liitettä

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Langattoman laajakaistaisen tietoliikennetekniikan kehittyminen on herättänyt kiinnostuksen sen ammattimaiseen hyödyntämiseen yleisen turvallisuuden ja kriisinhallinnan tarpeisiin. Hätätilanteissa usein olemassa olevat kiinteät tietoliikennejärjestelmät eivät ole ollenkaan käytettävissä tai niiden tarjoama kapasiteetti ei ole riittävä. Tästä syystä on noussut esiin tarve nopeasti toimintakuntoon saatettaville ja itsenäisille langattomille laajakaistaisille järjestelmille.

Tässä diplomityössä on tarkoitus tutkia langattomia ad hoc monihyppy -verkkoja yleisen turvallisuuden tarpeiden pohjalta ja toteuttaa testialusta, jolla voidaan demonstroida sekä tutkia tällaisen järjestelmän toimintaa käytännössä. Työssä tutkitaan pisteestä pisteeseen sekä erityisesti pisteestä moneen pisteeseen suoritettavaa tietoliikennettä. Mittausten kohteena on testialustan tiedonsiirtonopeus, lähetysteho ja vastaanottimen herkkyys. Näitä tuloksia käytetään simulaattorin parametreina, jotta simulaattorin tulokset olisivat mahdollisimman aidot ja yhdenmukaiset testialustan kanssa. Sen jälkeen valitaan valikoima yleisen turvallisuuden vaatimusten mukaisia ohjelmia ja sovellusmalleja, joiden suorituskyky mitataan erilaisten reititysmenetelmien alaisena sekä testialustalla että simulaattorilla. Tuloksia arvioidaan ja vertaillaan. Multicast monihyppy -video päätettiin sovelluksista valita tutkimusten pääkohteeksi ja sitä sekä sen ominaisuuksia on tarkoitus myös oikeissa kenttäkokeissa.

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## SYMBOLS AND ABBREVIATIONS

### Symbols

| | |
|---|---|
| $CW_{\min}$ | minimum contention window size |
| DIFS | DCF interframe space time |
| $m$ | expected number of neighbors per node |
| $N$ | number of nodes |
| $N_{\mathrm{DBPS}}$ | number of data bits per symbol |
| $P_{\mathrm{dBm}}$ | transmission power |
| $R$ | transmission range |
| $R_{\mathrm{bit}}$ | bitrate |
| $r$ | distance |
| $S$ | size of simulation area |
| SIFS | short interframe space time |
| $\sigma$ | slot time |
| $T_{\mathrm{ACK}}$ | time to send ACK |

| | |
|---|---|
| $T_b$ | frame exchange time for IEEE802.11b |
| $T_g$ | frame exchange time for IEEE802.11g |
| $T_{PHY}$ | time of PLCP preamble and PLCP header delay |
| $T_{sigext}$ | 802.11g signal extension time |
| $T_{sym}$ | symbol time |

**Abbreviations and acronyms**

| | |
|---|---|
| ACK | acknowledgement |
| AP | access point |
| BSS | basic service set |
| C3 | command and control center |
| CIF | common intermediate format |
| COTS | commercial off-the-shelf |
| CSMA/CA | carrier sense multiple access with collision avoidance |
| DB | database |
| DCF | distributed coordination function |
| DIFS | DCF interframe space |
| DVMRP | distance vector multicast routing protocol |
| ERP | extended rate PHY |
| fps | frames per second |
| GIS | geographic information system |
| GPRS | general packet radio service |
| GPS | global positioning system |
| HUD | head-up display |
| IBSS | independent basic service set |
| IEEE | institute of electrical and electronics engineers |
| IETF | Internet engineering task force |
| IP | internet protocol |
| IR | infrared |
| JREP | join reply |
| JREQ | join request |

| | |
|---|---|
| kbps | kilo bits per second |
| LAN | local area network |
| LLC | logical link control |
| LOS | line-of-sight |
| MAC | media access control |
| MANET | mobile ad hoc network |
| MOSPF | multicast open shortest path first |
| MPDU | MAC protocol data unit |
| MPEG | moving picture experts group |
| MPR | multi-point relay |
| MSDU | MAC service data unit |
| ODMRP | on-demand multicast routing protocol |
| OFDM | orthogonal frequency-division multiplexing |
| OLSR | optimized link state routing protocol |
| OS | operating system |
| PCI | peripheral component interconnect |
| PCMCIA | personal computer memory card international association, PC card |
| PDA | personal digital assistant |
| PER | packet error rate |
| PHY | physical layer |
| PIM-DM | protocol independent multicast – dense mode |
| PLCP | physical layer convergence procedure |
| PTT | push-to-talk |
| QoS | quality of service |
| RF | radio frequency |
| RTP | real-time protocol |
| RTS | request to send |
| RTSP | real-time streaming protocol |
| SAR | search and rescue |
| SIFS | short interframe space |

| | |
|---|---|
| SMF | simplified multicast forwarding |
| SNAP | subnetwork access protocol |
| TCP | transmission control protocol |
| TETRA | terrestrial trunked radio |
| TOS | type of service |
| TU | transmission unit |
| UDP | user datagram protocol |
| USB | universal serial bus |
| WAN | wide area network |
| WiMAX | world interoperability for microwave access (IEEE 802.16 standard) |
| VoIP | voice over internet protocol |

# 1. INTRODUCTION

Public safety is a field which requires various reliable methods of communication. Different emergency and every-day scenarios involve communication from headquarter level to man-to-man level. In addition to normal internal communication there is a need to exchange information between different authorities and external actors as well [1-2]. Majority of this communication is group-oriented which should be taken into consideration. Due to characteristics of this kind of communication multicast can be applied to make it more efficient to save network resources.

Voice is unquestionably the most important application but video and other data communication forms are very beneficial as well [3]. Therefore broadband connections can offer great value for public safety. Broadband communications and especially wireless variants of it are not yet widely applied in practical and usable manner in the field since there does not exist many suitable implementations. Real user input is required to understand and develop this kind of systems. This makes studying the topic very challenging and creates demand for prototypes that can be used to demonstrate and further investigate the possibilities for public safety.

The goal of this work is to develop and implement a wireless ad hoc multi-hop network test bed for prototyping and demonstrating actual possibilities of network itself and applications in the field of public safety. The work is supported with simulations. The problem is to find and implement simple but adequately extensive system consisting of hardware and software. One important purpose is to be able to co-operate with real end users. The end users must be able to assess the system and further discover new requirements.

The solution is based on available studies, user requirement documents and interviews for current and future requirements for public safety. The gathered information is used to choose a suitable set of software and hardware for the test bed. The purpose is to develop modular and reusable system that can be managed easily also in the future when modifying or adding new features to it. Testing new software applications should be as straightforward and simple as possible.

## 1.1 Ad Hoc Networks and Multicast

Mobile wireless ad hoc networks consist of mobile nodes that are connecting with each other either directly or by using multi-hop wireless links through a number of intermediate nodes. The multi-hop links allow communication beyond the limited point to point radio propagation range. Mobile wireless ad hoc networks are infrastructureless and they do not use any centralized control. Their nodes are mobile and the network topology changes at any time. Disaster recovery, search and rescue are typical scenarios where wireless ad hoc networks could be deployed.

Group-oriented communication is an important operation for many public safety scenarios. The multicast services can be considered as an important application in mobile ad hoc networks. It is possible to avoid overhead and save network resources with well applied multicast techniques. This type of services include different types of communication such as data dissemination, teleconference (VoIP), video conference and surveillance. The multi-hop and dynamic network topology make routing and multicasting extremely challenging. One main purpose in this thesis is to study the performance of wireless ad hoc networks with multicast traffic. Multicasting is investigated with real-time test bed experiments and with simulations. For real-time experiments a test bed of four nodes is set up. For simulations QualNet 3.9 network simulator is used [4]. To compare the throughput results of multicast traffic with test bed and QualNet their PHY and MAC layer parameter and settings should be matched to each other.

## 1.2 Studying the Performance and the Parameters of Test Bed

Studying the performance of the test bed and multicast routing is divided into two main parts. In the first part, the throughput performance the of test bed itself is measured and the PHY and MAC layer parameters are tried to be identified. Usually, manufacturers of WLAN cards do not provide enough information in their technical specification about these parameters. The parameters need to be studied and verified experimentally. After getting the PHY and MAC layer parameters of the test bed, the QualNet throughput performance is tested using these parameters. If the throughput measured by test bed and obtained by QualNet with the same PHY and MAC layer parameters is close enough, then we can assume that the simulator provides reliable results. Since the test bed comprises of four nodes only, the performance of multicast in a large-scale scenario should be investigated through simulation work. Provided that the throughput performance of test bed and simulator are close enough in a small scale scenario, the behavior of the real system in large-scale scenarios is tested via simulations.

## 1.3 Studying the Performance and the Behavior of Multicast Routing Protocols

Multicast routing in multi-hop wireless ad hoc networks is investigated in the second part. The experimental work utilizing Simplified Multicast Forwarding (SMF) protocol running in co-operation with Optimized Link State Routing (OLSR) protocol is carried out with the test bed [5]. OLSR is a pro-active unicast routing protocol [6]. On Demand Multicast Routing Protocol (ODMRP) system performance study is done with QualNet [7-8]. ODMRP is reactive routing protocol and can be run stand-alone without supporting unicast protocol. ODMRP is able to do unicast routing as well. It is a typical representative of mesh-based routing protocols using the concept of forwarding-group.

The degradation of multicast performance due to hidden node problem is evaluated under VoIP and video traffic. The results of test bed studies and

QualNet simulations are compared. The performance of ODMRP is tested by using some classical metrics in large-scale scenarios. The effectiveness and efficiency of ODMRP with network mobility, size of multicast group and varying network traffic load are studied through extensive simulation work. The performance of a typical mesh based scheme is studied under different network conditions and the possibility of multicasting video in a large-scale scenario is examined.

# 2. TEST BED AND NETWORK SIMULATOR

## 2.1 Hardware for the Wireless Ad Hoc Network

Choosing the main bearer for the ad hoc network was quite straightforward. Only currently widely available, used and standardized technique is IEEE 802.11 wireless LAN. Another possibility could be IEEE 802.16 WiMAX in the near future. The test bed comprises of WLAN capable equipment e.g. table and laptop PCs, mobile phones and cameras. PCs with their PCI and PCMCIA WLAN cards form the core of the network. These commercially available devices enable the IEEE 802.11a/b/g data rates from 1 to 54 Mbps which fulfills the broadband requirements completely. The MAC layer protocol utilized in WLAN is above mentioned 802.11 with carrier sense multiple access / collision avoidance (CSMA/CA). IPv4 was chosen as the network layer protocol above the hardware-integrated physical and MAC layer protocols. It is possible to utilize IPv6, since it is supported by the chosen hardware and the software. IPv4 was chosen for now to ensure maximum interoperability and comparability.

## 2.2 Description of Complementary Hardware Used in the Test Bed

In addition to computers and network adapters the test bed consists of various devices serving certain purposes. The complementary devices are presented in Table 1. Universal serial bus (USB) web cameras provide video, still images and also voice if that is required. There is also one stand-alone WLAN camera which is capable to operate in basic ad hoc or structured mode without advanced routing. Services of the camera are offered through integrated web server. Microphones and headsets including a microphone and headphones are used for voice communication.

For tracking of the nodes, which is very important application for public safety [9], global positioning system (GPS) devices connecting through Bluetooth radio, USB or serial interface are used. Some of the computers do not have integrated Bluetooth adapters and external USB Bluetooth dongles are used instead.

It is particularly beneficial to provide network connections to headquarters and other external resources in addition to LAN connections [10]. These wide area network (WAN) connections can be provided by utilizing the existing infrastructure e.g. The Internet or other dedicated communication means. To provide gateway functionalities between LAN and WAN in the system, various additional network interfaces can be utilized. Secondary WLAN adapter, Ethernet adapter, or general packet radio service (GPRS) connection through USB modem or mobile phone can be currently employed in the test bed.

**Table 1:** Test Bed Hardware

| Device | Description |
|---|---|
| Logitech QuickCam Messenger | USB camera and microphone (2 pcs) |
| SilverCrest Webkamera 1,3 MPix - 12021 | USB camera and microphone |
| Labtec webcam pro | USB camera and microphone |
| Grandtec Grand WiFi camera | WLAN / IP / USB (stand-alone) camera |
| Logitech Dialog 320 | desktop microphone |
| Linksys Wireless-G Notebook Adapter WPC54G | PCMCIA WLAN network adapter (2 pcs) |
| ORiNOCO 11b/g PC Card Gold 8470-WD | PCMCIA WLAN network adapter |
| WLAN 802.11(a/b/g) network adapters | integrated in part of the laptops |
| Nokia Wireless GPS Module LD-1W | Bluetooth GPS receiver (2 pcs) |
| Telewell GPRS modem | USB modem |
| Belkin Bluetooth USB Adapter | (2 pcs) |
| HP Laptop computer | (2 pcs) |
| Fujitsu Siemens laptop computer | (2 pcs) |

## 2.3 Test bed Software

### 2.3.1 Operating Systems

Both Microsoft Windows and Linux operating systems are used for running the test bed. Certain hardware and software applications require either of the environments to run properly or to make it possible to adjust important properties of them. Naturally, it is generally beneficial to support multiple environments for better compatibility. Windows XP Professional version 2002 SP2, Windows XP Home Edition version 2002 SP2, and Ubuntu 6.06 LTS Linux are the employed versions.

### 2.3.2 VIC (Video Conferencing Tool), Real-Time Multimedia Application for Video Conferencing over Various IP Networks

The UCB/LBNL video tool, VIC, was designed with a flexible and extensible architecture to support heterogeneous environments and configurations. For example, in high bandwidth settings, multi-megabit full-motion JPEG streams can be sourced using hardware assisted compression, while in low bandwidth environments like the Internet usually; aggressive low bit-rate coding can be carried out in software using different operating systems (OS). VIC can be used for unicast, broadcast, and multicast communication. It is based on the Internet Draft Standard Real-time Transport Protocol (RTP) developed by the IETF Audio/Video Transport working group. RTP is an application-level protocol implemented entirely within VIC. Many different implementations of VIC are tested and used in the test bed to provide better variety of video codecs and functionalities. Version 2.8ucl1.1.5 is mainly used for both MS Windows and Linux environments.

### 2.3.3 RAT (Robust Audio Tool), Open-Source Audio Conferencing and Streaming Application

RAT is also based on IETF standards, using RTP above UDP/IP as its transport protocol, and conforming to the RTP profile for audio and video conference with minimal control. RAT supports multiple operating systems and offers unicast and broadcast functionalities, and for multiparty conferencing it uses IP multicast. The type of communication can be either push-to-talk (PTT) or regular continuous voice connection. RAT features a range of different rate and quality codecs, receiver based loss concealment to mask packet losses, sender based channel coding in the form of redundant audio transmission, and encryption. There exist various implementations of RAT. The versions mostly used with the test bed are UCL 4.2.23 and 4.2.25.

### 2.3.4 VLC Media Player 0.8.4a, Cross-Platform Media Player and Streaming Server

VLC (initially Video LAN Client) media player is a cross-platform multimedia player and streaming server supporting various video and audio formats like MPEG-1, MPEG-2, MPEG-4, DivX, mp3, and ogg and various streaming protocols like RTSP and RTP. It can be used to stream in unicast, broadcast, and multicast using IPv4 and IPv6 protocols under multiple operating systems e.g. MS Windows and Linux. The streaming can simply be controlled from the server side, but also VOD (Video On-Demand) solutions are possible.

### 2.3.5 Naval Research Laboratory (NRL) Optimized Link State Routing (OLSR) Protocol Implementation

NRL OLSR is a link state routing protocol oriented for mobile ad hoc networks (MANET). It is largely based on the Optimized Link State Routing (OLSR) protocol specification RFC 3626 [6]; however the NRL code base has several additional options and features, including:

- support for IPv6
- operational in MS Windows, MacOS, Linux, and various embedded PDA systems such as Zaurus and PocketPC
- full link state topology can be distributed including non-multi-point relay (non-MPR) cross links
- a "willingness" attribute for localized multi-point relay (MPR) activation

- support for several MPR selection protocols: Classical flooding (CF), non-source-based MPR (NS-MPR), source-based MPR (S-MPR), MPR connected dominating set (MPR-CDS), and essential CDS (E-CDS)
- neighbor link quality assessed by a smoothed hysteresis function
- many run-time parameters available including: HELLO interval, link state update interval, timeout factors, link quality assessment parameters, MPR willingness, and message type of service (TOS)
- configurable debugging verboseness
- experimental features such as fuzzy-sighted routing and support for Simplified Multicast Forwarding (SMF)

NRL has implemented OLSR using the Protolib programming toolkit which allows for cross-platform support including building code into different network simulation environments (e.g. NS-2, OPNET). Protolib abstraction also provides for potential use of code in protocol stacks alternative to Internet Protocol (IP) if needed. Classical flooding and S-MPR are the primary algorithms used in conjunction with simplified multicast forwarding (SMF) [5] engine in the tests.

### 2.3.6 NRL Simplified Multicast Forwarding Engine

The NRL SMF [5] project includes software for a user-space forwarding engine. This software was developed by the (NRL) Protocol Engineering Advanced Networking (PROTEAN) Research Group. The goal of NRL SMF is to provide an implementation of experimental techniques for robust, efficient distribution of broadcast or multicast packets in dynamic, wireless networks such as MANET. The NRL SMF application can be run as a stand-alone application capable of providing the "classic" flooding of broadcast and multicast traffic with duplicate detection for a specified network interface or can be used in conjunction with a controlling program to perform more sophisticated multicast forwarding utilizing different algorithms. An interprocess communication "remote control" interface is provided so that a compatible program (e.g. NRL OLSR) may issue real-time commands to NRL SMF to control the multicast forwarding process. Both IPv4 and IPv6 operation are supported. NRL SMF can be built for the following operating systems: Linux, MacOS, BSD, Win32, and WinCE.

### 2.3.7 Other Test Bed Software

A list of other important software that is used directly as public safety applications or to support the running of the applications is presented below.

- **Google Earth 3.0, 3D interface to the planet**

  The program is used as GIS application to provide map interface and for tracking purposes.
- **TrackerGE, GPS/APRS real-time tracking data to Google Earth (or OziExplorer)**

  This small program converts the raw serial GPS data to suitable format for Google Earth.
- **Apache Tomcat 5.5, web server**

  The web server is used to reply (GPS) tracking information queries and to provide database access.
- **Java Runtime Environment 5.0**

  It is required to run the database and map application.
- **MySQL 5.0, SQL database server**

  It provides the database itself.
- **Demonstrational database and map application**

  This application provides information about available public safety resources and combines the information with geographical data.

## 2.4 Test Bed Applications

### 2.4.1 Unicast and Multicast Ad Hoc Routing

For unicast applications a typical representative of link state routing algorithms (OLSR) is used. For multicast applications, Simplified Multicast Forwarding (SMF) running in conjunction with OLSR (or stand-alone when flooding) is employed. Different algorithms for the selection of the multi-point relay set (MPR) are possible (Classical Flooding, S–MPR, NS–MPR, and ECDS).

Flooding and S-MPR were chosen to be used mostly. Flooding is the most reliable method in a way that packets are always forwarded in every node (except duplicates) and having small number of nodes (e.g. 2 - 10) does not cause excessive amount of simultaneous transmissions of packets unless all the nodes are within very close range and can hear most of the other nodes directly. The functionality of different kind of unicast data transfers were also tested (e.g. Internet connection, database (DB) transactions, GPS location data, map and other aerial information).

Both unicast and multicast communication are investigated. However, the main focus is on real-time video streaming application scheme over multiple hops using multicast with maximum three hops with the test bed and nine hops with the simulations.

### 2.4.2 Video Aspects

For video communication the Video Conference (VIC) program is used with H.261 codec. H.264 implementations were not robust enough for practical testing nor demonstrations since the software crashed quite often. Basic settings for video are CIF 352 x 288 or lower resolution and RGB 24 or fewer colors. At the times there is considerable movement in the video image, 15 fps frame rate is tried to be achieved at the sources to provide feasible video quality at the receiver for demonstrational purposes. With the chosen standard quality settings and 15 fps, the average data rate of the variable bitrate video was estimated to be around 300 kbps.

### 2.4.3 Audio Aspects

For voice over IP (VoIP) the Robust Audio Tool (RAT) application is used. Normal GSM coding scheme was selected which gives theoretical maximum data rate of 13.2 kbps (8 kHz mono audio is used). Redundant transmission is used as a sender based loss protection which means that heavily compressed copy of sent audio packets are piggy-packed onto the next packets. Pattern matching is also utilized in the receiver. That technique uses the audio before and after the loss to interpolate a suitable signal for replacing the missing and corrupted part of the audio in the middle.

### 2.4.4 Tracking

Tracking of the nodes belonging to the network is demonstrated utilizing Global Positioning System (GPS). The GPS data is retrieved from the files in the web servers (Apache Tomcat) running on the source nodes. The files containing the GPS location data are generated with the TrackerGE program which uses standard serial communication with the GPS devices.

### 2.4.5 DB Transactions and Downloads

A combined geographical information system (GIS) and resource database (DB) application is used for demonstrational purposes. Different data transfers (e.g. textual data and images) can be executed to demonstrate carrying out various public safety tasks utilizing DB applications. Only unicast communication is used for this kind of data traffic.

### 2.4.6 Gateway

One node belonging to the ad hoc network (LAN) acts as a gateway. It has WLAN interface for the LAN and general packet radio service (GPRS) interface is used to provide a connection to the Internet (WAN). There are no special gateway functionalities (e.g. link selection based on the link availability or specific QoS services) other than providing the WAN connectivity with an address translation between the LAN and WAN interface of the gateway. In the experiments fixed IP addresses were used for the LAN side and WAN interface has dynamic address provided by the operator of the connection. The use of TETRA/TEDS or WiMAX instead or in addition to GPRS will be considered in the near future.

## 2.5 QualNet Network Simulator

QualNet software is targeted for developing new communication technologies through network modeling and simulation. QualNet can be used as a tool for improving design, operation, and management of various networks through virtual networking concept. Virtual networking enables the testing, optimization and integration of network technologies in a laboratory environment at significantly lower costs compared to deploying physical test beds. QualNet can also interact with real test bed hardware and emulate e.g. a certain part of the network. This feature is called hardware in the loop. [4]

The performance of networks and networking protocols can be investigated and predicted through simulation and emulation using QualNet. Different kind of conditions can be arranged and repeated in a controllable way, e.g. investigating the behavior of the network under undesired conditions, which can be difficult to recreate artificially but are very important to study and understand. It is beneficial to use a simulator to investigate such difficult scenarios. Especially large-scale scenarios (i.e. high number of network nodes and long distances) are difficult and expensive, if not impossible to organize with real hardware in a laboratory environment.

### 2.5.1 Main Characteristics of QualNet Simulator

QualNet supports real-time and faster than real-time simulation speeds, but it can be slower than real-time as well. QualNet is extensible being able to connect to other hardware and software applications. This enables software, hardware, and human interaction within the simulation process. It is possible to connect a real network test bed with QualNet and extend the testing this way. For example, a real video feed can be established from a desktop computer which connects to a simulator server machine through e.g. Ethernet interface. Then the simulator emulates a complex network and finally the video feed is forwarded though e.g. WLAN interface back to another real desktop computer. The advantage of QualNet is its scalability and it can be used to simulate large scale simulations and complex protocol stack including all the network reference model layers from physical to application layer even for thousands of nodes within reasonable simulation times. Networking aspects can be modeled in detail either with available model libraries or by developing extensions of one's own making for QualNet. However, creating or modifying protocols, models and network architectures, or other networking functionality can be quite extensive task and it requires programming skills. C++ programming language is normally used, since it is native for QualNet and therefore it is possible to modify comprehensively the simulator itself. QualNet runs on a multiple platforms, including Linux, Solaris, Windows XP, and Mac OS X operating systems, distributed and cluster parallel architectures, and both 32- and 64-bit computing. It can be used both from graphical user interface or command line. [4]

### 2.5.2 Important Variable Attributes and Network Types in QualNet

There are multiple wired and wireless network media which can be utilized and also combined to establish hybrid networks. For example, a hybrid of IEEE 802.11 WLAN and Ethernet can be established like it is done for test case with access point described in paragraph 4.4. The amount and type of network nodes can be varied and groupings and hierarchies can be created for them. The environmental characteristics can be adjusted as well. It is possible to set channel properties such as frequency, slow and fast fading, pathloss according to different channel models and shadowing. It is basically possible to model any layer, device, or system. A big variety of statistics can be created and used as well. The statistics can be enabled and disabled on a per-node, per-protocol, or per-layer basis [4].

# 3. EXPERIMENT WORK ON ESTIMATION OF MAC LAYER AND PHY LAYER PARAMETERS FOR IEEE 802.11B AND IEEE 802.11G

## 3.1 Background

Certain important parameters in physical (PHY) and media access control (MAC) layers needed to be identified through experiments due to the fact that these parameters could not be verified from the manufacturer's technical specifications. On the other hand, it is also important to make the simulation results from QualNet simulator comparable with the results obtained from the test bed. Therefore, keeping the consistency of the parameters in both of the test bed and QualNet is a prerequisite for any further investigation. Some of the parameters may be considered as constants for all manufacturers while some of them may vary due to the differences among the implementations and the standard specifications as well [11]. Transmission power and receiver sensitivity along with many MAC protocol related parameters were among the ones needed to be measured. According to [12] the throughputs (packets per second) for different packet sizes in terms of payload at the transport layer were measured. The setup included two PCs connected in ad hoc mode with one of them constantly sending packets to achieve saturation condition (UDP flooding).

## 3.2 Performance and Verification Measurements of 802.11b/g System

The used test bed setup for the basic performance and parameter measurements consists of two Linksys Wireless-G WMP54G PCI WLAN adapters running on desktop PCs and one Linksys Wireless-G WPC54G PCMCIA WLAN adapter [13] running on a laptop computer. Open air in nearly optimal line-of-sight (LOS) conditions and coaxial RF cable were used as transfer media. The PCI cards provide antenna connection and can be used with RF cable whereas the PCMCIA card can only be used with open air connection. The RF cable medium included also fixed and variable attenuators for ensuring adequate signal quality and to simulate distance and other characteristics of radio path. Rohde&Schwarz FSEB20/30 spectrum analyzer was used for measurements. LanTrafficV2 [14] and LanEval software applications running on both PCs were used to create UDP unicast and broadcast traffic. The LanEval program can be used to detect packet error rate directly furthermore. Ethereal and AiroPeek NX were used for packet sniffing purposes. Some other packet generator and sniffer applications were also used to verify the results and their consistency. The duration of experiments for all cases was fixed to be 30 seconds.

The first experiments were performed to find out and verify the parameters of 802.11b/g physical and MAC layers implemented by the test bed equipment. The tests were performed between two stations (PCs) using both open air and RF cable without any contention for the medium, extra interference or additional traffic. The stations formed an independent basic service set (IBSS), which means they communicate directly with each other without any access points (AP) between them. The packet generator software was used to generate flooding (saturation) conditions i.e. there is no extra inter packet delay between sending the next packet after the previous one. UDP unicast traffic was used with different UDP payload sizes from 6 to 1472 bytes.

The parameters of 802.11 PHY and MAC should be possible to verify since we know how they are affecting the distributed coordination function (DCF) of the standard 802.11 MAC [15-17]. DCF is based on carrier sense multiple access / collision avoidance (CSMA/CA) protocol. In this listen-before-talk scheme each station listens to the medium and sends only when it is sensed idle. Naturally when two or more stations detect the medium as being idle and start sending at the same time, collision occurs, but in the test setup there is no contention i.e. there cannot be collisions because only one station transmits packets. Therefore the medium is always idle after the previous packet sending with one exception of 802.11 beacon frames sent regularly but rarely.

When the transmitting station starts trying to send a packet, it senses the medium and detects it as being idle for the time of DCF interframe space (DIFS). Then according to initial minimum contention window $CW_{min}$ and slot time (10 µs for 802.11b/g) parameters, it keeps sensing slot time multiplied by random number [0,$CW_{min}$-1], until it starts sending the MAC protocol data unit (MPDU), since the medium is still idle. The receiving stations replies with acknowledgement (ACK) after the time of short interframe space (SIFS), when the transmission succeeded. After successful delivery of the MPDU, this procedure starts from the beginning. The packet structure of the MPDU is depicted in Figure 1. The MPDU consists of 802.11 MAC header of 28 bytes, 802.2 LLC header of 8 bytes, and MAC service data unit, which on the other hand consists of IP header of 20 bytes, UDP header of 8 bytes and UDP payload. Thus, total overhead is 64 bytes.

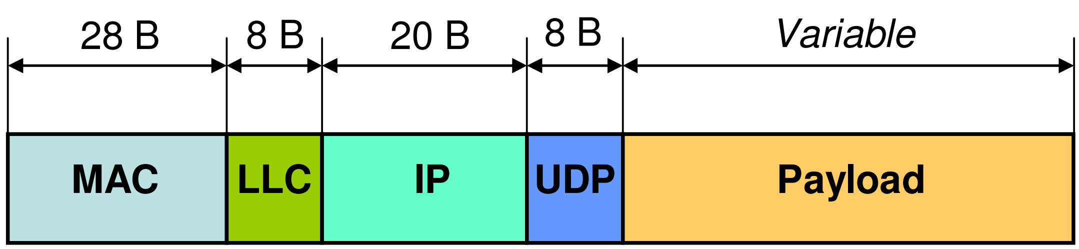


**Figure 1:** MPDU packet structure and header sizes

Using above mentioned values, variables and the measured throughput from the packet generator of the transmitting station, it is possible to calculate and estimate the variables SIFS, slot time $\sigma$ , $CW_{min}$ and DIFS.

$$\mathrm{DIFS} = 2 \times \mathrm{SIFS} + \sigma \tag{3.1}$$

This is possible since we know the allowed standard values for them. The value of the PLCP preamble and PLCP header delay $T_{PHY}$ at the PHY layer are 96 µs or 192 µs for 802.11b and 20 µs for 802.11g ERP OFDM. The slot time is always 20 µs for 802.11b and can be 9 µs or 20 µs for 802.11g according to the mode if some protection for co-operation with 802.11b equipment is utilized or not. The $CW_{\min}$ is 31 for 802.11b and can be 15 or 31 for 802.11g. The time for contention window is random, but it can be averaged since the amount of sent packets is very high.

$$\overline{T_{\text{backoff}}} = \frac{CW_{\min}}{2} \times \sigma \tag{3.2}$$

Time used for transmitting one packet with UDP payload with all the overhead and ACK can be calculated in microseconds IEE802.11b ($T_b$) and 802.11g ($T_g$) with

$$T_b = \text{DIFS} + \overline{T_{\text{backoff}}} + T_{\text{PHY}}^{b} + \left\lceil \frac{(\text{Overhead} + \text{Data}) \times 8}{R_{\text{bit}}} \right\rceil + \text{SIFS} + T_{\text{ACK}} \tag{3.3}$$

$$T_g = \text{DIFS} + \overline{T_{\text{backoff}}} + T_{\text{PHY}}^{g} + \left\lceil \frac{16 + (\text{Overhead} + \text{Data}) \times 8 + 6}{N_{\text{DBPS}}} \right\rceil \times T_{\text{sym}} + T_{\text{sigext}} + \text{SIFS} + T_{\text{ACK}} + T_{\text{sigext}} \tag{3.4}$$

where $T_{\text{ACK}}$ is the time to send the ACK, number of data bits per symbol $N_{\text{DBPS}}$ is the number of data bits sent per OFDM symbol (4 x data rate), symbol time $T_{\text{sym}}$ is $N_{\text{DBPS}}$ divided by data rate $R_{\text{bit}}$, and $T_{\text{sigext}}$ is a signal extension time. $R_{\text{bit}}$ can be 6, 9, 12, 18, 24, 36, 48, or 54 Mbps for 802.11g and 1, 2, 5.5, or 11 for 802.11b. Furthermore, 802.11g OFDM data transmit requires 16 PLCP service bits and 6 pad bits and utilizes 6 µs signal extension required by the high-rate coding. *ACK* is 14 bytes long.

## 3.3 Timing Scheme Estimation

### 3.3.1 PHY and MAC Layer in IEEE 802.11b

The highest channel bitrate of 11 Mbps has been used. Different UDP packet lengths have been employed. The UDP payload of the smallest packet equals 6 Bytes (LanTrafficV2 software limitation) and the highest one equals 1472 Bytes (Ethernet limitation). The RTS threshold has been set sufficiently high in order to exclude virtual carrier sensing (VCS) at the MAC layer i.e. all the transmitted packets are smaller than the threshold value and no RTS/CTS function is utilized. The results of the throughput are following in Table 2. More results are presented in the Table 1.1 of Appendix 1. It is noted that the results in this chapter are tested in ad hoc mode without Access Point (AP). The standard deviation of all the throughput performance measurements conducted here were found to be approximately below 0.5 percents of the average values and therefore the results are considered accurate enough for the comparisons and generally.

**Table 2:** Transmitted packets per second for 11Mbps in IEEE 802.11b in the test bed

| **UDP Payload (Bytes)** | 6 | 200 | 500 | 1000 | 1300 | 1472 |
|---|---|---|---|---|---|---|
| **Packets/sec** | 1700 | 1380 | 1057 | 757 | 651 | 600 |

The frame exchange sequences in CSMA/CA medium access scheme of IEEE 802.11 is depicted in Figure 2. The interval of RTS/CTS is optional and not used in the timing scheme estimations. Therefore it is not depicted. In the case of broadcast or multicast without AP no ACKs are utilized in the MAC layer [15-17].

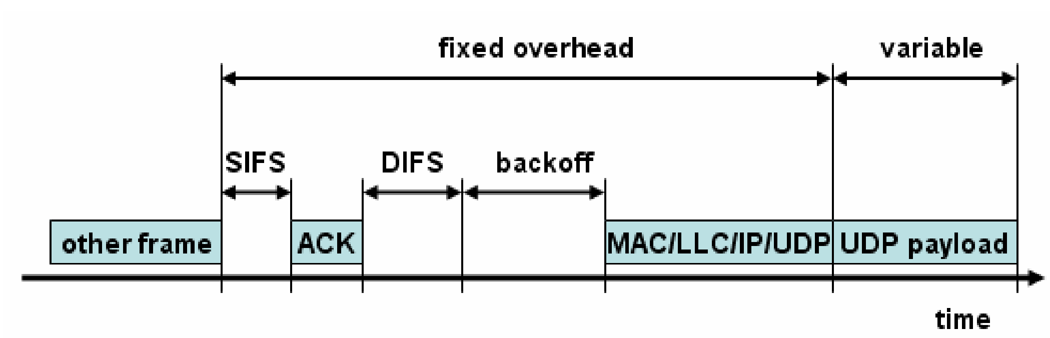


**Figure 2:** IEEE 802.11 frame exchange sequence with CSMA/CA scheme

Since the number of packets sent is quite high a mean experimental value of the time needed to transmit just one packet can be obtained. From theoretical point of view the calculation of the time needed to send one packet must include the time for the transmission of the UDP payload, the UDP, IP, LLC and MAC headers (8+20+8+28=64 bytes), the ACK (14 Bytes), the SIFS, the DIFS, two PLCP preambles and two PLCP headers (one for the data, one for the ACK) and the random backoff time. Thus, the average value calculated based on standards for one packet transmission is evaluated by adding the average backoff time to the other constant parameters. The sum of PLCP preamble and header can be either 96 μs or 192 μs and this is investigated. The two average values obtained by arithmetic calculation and experiment must be close enough.

The comparison between the two results rules out the possibility of long PLCP preamble in the test bed. The arithmetic calculation using equation (2.3) is presented below in the case of the largest packet when the short preamble is used. The largest packet size has been selected among all since its packet throughput results do not fluctuate at all.

$$\begin{aligned} T_b &= 50\mu s + \frac{31 \times 20\mu s}{2} + 96\mu s + \left\lceil \frac{(64B + 1472B) \times 8}{11Mbps} \right\rceil \\ &+ 10\mu s + 96\mu s + \left\lceil \frac{14B \times 8}{11Mbps} \right\rceil = 1691\mu s \end{aligned} \tag{3.5}$$

The experimental time 1667 μs to send the largest packet is obtained from the measured packet rate (600 packets/s).

Usually experiment results are slightly lower but close to the arithmetic results. This is quite surprising, particularly since the small beacon overhead is not included in the arithmetic calculation. Some possible reasons are speculated later. The conclusion for the values of the parameters used by the WLAN cards at the PHY and MAC layer is made from the results. Measurements of packet throughput have been carried out for the rest of the data rates of IEEE 802.11b as well. The results are summarized in Table 3 below.

**Table 3:** MAC and PHY layer parameters for IEEE 802.11b in the test bed.

| **Data Rate (Mbps)** | **Preamble (μs)** | $CW_{min}$ | **Slot Duration (μs)** | **SIFS (μs)** | **DIFS (μs)** |
|---|---|---|---|---|---|
| 11 | 96 | 31 | 20 | 10 | 50 |
| 5.5 | 96 | 31 | 20 | 10 | 50 |
| 2 | 96 | 31 | 20 | 10 | 50 |
| 1 | 96 | 31 | 20 | 10 | 50 |

The average time needed to transmit one packet is calculated based on the measurement of transmitter throughput in Table 1.1 of Appendix 1.

The use of beacons in the test bed has been verified by a packet sniffer program [16]. In the test bed beacons utilize the bitrate of 2 Mbps and their size equals 74 bytes. Beacons in ad hoc mode are transmitted with 2 Mbps even if the data is transmitted with 1 Mbps. They are transmitted every 100 ms in average.

The packet sniffer indicates that the length of the ACK employed by the test bed is 14 bytes and that ACK is sent with the same bitrate as the data packets. It is also verified with Sniffer that the time slot duration is indeed 20 μs. The consistency of the parameters between test bed and arithmetic calculations is plausible so far. Furthermore, the duration of the data payload and its overhead is calculated straightforward based on their length in bytes. The duration of SIFS does not affect much the result and its value is assumed to be the standard value of 10 μs. Therefore DIFS must equal 50 μs as well. Finally, the minimum contention window size and the PLCP preamble and PLCP header duration are left to be deduced. If the minimum contention window equals 15 time slots instead of 31 the theoretical duration of a frame decreases to 1527 μs. In this case the theoretical time needed to transmit one packet is shorter than the experimental but their large difference of 140 μs cannot be well justified.

It is known that the long PLCP preamble consists of 144 bits and the long PLCP header of 48 bits [14]. Both preamble and header are transmitted with the lowest bitrate (1 Mbps) and hence their total duration is 192 µs. On the other hand, the short PLCP preamble consists of 72 bits transmitted at 1 Mbps and the short PLCP header consists of 48 bits transmitted at 2 Mbps, and thus the duration of preamble plus header is 96 µs. If it is assumed that in the test bed the PLCP preamble and header are transmitted both at 2 Mbps, then the hypothesis reduces the duration from 96 µs to 60 µs. In this case, theoretical duration for the largest frame transmission reduces to 1615 µs. The differences in frame duration between theory and test bed for the packet sizes depicted in Table 1.1 of Appendix 1 fluctuate approximately between 30 µs and 50 µs when short preamble of 60 µs is supposed in the theoretical calculation. Experimental value is always higher. It is noted that the beacons transmitted at 2 Mbps every 100 ms, only add 4 µs to the frame duration of the largest packet size transmitted at 11 Mbps.

Measurements for packet throughput have been carried out for 5.5 Mbps and 2 Mbps as well. The frame duration evaluated theoretically is always 30 µs to 50 µs lower than the value obtained by experiment if the PLCP preamble and PLCP header duration is assumed to equal 60 µs. However, there is no official document to confirm our assumption. The better performance of the test bed may be caused by some other reason.

### 3.3.2 PHY and MAC Layer in IEEE 802.11g

The same idea of UDP packet flooding has been employed in order to estimate the MAC and PHY layer parameters of the IEEE 802.11g as well. The results for 54 Mbps are presented in Table 4. Extended results are given in the Table 1.2 of Appendix 1.

**Table 4:** Transmitter throughput (packets per second) IEEE 802.11g @ 54 Mbps in the test bed

| **UDP Payload (Bytes)** | 6 | 200 | 500 | 1000 | 1300 | 1472 |
|---|---|---|---|---|---|---|
| **Packets/sec** | 4900 | 4170 | 3517 | 2774 | 2458 | 2308 |

For the timing scheme the comparison between experimental and theoretical time needed to transmit a packet in the case of IEEE 802.11g is presented. The bitrate of 54 Mbps for the largest packet is chosen. It is found out with the packet sniffer that acknowledgements are sent at maximum bitrate of 24 Mbps. The minimum value of contention window equals to 15 slots, which can be deduced through the arithmetic calculation based on (3.3) as follows.

$$\begin{aligned} T_b &= 50\mu\text{s} + \frac{15\times 20\mu\text{s}}{2} + 20\mu\text{s} + \left\lceil \frac{16\text{b} + (64\text{B} + 1472\text{B})\times 8 + 6\text{b}}{216\text{b}} \right\rceil \times 4\mu\text{s} + 6\mu s \\ &+ 10\mu\text{s} + 20\mu\text{s} + \left\lceil \frac{16\text{b} + 14\text{B}\times 8 + 6\text{b}}{96\text{b}} \right\rceil \times 4\mu\text{s} + 6\mu\text{s} = 498\mu\text{s} \end{aligned} \tag{3.6}$$

The average time needed to transmit one packet in the test bed can be calculated according to Table 4. Experimental time for the largest packet (1472 B) is $10^6/2308$ µs = 433 µs. It is noted that the calculation of transmission time is based on the formula described in [16-17] and follows the same principles as described in chapter 3.3.1.

Beacons have been found again to use 2 Mbps bitrate for transmission and their overhead is not included in the theoretical calculation above. They are transmitted roughly every 100 ms. Similar to IEEE 802.11b case the experimental value is lower than the theoretical even if beacon overhead is not accounted in the latter one. This behavior is observed for all the bitrates of IEEE 802.11g in the test bed. The minimum contention window has been selected equal to 15 slots and this is the smallest value appearing in the standard. The slot duration of 20 µs has been verified with sniffer and it is consistent with the standard [16] for ad hoc only mode scenario. The DIFS value is calculated straightforward based on the slot duration and the SIFS value.

To sum up it has been found that the test bed performs better than theory in terms of packet throughput for all the bitrates of IEEE 802.11b and even more significantly for IEEE 802.11g. In both cases the lowest values of parameters appearing in the standards have been used. The results for PHY and MAC layer parameters when IEEE 802.11g is employed in the Test Bed are presented in Table 5.

**Table 5:** MAC and PHY layer parameters for IEEE 802.11g in the test bed

| Data Rate (Mbps) | Preamble (μs) | $CW_{min}$ | Slot Duration (μs) | SIFS (μs) | DIFS (μs) |
|---|---|---|---|---|---|
| 54 | 20 | 15 | 20 | 10 | 50 |
| 48 | 20 | 15 | 20 | 10 | 50 |
| 36 | 20 | 15 | 20 | 10 | 50 |
| 24 | 20 | 15 | 20 | 10 | 50 |
| 18 | 20 | 15 | 20 | 10 | 50 |
| 12 | 20 | 15 | 20 | 10 | 50 |
| 6 | 20 | 15 | 20 | 10 | 50 |

Real-time experiments for both IEEE 802.11b and IEEE 802.11g have been carried out in the presence of an AP as well. For this case the results for the throughput of the transmitter are presented in the Tables 1.3, 1.4 and 1.5 of Appendix 1. More discussion about the experiment results for both ad hoc and AP mode will be given in chapter 4 (Verification between Test Bed and QualNet).

## 3.4 Transmission Power Estimation

### 3.4.1 Problem Description

The aim of this experiment was to estimate the transmission power of the WLAN cards used in the test bed for the different data rates of IEEE 802.11b and IEEE 802.11g. The achieved results are critical in order to make the simulation and test bed results comparable. The experiment setup is given in Figure 3. The actual attenuation value of the attenuators, the splitter, the adapters, the connectors and the cables have been accurately measured by using a network analyzer [18] after careful calibration. The LanTrafficV2 software has been used to generate UDP traffic. Unlike the experiments executed so far we use UDP broadcast traffic for the measurement of transmission power and sensitivity. Under broadcast transmission no extra power comes from the receiver back to the sender, since no acknowledgements are sent back to the source. Therefore the measurement of transmission power is more accurate. The same applies for the sensitivity measurement.

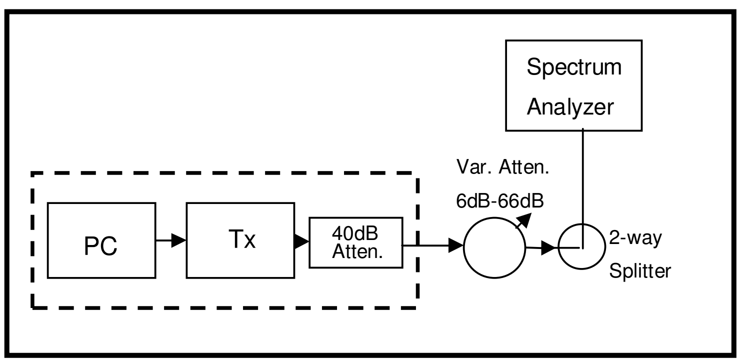


**Figure 3:** Transmission power measurement block diagram

### 3.4.2 Experimental Results

The Rohde&Schwarz FSEB20/30 spectrum analyzer [19] was used in the experiment. The channel power value is calculated using Parseval's relation law. The channel power measurement is performed by an integration of the measurement points within the channel bandwidth in frequency domain. The channel bandwidth is fixed to 20 MHz, which approximately contains all the significant channel power. This was verified visually as well. The standard states that the transmitted spectral products shall be less than -30 dB in the frequency area from 11 to 22 MHz away from the center frequency and -50 dB after that. For example, in the case of 54 Mbps the received channel power at the spectrum analyzer is -41.5 dBm. The actual value of the 40 dB attenuator has been measured to be 39.2 dB. The total attenuation of the rest of the elements including variable attenuator 6 dB, splitter, cables, connectors and adapters equals 12.3 dB. Transmission power $P_{\mathrm{dBm}}$ can be concluded for example for 54 Mbps bitrate

$$P_{\mathrm{dBm}} - 39.2\mathrm{dB} - 12.3\mathrm{dB} = -41.5\mathrm{dB} \Rightarrow P_{\mathrm{dBm}} = 10\mathrm{dBm} \tag{3.7}$$

(The value given in the specification of the WLAN cards equals 15 dBm). The transmission powers of the other data rates of IEEE 802.11b/g can be calculated experimentally in the same way. The results are presented in the following Table 6.

**Table 6:** Transmission power for different data rates

| Data Rate (Mbps) | Tx power (dBm) | Data Rate (Mbps) | Tx power (dBm) |
|---|---|---|---|
| 54 | 10 | 11 | 16.5 |
| 48 | 10 | 5.5 | 16.5 |
| 36 | 12 | 2 | 16.5 |
| 24 | 12 | 1 | 16.5 |
| 18 | 13 | | |
| 12 | 13 | | |
| 6 | 13 | | |

## 3.5 Sensitivity Measurement

### 3.5.1 Problem Description

The purpose of this experiment was to estimate the sensitivity of the WLAN cards for the different data rates of IEEE 802.11b and IEEE 802.11g. The sensitivity is assumed to be the ability of a radio to lock on to a sufficiently strong signal in the presence of interfering signal, e.g., the radio capture effect. WLAN card specifications defines the value of packet error rate equaling to 10 % for 54 Mbps, 48 Mbps, 36 Mbps, 24 Mbps, 18 Mbps, 12 Mbps and 6 Mbps and equaling to 8 % for 11 Mbps, 5.5 Mbps, 2 Mbps and 1 Mbps. In order to be able to track the value of packet error rate the LANEVAL program was used to generate and measure broadcast UDP traffic [20]. The software must run on both PCs.

### 3.5.2 Procedure of Experiment

The experiment setup is displayed in Figure 4. The attenuation caused by splitter is the same for both two outputs. The actual attenuation between splitter and receiver PC including cables, connectors, adapters and 16 dB constant attenuation was 20.2 dB (The actual value of the attenuation was measured accurately with the network analyzer). UDP packets are transmitted at certain data rate and then the variable attenuation is increased slowly until the packet error rate at the receiver reaches 10 % under IEEE 802.11g and 8 % under IEEE 802.11b. The corresponding received power at this moment is observed at the spectrum analyzer. Then the receiver sensitivity can be calculated straightforward by subtracting 20.2 dB from the spectrum analyzer value. [21]

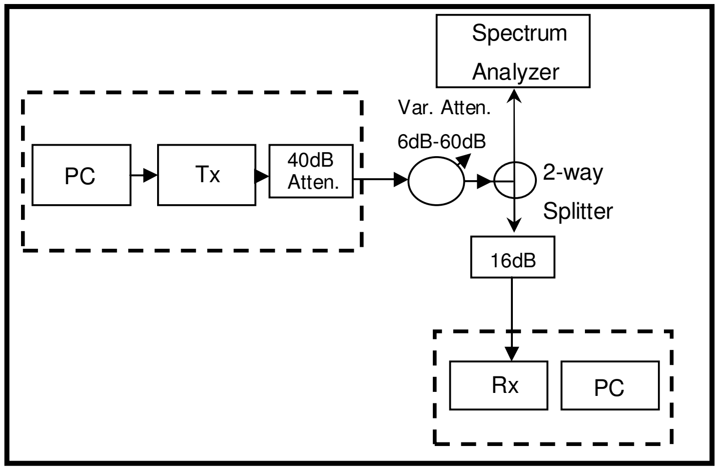


**Figure 4:** Block diagram for sensitivity measurement

### 3.5.3 Experimental Results

After carrying out the test procedure described above the results are tabulated in Table 7 for the values of the sensitivity for the different data rates.

**Table 7:** Sensitivity values for different data rates

| Data Rate (Mbps) | Sensitivity (dBm) |
|---|---|
| 54 | -67.4 |
| 48 | -69.8 |
| 36 | -73.8 |
| 24 | -74.6 |
| 11 | -77.5 |

## 3.6 Fallback Measurement

### 3.6.1 Problem Description

The purpose of this experiment was to measure attenuation (pathloss) between transmitter and receiver PC at the moment of a fallback to a lower data rate. These values are used as test inputs for the pathloss matrix in QualNet 3.9

Network Simulator. The setup used is the same as the one used to evaluate the MAC and PHY layer parameters in chapter 3. The attenuation along the cable is slowly increased by adjusting the value of a variable attenuator until a fallback occurs. The value of the total attenuation along the cable at the moment of occurred fallback is tracked down. This value is equivalent to the value of the pathloss between the 2 PCs. The experimental results are presented into the Table 8 below.

**Table 8:** Path loss values at the data rate fallbacks

| Data Rates (Mbps) | Path Loss (dB) |
|---:|---:|
| 54 → 48 | 81.6 |
| 48 → 36 | 84.6 |
| 36 → 24 | 90.1 |
| 11 → 5.5 | 98.1 |

# 4. VERIFICATION BETWEEN TEST BED AND QUALNET

## 4.1 Comparability of the test bed and simulations

The purpose of this task was to achieve consistency of the parameter settings between the test bed and the QualNet simulator. The parameters of IEEE 802.11b and IEEE 802.11g in QualNet are configured based on the results presented in Tables 3 and 5 respectively. A single client-server scenario in ad hoc mode was simulated to measure the maximum UDP payload throughput at the application layer as a function of the UDP payload size. The throughput results from the test bed and QualNet are compared.

The configuration file of QualNet includes the transmission powers and sensitivity values presented in Tables 6 and 7. The goal of the simulation was to check how the results of the simulator correspond to those of the test bed presented in Tables 2 and 4. For the fallback study, a simulation scenario was designed to measure the pathloss value between two nodes at the moment of a fallback.

## 4.2 Parameter Configuration of MAC and PHY Layer of IEEE 802.11b

The MAC and PHY parameters in QualNet were set according to the results depicted in Table 3. The consistency of parameter settings between test bed and simulator has to be maintained. Table 9 presents the parameter settings in QualNet. The preamble duration equals 96 μs in the simulations.

**Table 9:** MAC and PHY layer parameters of IEEE 802.11b in QualNet

```
#define PHY802_11b_SYNCHRONIZATION_TIME (96*MICRO_SECOND)
#define M802_11b_CW_MIN           31
#define M802_11b_CW_MAX           1023
#define M802_11b_SLOT_TIME        (20*MICRO_SECOND)
#define M802_11b_SIFS             (10*MICRO_SECOND)
M802->difs = M802->sifs + (2*M802->slotTime)
```

A UDP flooding scenario was used to measure the UDP payload throughput in QualNet. UDP flooding means that a UDP packet must always exist in the queue and be ready for transmission. The constant bitrate (CBR) traffic generator sends out UDP frames constantly. The number of generated packets has to be set sufficiently high. The number of dropped packets from the queue of the transmitter should be observed at the end of the simulation. If this number equals zero, a flooding situation has not been achieved and the number of generated packets should be increased. Preferably, the number of dropped packets from the transmitter queue should be high to secure UDP flooding situation i.e. the transmit buffer is saturated.

The duration of CBR traffic in QualNet was fixed to 10 seconds for all the simulations. The number of transmitted packets per second and the throughput of the transmitter were the criteria to be monitored. In the case of 11 Mbps, we observed that 7445 1000 byte packets were transmitted within the specified interval. The average backoff value in the simulations has been found to be 15.47 time slots and it is quite close to the theoretical average value of 15.5 time slots in case the initial minimum contention window is 31.

Tables 2.1 and 2.2 in Appendix 2 present the throughput values obtained by QualNet in ad hoc mode (no AP). The channel data rate is set to 1 Mbps and 11 Mbps respectively. It is worth mentioning that the results produced by QualNet match almost exactly the arithmetically calculated values. This happens because beacons are not generated by QualNet in the ad hoc mode. On the contrary, it has been discovered with the sniffer [20] that beacons are transmitted always with 2 Mbps in the ad hoc mode of the test bed.

Based on the results presented in the Appendices in Tables 1.1, 2.1 and 1.2, 2.2 the throughput of the transmitter versus UDP payload size was plotted. This way QualNet and test bed throughputs at 11 Mbps can be compared. The maximum UDP payload size equals 1472 Bytes. The Ethernet fragmentation threshold is 1500 bytes which corresponds to 1472 bytes UDP payload. This limitation does not exist in QualNet. However, in QualNet the default IP fragmentation unit equals 2048 bytes. Fragmentation threshold at the Ethernet limit can be imposed in QualNet by adjusting the value of IP fragmentation unit to 1500 bytes. Since the packet size employed even for high resolution video transmission is normally lower than the Ethernet limit such large packets are intentionally excluded from our graphs.

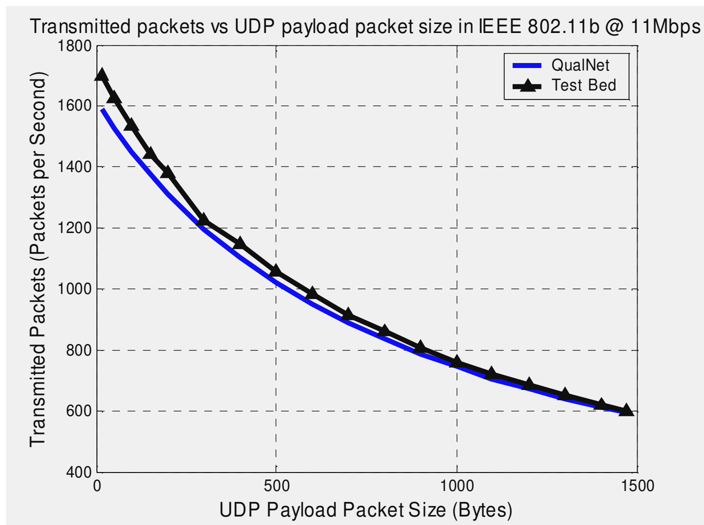


**Figure 5:** Transmitted packets per second in QualNet and test bed with IEEE 802.11b@11 Mbps in ad hoc mode

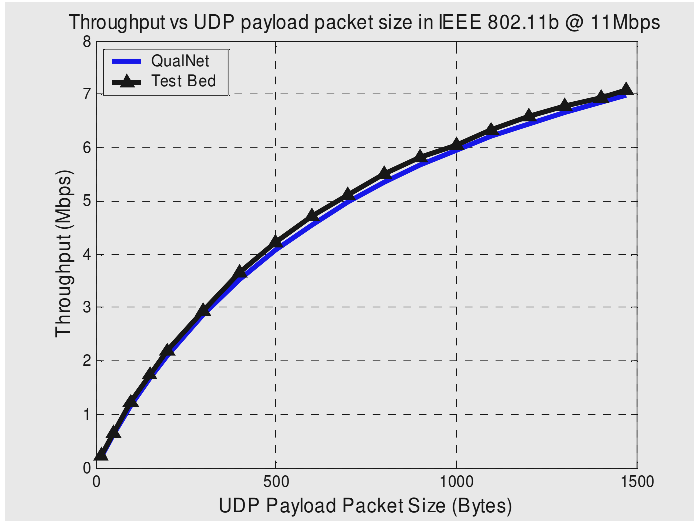


**Figure 6:** UDP throughput in QualNet and test bed with IEEE 802.11b@11 Mbps in ad hoc mode

From both Figures 5 and 6 it is seen that real system and QualNet perform quite close to each other. The closely matching performance has been verified for 5.5 Mbps bitrate as well. As it has been mentioned in 3.3.1, the better performance of the test bed compared to the simulator could be caused by certain undetermined reason e.g. using very short nonstandard preamble of 60 µs duration.

### 4.3 Parameter Configuration of MAC and PHY Layer of IEEE 802.11g

The PHY IEEE 802.11g model is not offered by the current version of QualNet. The PHY IEEE 802.11a at 5 GHz is available on the other hand. It is known that the ERP-OFDM is a physical layer introduced by IEEE 802.11g in order to provide IEEE 802.11a data rates in 2.4 GHz frequency band. The ERP-OFDM has been employed in the test bed experiments. The data rate set of IEEE 802.11g is emulated in QualNet by using the available data rates from IEEE 802.11a library.

For this purpose the frequency range defined in the configuration file of QualNet is altered from 5GHz to 2,4GHz. The other parameter settings in QualNet are kept unchanged. Table 10 presents the corresponding parameters in the configuration file of QualNet. The consistency of parameters between QualNet and test bed has been maintained.

**Table 10:** MAC and PHY layer parameters of IEEE 802.11a in QualNet

```
#define PHY802_11a_SHORT_TRAINING_SIZE  8 // 10 short symbols
#define PHY802_11a_LONG_TRAINING_SIZE   8 // 2 OFDM symbols
#define PHY802_11a_SIGNAL_SIZE          4 // 1 OFDM symbol
#define PHY802_11a_OFDM_SYMBOL_DURATION 4 // 4 usec
#define PHY802_11a_SERVICE_BITS_SIZE    16 // 2bytes * 8 = 16bits.
#define PHY802_11a_TAIL_BITS_SIZE       6 // 6bits  / 8 = 3/4bytes.
#define PHY802_11a_PREAMBLE_AND_SIGNAL \
    (PHY802_11a_SHORT_TRAINING_SIZE + \
     PHY802_11a_LONG_TRAINING_SIZE + \
     PHY802_11a_SIGNAL_SIZE)
#define M802_11a_CW_MIN          15
#define M802_11a_CW_MAX          1023
#define M802_11a_SLOT_TIME       (20*MICRO_SECOND)
#define M802_11a_SIFS            (16*MICRO_SECOND)
M802->difs = M802->sifs + (2*M802->slotTime)
```

Table 1.2 in Appendix 1 presents the throughput values in the test bed. Similarly, Table 2.3 in Appendix 2 presents the corresponding results obtained with QualNet. Based on the results from both of the Tables 1.2 and 2.3 the throughput comparison between test bed and QualNet is plotted in Figures 7 and 8. It is noted again that the performance of the test bed is always better than the theory and simulations. The simulations constitute lower bound for the real performance.

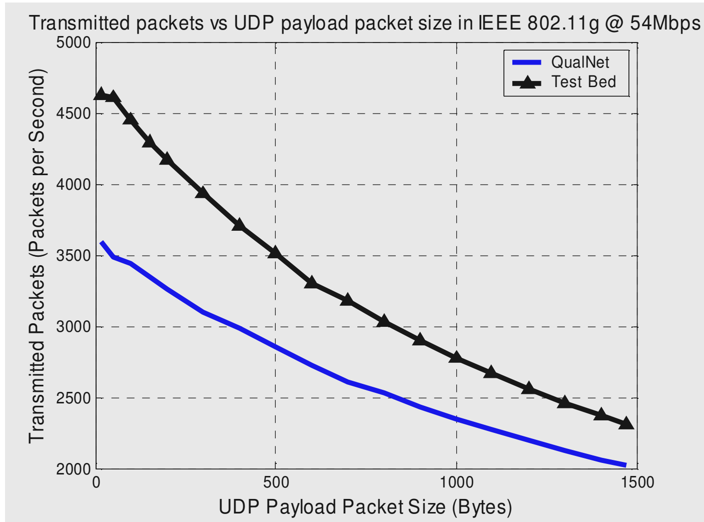


**Figure 7:** Transmitted UDP packet per second in QualNet and test bed with 802.11g@54 Mbps in ad hoc mode

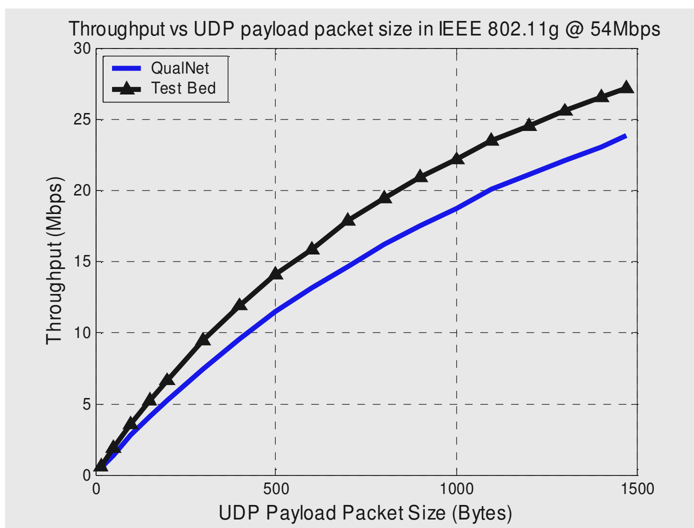


**Figure 8:** UDP throughput in QualNet and test bed with 802.11g@54 Mbps in ad hoc mode

### 4.4 Simulation for UDP Throughput with Access Point

In this task the consistency between parameter settings of test bed and QualNet in the presence of an AP was verified. The reason for repeating the experiments with AP is the use of time slot equaling to 9 μs in IEEE 802.11g. The difference in time slot duration between ad hoc and AP mode in the test bed has been discovered with the sniffer. The difference between the two operation modes: basic service set (BSS) and independent BSS (IBSS) can be found in [22].

The implementation of the scenario with AP in the test bed was straightforward. There is a wireless connection between the server PC and the AP and a fast Ethernet connection of 100 Mbps between the AP and the client PC. The LanTrafficV2 software was used to send constantly UDP packets in order to achieve flooding. Furthermore, the configuration file of QualNet needed to be modified to include an AP. The AP must be configured to have two interfaces: one wired and one wireless. The MAC layer parameter settings in QualNet are listed in Table 11 below.

**Table 11:** MAC layer parameter settings in QualNet configuration file with AP

```
MAC-802.11-BEACON-START-TIME 1
MAC-802.11-BEACON-INTERVAL 100
MAC-802.11-PC-CONTENTION-FREE-DURATION 20
MAC-802.11-PC-DELIVERY-MODE DELIVER-ONLY
MAC-802.11-STATION-POLL-TYPE NOT-POLLABLE
MAC-802.11-ASSOCIATION DYNAMIC

#Probing from the station is possible
MAC-802.11-SCAN-TYPE ACTIVE
MAC-802.11-SSID Test1
MAC-802.11-STA-CHANNEL 1
```

In the above configuration the transmission of beacons starts at first time-unit (TU) from the beginning of the simulation. One default TU equals 1024 microseconds. Beacons are transmitted every 100 TUs because this is the average transmission interval of beacons in the test bed. The parameter PC-CONTENTION-FREE-DURATION should be provided but it does not affect the results keeping in mind that the STATION-POLL-TYPE is NOT-POLLABLE or the DELIVERY-MODE is DELIVER-ONLY. Both of these parameters have been set accordingly. Dynamic association with the AP and active probing at channel

1 is defined. Since only one AP is included in the scenario, there is no difference between dynamic and static association. In addition the service set identification number is being given the string name ″Test1″. In fact, the configuration above emulates almost exactly the experiment carried out in the test bed. Although an AP is employed, no data packets are transmitted under point coordination function (PCF) since the server node is not pollable. In addition, for IEEE 802.11b the beacon frames are transmitted using the lowest bitrate (1 Mbps) both in QualNet and the test bed. In QualNet the beacon frames are of 56 byte size while the beacon size in the test bed equals to 74 bytes. In the test bed beacons are transmitted at 2 Mbps in ad hoc mode and at 1 Mbps in AP mode. However, for IEEE 801.11g beacon frames are transmitted at 1Mbps in test bed and 6 Mbps in QualNet in AP mode. The UDP throughput comparison between QualNet and the test bed is presented in the Figures 9, 10, 11, 12, and 13 below. The detailed results for transmitted packets per second for the different bitrates are given in the tables in the Appendices 1 and 2.

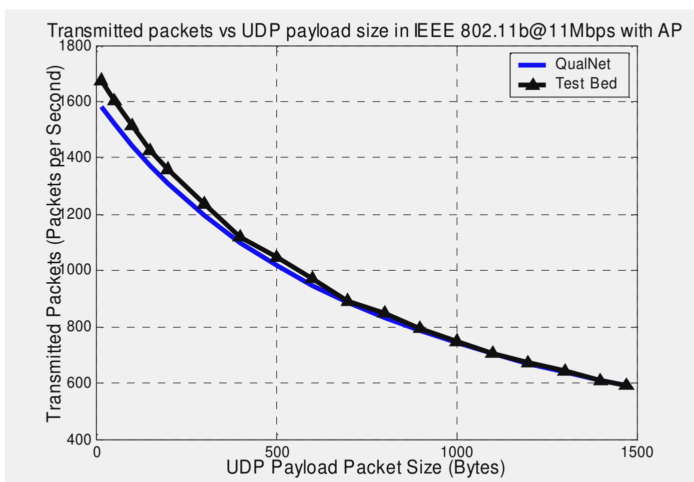


**Figure 9:** UDP packets per second in QualNet and test bed with 802.11b @11 Mbps and AP

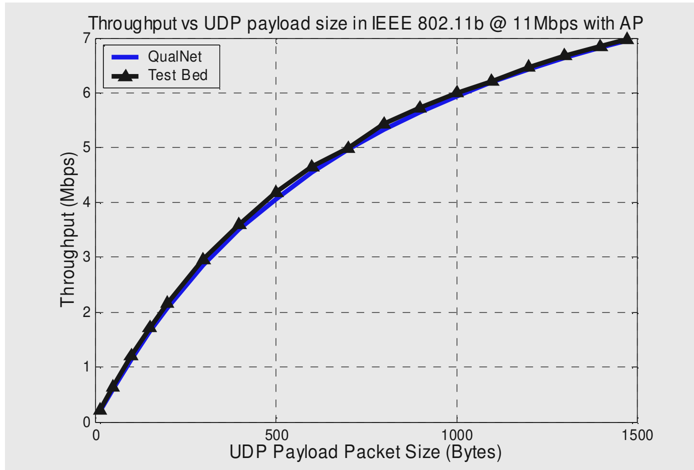


**Figure 10:** UDP throughput in QualNet and test bed with 802.11b@11 Mbps and AP

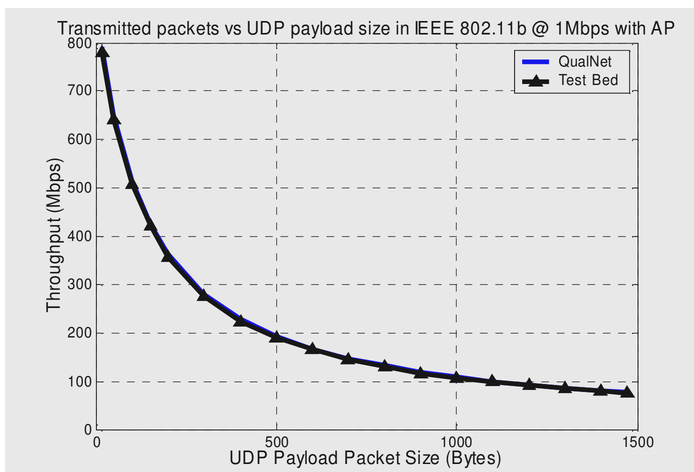


**Figure 11:** UDP packets per second in QualNet and test bed with PHY 802.11b@1 Mbps and AP

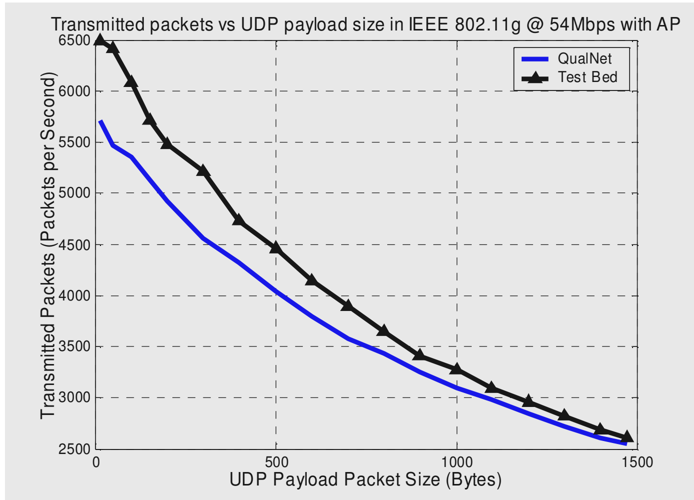


**Figure 12:** Transmitted UDP packets per second in QualNet and test bed with 802.11g@54 Mbps and AP

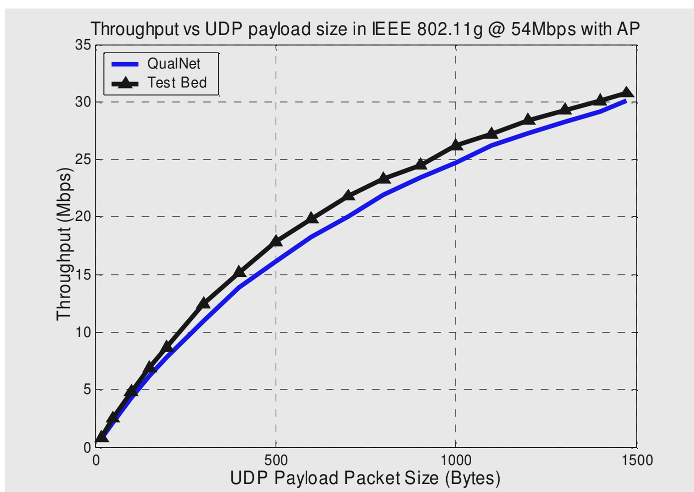


**Figure 13:** UDP throughput in QualNet and test bed with 802.11g@54 Mbps and AP

### 4.5 The Performance Difference between Test Bed and QualNet

The comparison of transmission throughput showed that the test bed performs better than QualNet simulator in both ad hoc mode and scenarios with AP. Especially; in IEEE 802.11g the better performance of test bed is significant throughout the range of UDP payload sizes. Furthermore, it is noticed that higher throughputs of test bed are more apparent in ad hoc mode scenarios. We note that the simulation results are approximately similar to the expected results because the simulator is calibrated according to the standards [15-17]. The reason that the test bed performs better than the theory has not been cleared.

## 4.6 Fallback Measurement Simulations

In Chapter 3.4 and 3.5 the transmission power and the sensitivity values have been estimated experimentally. The pathloss between client and server when a fallback occurs has been measured in Chapter 3.6. The experimental results have been summarized in Tables 6, 7, and 8. The sensitivity values for low data rates are omitted because of PER fluctuation at the receiver. For example, in the case of IEEE 802.11g the fallbacks to 9 Mbps or 6 Mbps could be observed after increasing the total attenuation beyond 90,1dB, and a fallback to 1Mbps of 802.11b happened when the attenuation was increased more than 98,1 dB. However no stable values could be observed. Therefore, only fallbacks to 48 Mbps, 36 Mbps, 24 Mbps in IEEE 802.11g and 5.5 Mbps in IEEE 802.11b are reported. For that reason only four data rate fallbacks are be compared between the test bed and the simulator. The transmission power and the sensitivity values obtained experimentally had to be imported into QualNet to make the comparison of the fallback results reliable. Since it is unnecessary complicated to derive radio channel propagation model appearing in indoor environment and the main purpose is to investigate protocol related performance factors, the propagation model used in the simulations is the simple pathloss propagation. This way the scenario can be kept adequately simple and matches quite well the radio cable implementation used in the test bed. An example of pathloss matrix in a two-node scenario at 2.4 GHz frequency band is given in Table 12. In the example, the pathloss values between the two nodes changes from 70.0 dB to 90.0 dB after the 160$^{th}$ second of the simulation time.

**Table 12:** Pathloss matrix file for two-node scenario used in QualNet

```
Freq:1:2.4
Nodes:2
#
#time    1stNode     2nd Node    pathloss(dB)
#------------------------------------------------
0           1           2           70.0
50          1           2           70.0
100         1           2           70.0
120         1           2           70.0
140         1           2           70.0
150         1           2           70.0
160         1           2           70.0
180         1           2           90.0
200         1           2           90.0
220         1           2           90.0
240         1           2           90.0
250         1           2           90.0
```

At the beginning the channel bitrate equals 54 Mbps and the PHY802.11-AUTO-RATE-FALLBACK is activated. An adequately small pathloss value between client and server is imposed in order to secure constant channel bitrate at 54 Mbps. At the 160$^{th}$ second of the simulation the pathloss value is increased. The data rate of the transmitter is observed throughout the simulation and the pathloss attenuation when a fallback to 48 Mbps occurs is written down. The same procedure is applied to track down the pathloss value when the channel bitrate falls from 48 Mbps to 36 Mbps, from 36 Mbps to 24 Mbps, and from 11 Mbps to 5.5 Mbps. The results are displayed in Table 8. The comparison of experimental and simulations results is given in Table 13.

**Table 13:** Pathloss values at the data rate fallbacks in test bed and QualNet

| Data Rates (Mbps) | Test Bed Pathloss (dB) | QualNet Pathloss (dB) |
|---:|---:|---:|
| 54 → 48 | 81.6 | 78.4 |
| 48 →36 | 84.6 | 80.0 |
| 36 → 24 | 90.1 | 85.4 |
| 11 → 5.5 | 98.1 | 94.4 |

The pathloss values observed in test bed and obtained by QualNet differ from 3 dB to 5 dB. The values from test bed are always larger. A possible reason for the offset could be explained with the different algorithms employed by the test bed and by the simulator for the decision of a fallback. The criterion being used to implement rate changes is not defined in the specifications of the WLAN cards since the interference detection and adaptive transmission rate control is not part of the standard and is implemented in a proprietary, undisclosed manner [12]. On the other hand, the source code of QualNet utilizes a scheme where the fallback to a lower bitrate happens when the number of unacknowledged packets equals `M802_11_NUM_ACKS_FOR_RATE_DECREASE` (default value is 2). In other words, the data rate is reduced only when acknowledgment timeout occurs for two times.

# 5. SYSTEM PERFORMANCE OF AD HOC ROUTING WITH MULTICAST PROTOCOLS

## 5.1 Metrics for Multicast Protocol Performance

The following performance metrics are considered [23-27].

**Data packet delivery ratio** measures the protocol effectiveness [24]. It is defined as the ratio of data packets delivered to the multicast receivers versus the number of data packets supposed to be delivered. The total number of received packets by all receivers is divided by the number of packets sent from senders multiplied with the total number of receivers.

**Average end-to-end delay** presents the average delay of data packets from application layer of multicast source to application layer of the multicast members [27]. It includes all the queuing and protocol processing delays as well as propagation and transmission delays. The lost packets are not accounted in the result.

**Control Overhead** describes the efficiency of the protocol [24-25]. Besides the delivery performance, the efficiency of the protocol in terms of the produced overhead should also be taken into consideration. For this purpose, the ratio of control and overhead bytes transmitted per data byte delivered is evaluated. In Control byte calculation the following kinds of packets contribute: the JREQs, the transmission and retransmission of JREPs, the active ACKs by the multicast sources and the MAC layer ACKs to any unicast transmission. Additionally, all the headers of the data packets transmitted and forwarded into the network are counted. On the other hand, only the data payload contributes to the data bytes delivered. It is noted that in control byte calculation every individual transmission of control packets is accounted.

## 5.2 Multi-Hop Multicast with SMF Routing Protocol

### 5.2.1 SMF Protocol and its Implementation in Test Bed

Simplified Multicast Forwarding (SMF) provides a basic IP multicast forwarding capability within a mobile Ad hoc network as well as mechanisms for interoperability with a connected wired infrastructure (e.g. Internet). SMF is open source software for LINUX or Microsoft Windows operating systems and publicly available [28]. Normally, SMF does not operate as a stand alone routing protocol but it works in co-operation with a unicast routing protocol like optimized link state routing (OLSR) or ad hoc on-demand distance vector (AODV). A unicast protocol is used to achieve efficient selection of a set of relaying nodes. Otherwise it performs Classical Flooding (CF) resulting in redundant broadcast transmissions and performance degradation due to network congestion and collisions. A detailed description of SMF functionality and a reference to some of the most common algorithms for relaying nodes selections can be found publicly [5].

In our test bed experiments OLSR source codes offered by U.S. Naval Research Laboratory (NRL) [28] has been used as the basis for supporting unicast routing protocol. To achieve better functionality of SMF, several algorithms for the selection of multi-point relay set (MPR) can be employed. In the version provided by NRL the source-specific MPR (S-MPR), non-source specific MPR (NS-MPR) and essential connected dominating set (ECDS) are implemented [22]. The performance of these algorithms is compared under square topology later in this chapter. It is possible that different algorithms for selection of MPRs could be employed in the future to optimize the SMF performance.

In order to measure accurately the average end-to-end delay, the network time protocol (NTP) has been installed at all the computers for clock synchronization. All the nodes are aligned to the clock of the server. During our experiments the time difference between the clocks were checked regularly. The experimental results were not accepted when the time differences between the source and any of the multicast receivers were found to be more than 0.1 ms. Better synchronization was achieved with NTP than with standard GPS devices.

### 5.3 Multicast Multi-Hop Forwarding with Linear Configuration, IEEE 802.11b/g

The first network configuration is depicted in Figure 14. The link connections were established by using cables. In this way it allows implementing the desired topology even within a single room. The objective is to achieve as optimal conditions for the links as possible to avoid any undetermined factors degrading the links. In fact, it must be noted that the use of the cables rules out the interference between the links. The nodes can communicate with high signal quality with their neighbors but non-adjacent nodes cannot see each other. This results in performance degradation due to hidden node problem. For instance, the multicast sender and the second hop node broadcast simultaneously their data packets resulting in collision at the first hop node. Similarly, collisions at the second hop happen because first and third hop nodes are hidden from each other. Every frame that is lost at a hop cannot be forwarded beyond and it counts as lost at the next hops since there are no retries in multicast or broadcast. Therefore it is expected that the packet delivery ratio is reduced from hop to hop. The different algorithms for the selection of the MPR set give the same result for a linear configuration of nodes. In out scheme, first and second hop nodes would form always the MPR set.

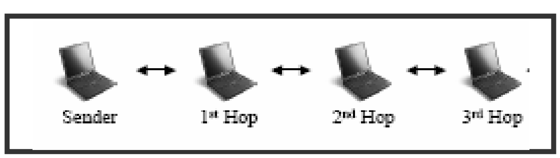


**Figure 14:** Linear Test Bed configuration for multi-hop multicast experiment

The average end-to-end delay and the packet delivery performance were tested at the different hops for various types of traffic and bitrates. For traffic generation the Multi-(Message) Generator (MGEN-3.3) available at [28] was installed. Video traffic and VoIP with different audio data payload settings were emulated. The end-to-end performance results for 2 Mbps and 11 Mbps of IEEE 802.11b and for 54 Mbps of IEEE 802.11g are presented in Tables 14, 15, and 16. Even the 2 Mbps datarate is adequate for managing the most demanding video and VoIP flows that were experimented in terms of packet loss and delay. The results are discussed further in this thesis when SMF-OLSR and ODMRP are compared.

**Table 14:** Linear Configuration of Test Bed at 2 Mbps

| **First Hop** | **Video 33pkts/s** | **VoIP 100pkts/s** | **VoIP 50pkts/s** | **VoIP 35pkts/s** | **VoIP 25pkts/s** |
|---|---|---|---|---|---|
| **Packet Loss (%)** | 0.2 | 0.32 | 0.3 | 0.18 | 0.3 |
| **Delay (s)** | 0.00675 | 0.002 | 0.002 | 0.003 | 0.003 |
| **Jitter (s)** | 0.2035 | 0.04 | 0.036 | 0.0017 | 0.033 |
| **Second Hop** | **Video 33pkts/s** | **VoIP 100pkts/s** | **VoIP 50pkts/s** | **VoIP 35pkts/s** | **VoIP 25pkts/s** |
| **Packet Loss (%)** | 0.63 | 0.59 | 0.46 | 0.28 | 0.68 |
| **Delay (s)** | 0.012 | 0.003 | 0.004 | 0.003 | 0.0023 |
| **Jitter (s)** | 0.0273 | 0.0235 | 0.003 | 0.0033 | 0.016 |
| **Third Hop** | **Video 33pkts/s** | **VoIP 100pkts/s** | **VoIP 50pkts/s** | **VoIP 35pkts/s** | **VoIP 25pkts/s** |
| **Packet Loss (%)** | 0.898 | 0.796 | 0.718 | 0.613 | 0.725 |
| **Delay (s)** | 0.0256 | 0.0046 | 0.006 | 0.0067 | 0.0087 |
| **Jitter (s)** | 0.145 | 0.215 | 0.158 | 0.172 | 0.049 |

**Table 15:** Linear Configuration of Test Bed at 11 Mbps

| **First Hop** | **Video 33pkts/s** | **VoIP 100pkts/s** | **VoIP 50pkts/s** | **VoIP 35pkts/s** | **VoIP 25pkts/s** |
|---|---|---|---|---|---|
| **Packet Loss (%)** | 0.37 | 0.25 | 0.41 | 0.508 | 0.355 |
| **Delay (s)** | 0.002 | 0.001 | 0.001 | 0.001 | 0.001 |
| **Jitter (s)** | 0.0103 | 0.061 | 0.0083 | 0.01 | 0.0077 |
| **Second Hop** | **Video 33pkts/s** | **VoIP 100pkts/s** | **VoIP 50pkts/s** | **VoIP 35pkts/s** | **VoIP 25pkts/s** |
| **Packet Loss (%)** | 0.64 | 0.47 | 0.415 | 0.635 | 0.489 |
| **Delay (s)** | 0.003 | 0.001 | 0.001 | 0.001 | 0.0013 |
| **Jitter (s)** | 0.027 | 0.0447 | 0.0083 | 0.01 | 0.0077 |
| **Third Hop** | **Video 33pkts/s** | **VoIP 100pkts/s** | **VoIP 50pkts/s** | **VoIP 35pkts/s** | **VoIP 25pkts/s** |
| **Packet Loss (%)** | 1.19 | 1.27 | 0.69 | 1.03 | 0.578 |
| **Delay (s)** | 0.013 | 0.003 | 0.003 | 0.004 | 0.005 |
| **Jitter (s)** | 0.016 | 0.046 | 0.019 | 0.01 | 0.0095 |

**Table 16:** Linear Configuration of Test Bed at 54 Mbps

| First Hop | Video 33pkts/s | VoIP 100pkts/s | VoIP 50pkts/s | VoIP 35pkts/s | VoIP 25pkts/s |
|---|---|---|---|---|---|
| Packet Loss (%) | 0.651 | 0.585 | 0.555 | 0.455 | 0.68 |
| Delay (s) | 0.001 | 3.33E–4 | 0.001 | 0.001 | 0.001 |
| Jitter (s) | 0.0063 | 0.0146 | 0.013 | 0.0093 | 0.0083 |
| **Second Hop** | **Video 33pkts/s** | **VoIP 100pkts/s** | **VoIP 50pkts/s** | **VoIP 35pkts/s** | **VoIP 25pkts/s** |
| Packet Loss (%) | 1.874 | 1.459 | 1.288 | 1.248 | 1.407 |
| Delay (s) | 0.0013 | 0.001 | 0.001 | 0.001 | 0.001 |
| Jitter (s) | 0.0119 | 0.0116 | 0.0103 | 0.009 | 0.0083 |
| **Third Hop** | **Video 33pkts/s** | **VoIP 100pkts/s** | **VoIP 50pkts/s** | **VoIP 35pkts/s** | **VoIP 25pkts/s** |
| Packet Loss (%) | 1.975 | 1.548 | 1.37 | 1.375 | 1.452 |
| Delay (s) | 0.013 | 0.0026 | 0.0033 | 0.004 | 0.0043 |
| Jitter (s) | 0.0106 | 0.0167 | 0.0103 | 0.009 | 0.0087 |

### 5.3.1 Multicast Multi-Hop Forwarding with Square Configuration, IEEE 802.11b

Next the SMF-OLSR was tested under square topology. The configuration is presented in Figure 15. The multicast sender and receiver are placed diagonally to each other. Adjacent nodes communicate under high signal quality but nodes placed diagonally cannot see each other. This might lead to packet collisions due to hidden node problem.

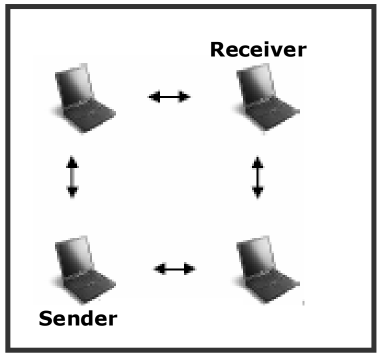


**Figure 15:** Square Test Bed configuration for multi-hop multicast experiment with hidden stations

Depending on the chosen algorithm for the selection of the MPR set, a different performance can be observed. Under S-MPR one of the two remaining nodes are sufficient to connect the sender with the receiver two-hop away. Therefore, no collisions are expected. On the other hand under NS-MPR all nodes can be relays even at the same time leading to simultaneous transmissions. The results for packet delivery rate and average end-to-end delay of video traffic at 11 Mbps for the three different algorithms are presented in Tables 18, 19, and 20. In Table 17 the results refer to the situation where the two nodes except sender and receiver are forced to be MPRs. Discussion is provided in the chapter 6.4.

**Table 17:** Performance of square topology and flooding

| Experiment | Total Packets | Packet Rate | Delay (s) | Jitter (s) |
|---|---|---|---|---|
| 1 | 47496 | 419.954 | 0.125 | 0.157 |
| 2 | 44970 | 397.653 | 0.125 | 0.157 |
| 3 | 63123 | 558.071 | 0.125 | 0.160 |
| 4 | 81594 | 721.451 | 0.125 | 0.157 |
| 5 | 76336 | 674.971 | 0.125 | 0.158 |
| **Average:** | **62704** | **554.4** | **0.125** | **0.158** |

**Table 18:** Performance of square topology and S-MPR SMF-OLSR

| Experiment | Total Packets | Packet Rate | Delay (s) | Jitter (s) |
|---|---:|---:|---:|---:|
| 1 | 88071 | 764.865 | 0.127 | 0.251 |
| 2 | 86354 | 749.851 | 0.127 | 0.243 |
| 3 | 88026 | 762.405 | 0.127 | 0.245 |
| **Average:** | **87484** | **759** | **0.127** | **0.246** |

**Table 19:** Performance of square topology and NS-MPR SMF-OLSR

| Experiment | Total Packets | Packet Rate | Delay (s) | Jitter (s) |
|---|---:|---:|---:|---:|
| 1 | 57125 | 475.275 | 0.152 | 4.699 |
| 3 | 72706 | 503.448 | 0.180 | 5.902 |
| 4 | 70682 | 364.181 | 0.286 | 7.904 |
| 5 | 57057 | 351.205 | 0.235 | 6.869 |
| **Average:** | **64392** | **423.5** | **0.21325** | **6.3435** |

**Table 20:** Performance of square topology and ECDS SMF-OLSR

| Experiment | Total Packets | Packet Rate | Delay (s) | Jitter (s) |
|---|---:|---:|---:|---:|
| 1 | 73751 | 525.686 | 0.195 | 6.875 |
| 2 | 85495 | 637.355 | 0.238 | 7.438 |
| **Average:** | **79623** | **581.5** | **0.2165** | **7.1565** |

# 6. MULTI-HOP MULTICAST ROUTING PROTOCOL ODMRP IN QUALNET

## 6.1 On-Demand Multicast Routing Protocol

In QualNet Simulator the libraries for wireless multicast routing protocols are quite limited. ODMRP is the only implemented protocol for wireless multicast. Other available protocols in QualNet like DVMRP, MOSPF, and PIM-DM work for wired networks only. ODMRP is designed for a single subnet wireless ad hoc multicast and it is a typical representative of mesh based protocols. It performs similarly to an on-demand unicast routing protocol (e.g. AODV) by applying source based on-demand procedures to build routes dynamically. It also uses the concept of forwarding group (FG) to reduce the forwarding overhead of data packets (only a subset of nodes is responsible for forwarding multicast data) and it uses a soft state approach in the maintenance of the multicast group.

The functionality of the protocol is described in detail in ODMRP IETF draft [7] and its performance under different network conditions is discussed in many studies [8,23-26]. It is out of scope of this thesis to come into details about the functionality of the protocol from algorithmic point of view. On the other hand it is mandatory to report the parameter values being used in our simulations since they affect the overall performance of the protocol. The parameter settings of ODMRP in the simulator are summarized in the following Table 21.

**Table 21:** Parameter values for ODMRP

| | |
|---|---|
| **Join Query Refresh Interval** | 3 s |
| **Forwarding Group Timeout** | 12 s |
| **Join Reply Acknowledgment Timeout** | 75 ms |
| **Maximum Join Reply Retransmissions** | 2 |

In the next paragraphs the performance of ODMRP in linear and square topology under various bitrates is considered. A comparison between SMF and ODMRP with respect to coverage, and performance limits under hidden node problem is presented.

### 6.2 Multicast Multi-Hop Forwarding with Linear Configuration, IEEE 802.11b/g

The configuration which was used for the linear topology of nodes in QualNet is depicted in Figure 16. Node $N_1$ is the multicast source and nodes $N_{2\text{-}10}$ are multicast receivers. Nodes $N_{2\text{-}9}$ become members of the FG.

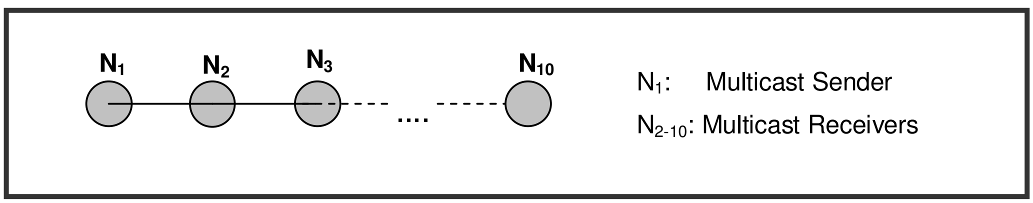


**Figure 16:** Node placement for the linear configuration

The simulation set up is similar to the test bed configuration. High link quality has been used between neighbors and the nodes are only within the range of their closest neighbors. The simulator provides the opportunity to carry out the relevant tests for larger number of nodes and wider range of bitrates. The simulations are carried out for a linear configuration of ten nodes at 2, 5.5, 11 and 54 Mbps. In the Table 22 the packet loss and average delay for each receiver are depicted for 64 Kbps VoIP traffic with 10 ms audio payload. In addition to the IP and UDP headers, each packet consists of an additional RTP overhead of twelve bytes. Therefore, the UDP payload of audio packets sent every 10 ms equals 92 bytes. More results with various audio data payload settings are provided in Appendix 3. It is noted that all the results refer to unidirectional data streams. For each kind of traffic 100 experiments with different seeds (randomizing) were executed and the results were averaged.

**Table 22:** 64 Kbps VoIP performance for linear configuration, IEEE 802.11b/g, G711 codec and 92 bytes per packet

| Multicast Receivers | $N_2$ | $N_3$ | $N_4$ | $N_5$ | $N_6$ | $N_7$ | $N_8$ | $N_9$ | $N_{10}$ |
|---|---|---|---|---|---|---|---|---|---|
| **Data Rate: 2 Mbps** | | | | | | | | | |
| **Packet Loss (%)** | 11.43 | 19.49 | 25.761 | 31.06 | 35.5 | 39.5 | 43.07 | 43.166 | 43.268 |
| **Delay (s)** | 0.00587 | 0.0118 | 0.0178 | 0.0297 | 0.029 | 0.035 | 0.0416 | 0.047 | 0.053 |
| **Data Rate: 5.5 Mbps** | | | | | | | | | |
| **Packet Loss (%)** | 5.42 | 9.93 | 13.84 | 17.355 | 20.51 | 23.41 | 26.17 | 26.237 | 26.248 |
| **Delay (s)** | 0.0054 | 0.0108 | 0.016 | 0.022 | 0.027 | 0.0327 | 0.038 | 0.0436 | 0.049 |
| **Data Rate: 11Mbps** | | | | | | | | | |
| **Packet Loss (%)** | 3.62 | 6.76 | 9.6 | 12.2 | 14.6 | 16.95 | 19.18 | 19.25 | 19.25 |
| **Delay (s)** | 0.0053 | 0.0106 | 0.016 | 0.0212 | 0.0265 | 0.032 | 0.037 | 0.042 | 0.047 |
| **Data Rate: 54Mbps** | | | | | | | | | |
| **Packet Loss (%)** | 0.855 | 1.653 | 2.4055 | 3.137 | 3.863 | 4.572 | 5.298 | 5.338 | 5.339 |
| **Delay (s)** | 0.0051 | 0.0102 | 0.0153 | 0.0204 | 0.0254 | 0.0305 | 0.0356 | 0.0407 | 0.0458 |

The results presented in the tables confirm some of the assumptions made beforehand. The packet delivery ratio deteriorates as the number of hops increases and the average end-to-end delay is increased proportionally from hop to hop. This behavior is observed for all the data rates and every data payload setting.

Both end-to-end delay and packet delivery performance are improved as the data rate increases since the probability of channel contention and collisions due to the hidden node problem is reduced.

The performance in terms of packet loss is greatly benefited by increasing the payload of the audio data packets since the bitrate stays the same. This means that there exists significantly less overhead when the packet delivery rate is lower with bigger payloads. The delays remain almost similar with very low growth toward the larger packet size. On the other hand, QoS constraints for the voice can require higher delivery rate and smaller payload size, which generally implies smaller bit error ratio. When audio data packets are sent every 40 ms, acceptable packet loss of 3.67 % is measured at the last hop with 2 Mbps bitrate. In this case the area covered by the ten node configuration is enlarged. On the other hand when the smallest data payload is employed the quality of service at the last hop is acceptable only at 54 Mbps and the coverage area is not that large.

As the next step the performance of the linear scheme is tested with video traffic. UDP payload of 1344 bytes is transmitted by the source node every 30 ms. The results are summarized in Table 23. It is observed that video traffic cannot be forwarded adequately even at the first hop when 2 Mbps bitrate is used. On the other hand when 11 Mbps or higher data rate is employed the video traffic meets the quality constraints (packet loss and average end-to-end delay) for all the receivers.

**Table 23:** Video Transmission performance for linear configuration, IEEE 802.11b/g

| Multicast Receivers | N2 | N3 | N4 | N5 | N6 | N7 | N8 | N9 | N10 |
|---|---|---|---|---|---|---|---|---|---|
| **Data Rate: 2 Mbps** | | | | | | | | | |
| **Packet Loss (%)** | 25.54 | 33.56 | 38.05 | 41.01 | 43.25 | 44.86 | 46.17 | 46.21 | 46.27 |
| **Delay (s)** | 0.0112 | 0.0224 | 0.0334 | 0.0443 | 0.0553 | 0.0662 | 0.0771 | 0.0879 | 0.0987 |
| **Data Rate: 5.5 Mbps** | | | | | | | | | |
| **Packet Loss (%)** | 1.39 | 3.85 | 6.77 | 9.73 | 12.54 | 15.14 | 17.53 | 17.56 | 17.56 |
| **Delay (s)** | 0.0072 | 0.0145 | 0.022 | 0.029 | 0.0367 | 0.044 | 0.0515 | 0.0588 | 0.066 |
| **Data Rate: 11Mbps** | | | | | | | | | |
| **Packet Loss (%)** | 0.4 | 1.21 | 2.387 | 3.82 | 5.37 | 6.95 | 8.55 | 8.576 | 8.578 |
| **Delay (s)** | 0.0061 | 0.012 | 0.0186 | 0.025 | 0.031 | 0.0375 | 0.044 | 0.05 | 0.056 |
| **Data Rate: 54Mbps** | | | | | | | | | |
| **Packet Loss (%)** | 0.0706 | 0.1846 | 0.3678 | 0.625 | 0.9285 | 1.2737 | 1.646 | 1.6561 | 1.657 |
| **Delay (s)** | 0.0052 | 0.0105 | 0.0158 | 0.0211 | 0.0264 | 0.0317 | 0.0369 | 0.0422 | 0.047 |

## 6.3 Multicast Multi-Hop Forwarding with Square Configuration, IEEE 802.11b/g

In this section the multicast multi-hop forwarding under ODMRP is presented with hidden node scenario for various bitrates, size of multicast group, and timer settings of the protocol. The tested network topology is depicted in figure 17. Node $N_1$ is the multicast sender and it can send packets to nodes $N_2$ and $N_4$ directly. Nodes $N_2$ and $N_4$ can forward traffic to node $N_3$ but they cannot hear each other. Nodes $N_2$ and $N_4$ can become members of the FG at the same time. In this way, performance degradation due to hidden terminal problem is observed at the multicast receiver $N_3$. The UDP payload size of the CBR traffic equals 1344 bytes. Packets are transmitted every 30 ms to emulate video transmission.

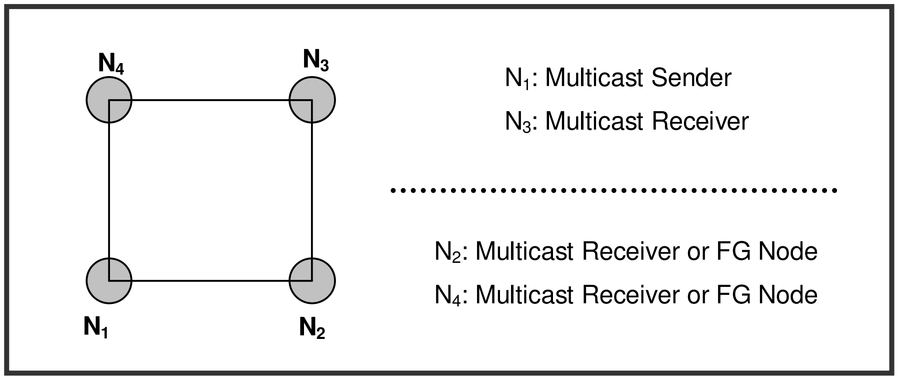


**Figure 17:** Node placement for square configuration

Two types of simulation experiments were conducted. In experiment type 1, nodes $N_2$, $N_3$ and $N_4$ are all multicast receivers. In experiment type 2 node $N_3$ is the only multicast receiver. In this case, nodes $N_2$ and $N_4$ can be selected only as data forwarders. Furthermore, two cases are studied regarding the value of the protocol timers. Firstly, a typical value is assigned to the forwarding group timeout (12 s) according to [23]. Secondly, the worst case value (500 s) is examined i.e. once a node becomes a forwarder it remains one until the end of the simulation. Results are presented in Tables 24, 25, 26, and 27. Relevant results for other bitrates of IEEE 802.11b/g are presented in the Appendix 4. All the values are averages over 100 runs.

**Table 24:** Packet loss and end-to-end delay at 11 Mbps with 12 s FG timeout

| Experiment | Feature | $N_2$ | $N_3$ | $N_4$ |
|---|---|---|---|---|
| **Type 1** | **Packet Loss (%)** | 0 | 15.628 | 0 |
| **Type 1** | **Delay (sec)** | 0.00623 | 0.011247 | 0.00623 |
| **Type 2** | **Packet Loss (%)** | × | 15.409 | × |
| **Type 2** | **Delay (sec)** | × | 0.011126 | × |

**Table 25:** Packet loss and end-to-end delay at 11 Mbps with 500 s FG timeout

| Experiment | Feature | $N_2$ | $N_3$ | $N_4$ |
|---|---|---|---|---|
| **Type 1** | **Packet Loss (%)** | 0 | 21.61 | 0 |
| **Type 1** | **Delay (sec)** | 0.006227 | 0.0105 | 0.006227 |
| **Type 2** | **Packet Loss (%)** | × | 21.325 | × |
| **Type 2** | **Delay (sec)** | × | 0.010418 | × |

**Table 26:** Packet loss and end-to-end Delay at 54 Mbps with 12 s FG timeout

| Experiment | Feature | $N_2$ | $N_3$ | $N_4$ |
|---|---|---|---|---|
| **Type 1** | **Packet Loss (%)** | 0 | 3.792 | 0 |
| **Type 1** | **Delay (sec)** | 0.005274 | 0.009194 | 0.005274 |
| **Type 2** | **Packet Loss (%)** | × | 3.792 | × |
| **Type 2** | **Delay (sec)** | × | 0.009158 | × |

**Table 27:** Packet loss and end-to-end delay at 54 Mbps and 500 s FG timeout

| Experiment | Feature | $N_2$ | $N_3$ | $N_4$ |
|---|---|---|---|---|
| **Type 1** | **Packet Loss (%)** | 0 | 4.665 | 0 |
| **Type 1** | **Delay (sec)** | 0.005277 | 0.008842 | 0.005277 |
| **Type 2** | **Packet Loss (%)** | × | 4.67 | × |
| **Type 2** | **Delay (sec)** | × | 0.008815 | × |

The results presented in the tables verify again some of the intuitional assumptions. In experiments of type 1 the packet loss at nodes $N_2$ and $N_4$ equals zero and the average delay of data packets at node $N_3$ is approximately double compared to the average delay at its upstream neighbors.

The packet delivery performance at node $N_3$ is degraded for lower data rates and larger forwarding group timeout values since the probability of collision is higher.

The overall performance at node $N_3$ is unrelated to the type of the experiment. Since nodes $N_2$ and $N_4$ can become forwarders whether they are multicast members or not the probability of collision at node $N_3$ remains the same at both experiments.

### 6.4 Performance Comparison between SMF and ODMRP, IEEE 802.11b/g

The performance comparison between ODMRP and SMF−OLSR for the linear and square topology is discussed in this section. In the linear configuration ODMRP and SMF-OLSR select the same MPR set. However, the packet loss measured in test bed and obtained by QualNet differs a lot for 2 Mbps. For video transmission the packet loss with ODMRP equals 25.54 % at the first hop while the corresponding packet loss with SMF-OLSR is only 0.2 %. Since there is just one multicast sender, the control overhead of ODMRP cannot justify this high difference. One but not significant reason for the better performance of SMF-OLSR is the higher UDP throughput of the test bed compared to QualNet. The most considerable reason causing the difference between SMF-OLSR and ODMRP is caused by the difference in the (average) per hop delay of data packets. For 2 Mbps and video communication the average per hop delay is 6 ms in the test bed. On the other hand the average per hop delay with ODMRP in QualNet equals 12 ms. The larger delay in data forwarding results in packet loss due to hidden node problem. Indeed, node $N_4$ receives its multicast traffic from node $N_3$ every 33 ms, while the multicast sender transmits every 30 ms. This is translated to clear collision interval and packet loss at node $N_2$ with ODMRP. On the other hand with SMF-OLSR, the short per hop delay rules out the possibility of collisions.

For 11 Mbps and 54 Mbps the results of the two routing protocols are much closer to each other. Actually, ODMRP performance is greatly improved even from 5.5 Mbps and starts being close to that of SMF-OLSR. It has been verified with simulations that the performance with ODMRP remains practically the same after 24 Mbps. Since the bitrate is already so high, the significant part of the delay is caused by other factors than the data transfer itself. Therefore also the loss stays approximately the same. In the test bed on the other hand, it was noticed that the links between the nodes were not as close to optimal and there was slightly increasing degradation with higher bitrates with more sensitive coding.

In the square configuration, the test bed with S-MPR resulted in no extra packet loss because packet collisions are avoided. Flooding and NS-MPR end up to quite similar throughputs because in case of flooding both nodes (except sender and multicast receiver) are selected for data forwarding and in case of NS-MPR every node can be a forwarder. The degradation was around 26 % compared to the ideal S-MPR case. Under ECDS the test bed performs better than under NS-MPR and worse than under S-MPR.

For ODMRP in QualNet the packet loss equals almost 22 % when the FG timeout interval equals 500 s. In this case the QualNet performance is comparable to the test bed performance under both flooding and NS-MPR because both remaining nodes become data forwarders at the same time. The packet loss for ODMRP is around 15 % when the FG timeout interval equals 12 s. By decreasing the duration of FG timeout the packet loss is decreased but the collisions due to hidden node problem cannot be eliminated. Summing up, the SMF-OLSR with S-MPR outperforms ODMRP in the square topology.

## 6.5 System Performance of ODMRP and SMF−OLSR with Unidirectional Links

The performance comparison between the multicast protocols employed by the Test-Bed and the simulator is continued in networks with unidirectional links. Since SMF is not available in QualNet, the testing is limited to unicast functionality of OLSR in the presence of unidirectional links. The selection of MPRs in OLSR is based on periodical HELLO message exchange and OLSR considers a link to be valid only if it is bidirectional. In other words, MPRs are selected among the one hop neighbors with a bidirectional link and transferring of data packets on unidirectional links is avoided [29]. On the other hand ODMRP does not have such a protection mechanism.

A topology for protocol performance investigation in a network with unidirectional links is depicted in Figure 18. Node $N_3$ does not see node $N_4$ as its one hop neighbor. In order to set up in QualNet the Network depicted in the Figure 18 below it is necessary to find a model for unidirectional links. One possible solution could be to use access lists. In this case, access lists filter all kinds of traffic that node $N_4$ receives from node $N_3$. However Node $N_4$ can receive any traffic from nodes $N_7$ and $N_1$. The access list file applied for the scenario is given in the following Table 28.

**Table 28:** Access list file

```
#Node 4 does not accept any UDP packets from Node 3
#
NODE-IDENTIFIER 4
INTERFACE 0
ACCESS-LIST 12 deny 192.0.0.3 0.0.0.0
ACCESS-LIST 12 permit any
IP ACCESS-GROUP 12 IN
```

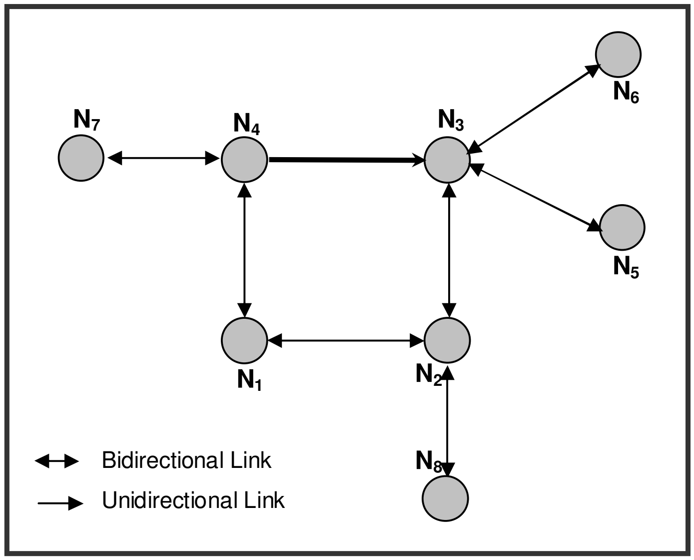


**Figure 18:** Node placement for network with unidirectional links

Node $N_7$ is the multicast source and nodes $N_5$ and $N_6$ are the multicast group receivers. Under ODMRP the join requests (JREQ) are processed through the shortest path ($N_7$-$N_4$-$N_3$-$N_{5/6}$) but all the join replies (JREP) are blocked at Node $N_3$. The alternate route to the source ($N_{6/5}$-$N_3$-$N_2$-$N_1$-$N_4$-$N_7$) is not found. Node $N_3$ broadcasts the JREPs originated from node $N_5$ or $N_6$ but the next hop ID in the JREP packet is node $N_4$ and not node $N_2$. Therefore node $N_2$ silently discards the JREPs forwarded by node $N_3$. At the same time node $N_4$ cannot become a forwarder since it is linked unidirectionally to node $N_3$ and cannot receive JREPs. The JREPs never reach the multicast source. Summing up, nodes $N_5$ and $N_6$ receive as many multicast data packets as the number of JREQs introduced by node $N_7$ because the JREQs are piggybacked with multicast data packets. The timer of forward group timeout does not affect the overall results.

The ODMRP can have a local route repair scheme as it is described in [23]. The recover mechanism for ODMRP is not available in QualNet. Next the behavior of ODMRP with unidirectional links was further investigated based on the chosen multicast receiving nodes. Some remarks confirmed by simulations are provided.

If nodes $N_1$, $N_5$ and $N_6$ are selected as multicast receiving nodes they all get most of the data packets. This happens because Node $N_7$ receives the JREPs originated by node $N_1$. Therefore node $N_7$ broadcasts regularly the multicast traffic which is forwarded to nodes $N_5$ and $N_6$ via the unidirectional link. In this scenario the performance improvement is unrelated to the capability of ODMRP to build mesh (multiple routes). Multiple routes from source to destinations are not established.

In case node $N_2$ is selected as multicast receiving node, nodes $N_1$, $N_5$ and $N_6$ still receive the multicast traffic with a high packet delivery ratio. In this case node $N_1$ could act as forwarder only as well and the situation would be the same. However, node $N_2$ suffers some packet loss due to the hidden node problem. Nodes $N_1$ and $N_3$ are unaware of each other's transmissions and collisions happen at node $N_2$. The average packet loss over 50 runs for the four multicast receivers is given in the following Table 29.

**Table 29:** Average packet loss for the four multicast receivers

| **Multicast Receivers** | **$N_1$** | **$N_2$** | **$N_5$** | **$N_6$** |
|---|---|---|---|---|
| **Packet Loss (%)** | 0.1845 | 19.138 | 0.205 | 0.205 |

As the next step, the behavior of ODMRP with respect to the number of multicast sources is investigated. The efficient building of mesh is now implemented.

It is assumed that node $N_8$ and $N_7$ generate multicast traffic to nodes $N_5$ and $N_6$. Nodes $N_5$, $N_6$, $N_7$ and $N_8$ belong to the same multicast group. Due to the multiple sources, mesh is formed. In particular, node $N_2$ stores in its routing table an entry for node $N_7$ because it receives non duplicate JREQs from that node and it receives JREPs as well. Node $N_2$ receives the JREPs from Node $N_3$ containing that the destination node for next hop for node $N_8$ is node $N_2$ and the destination node for next hop for node $N_7$ is node $N_4$. Unlike the case with one multicast source, node $N_2$ finds its IP address in one of the next hop addresses. Therefore it does not discard the JREPs. At the same time it has the routes for both multicast sources. To sum up, nodes $N_2$ and $N_3$ are forwarders for both sources while nodes $N_1$ and $N_4$ are forwarders only for the source $N_7$. The alternate route from sender $N_7$ to the receivers has been found. The average packet delivery ratio of the multicast traffic generated by nodes $N_7$ and $N_8$ is presented in the following Table 30 for the two receiving nodes.

**Table 30:** Average packet delivery ratio for the two multicast receivers

| **Multicast Receivers** | **Packet Delivery Ratio (%)** | |
|---|---|---|
| | **Source $N_7$** | **Source $N_8$** |
| **$N_5$** | 0.8124 | 0.8117 |
| **$N_6$** | 0.8124 | 0.8117 |

### 6.6 System Performance of ODMRP Protocol for 802.11b in QualNet

The scope of this chapter is to evaluate the multicast performance of ODMRP using QualNet 3.9 Simulator. A list of performance metrics is presented in 6.6.1 and a catalog of appropriate test case scenarios is presented in 6.7. All the simulations share some common setup parameters which are discussed in 6.6.2. The performance of the protocol is evaluated under various network conditions. Different mobility attribute, various numbers of senders and multicast group members, and different network traffic loads are investigated.

#### 6.6.1 Fixed Parameter Settings for Multicast Protocol Simulation

The simulation scenarios include protocol testing with IEEE 802.11b. The data rates being tested are 11 Mbps and 2 Mbps. The transmission range for both bitrates is easily obtained by executing point-to-point simulations. The transmission range equals 380 m at 2 Mbps and 310 m at 11 Mbps. High link quality has been ensured for the above calculated transmission ranges meaning that all packets are received without errors.

If N nodes are placed within an area of size S ($m^2$) and the transmission range of each node is R (m), the expected number of neighbors per node is [30]:

$$m = \pi \frac{R^2}{S^2} N \tag{6.1}$$

Simulation scenarios comprise of forty nodes placed randomly. Based on the transmission range the dimensions of the area are chosen to give ten neighbors per node. This is translated to a terrain of 1350×1350 $m^2$ at 2 Mbps and 1100×1100 $m^2$ at 11 Mbps. The size of the area should be chosen carefully because the density of nodes affects the efficiency and the effectiveness of the protocol. Namely, more dense topologies result in higher data throughputs at the expense of larger control overhead due to the mesh nature of the protocol. The free space path loss (attenuation proportional to $1/r^2$) for short distances and two-ray path loss (attenuation proportional to $1/r^4$) for distances larger than a critical point (breakpoint) is used. The value of the critical distance depends upon the wavelength and the antenna heights of the transmitter and the receiver.

For the mobility study, the random waypoint mobility model was used. With this model the number of broken links increases and the mesh based nature of the protocol becomes more evident. There are several parameters to be defined for the random waypoint mobility model. In [23] the maximum mobility speed is varied between 0 m/s and 20 m/s and in [25] the fraction of moving nodes varies. In [23] a highly mobile network is represented with 0 s pause time and a low mobility scenario is emulated with a high value of pause time. In our simulations the latter option is used. Random waypoint mobility with speeds uniformly distributed between 0 m/s and 20 m/s are applied. The pause time varies from 0 s to 500 s. According to [31] one has to be extremely careful about the parameter settings of RWP mobility model and especially for the minimum velocity. For instance when the minimum node speed equals zero more and more nodes would be trapped to low speeds and the average instantaneous network velocity would be gradually decreased. As a result the calculation of performance metrics in the form of an average over an arbitrary set of time would be inaccurate.

The simulator uses IEEE 802.11 model for the MAC layer. Control messages and data packets are broadcasted under ODMRP but there are unicast transmissions as well like the active ACKs. To avoid the use of virtual carrier sensing at the MAC, the RTS threshold is set sufficiently high. Furthermore, constant bitrate UDP traffic is used for the data in order to emulate video transmission. The same packet size (1344 bytes) is used in the test bed experiments and the simulations. Unlike proactive routing protocols the on-demand nature of ODMRP permits the generation of traffic from the beginning of the simulation. The simulation time is fixed to 500 seconds and the generation of UDP traffic starts at zero seconds in all the scenarios.

### 6.7 Simulation Scenarios

Different tests are carried out in order to investigate the performance of the protocol under various network conditions. In particular, the reliability of the protocol is tested with respect to mobility, the number of senders, the multicast group size and the network traffic load. Additionally, the control overhead of the protocol is evaluated for different networks and protocol timer settings. The simulation studies are categorized according to the performance metrics and the network conditions. For every category the simulation scenario is described in detail and the relevant results are provided.

#### 6.7.1 Effectiveness versus Mobility and Different Number of Multicast Sources

Forty nodes are placed randomly in the specified area. Ten multicast members are selected randomly among the nodes. The number of multicast senders is 1, 2, or 5. In QualNet, a multicast sender must belong to its multicast group and thus a node can be both sender and receiver simultaneously. In the following simulations all the members form a single multicast group. All the multicast members join their group from the beginning of the simulation and remain members of the group until the end of the simulation. The network traffic load is kept relatively light (10 packets/s). This is translated to 10 packets/s/sender when one multicast source exists and 2 packet/s/sender when five sources are employed.

For every mobility pattern, ten runs are executed and the results are averaged. The seed value can also be changed among the different runs to introduce more randomness. Nevertheless, constant seed value equal to 1 is used in all the simulations. The packet delivery ratio versus mobility is plotted for 2 Mbps in Figure 19 and 11 Mbps in Figure 20. It is expected that the network performs more reliable and faster under the highest data rate at the expense of reduced size of the area covered by the nodes.

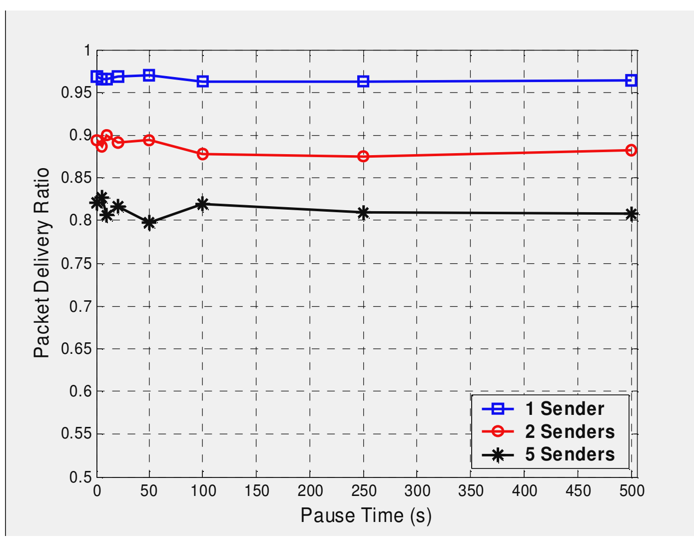


**Figure 19:** Packet delivery ratio for the 40-node model and various number of multicast senders versus mobility at 2 Mbps

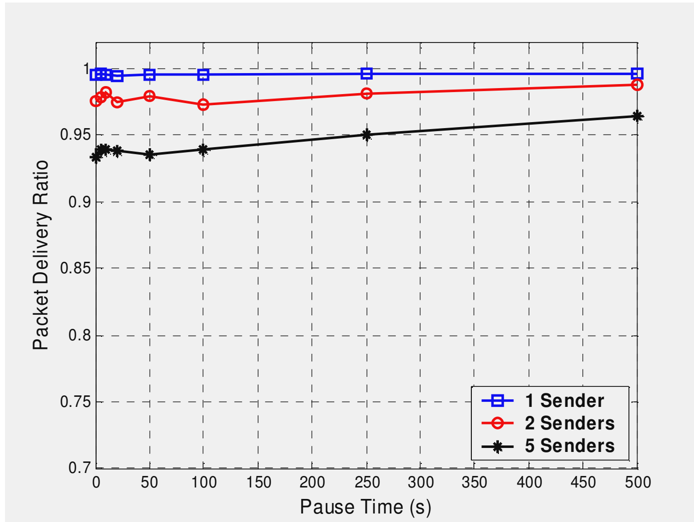


**Figure 20:** Packet delivery ratio for the 40-node model and various number of senders versus mobility at 11 Mbps

The simulations show that ODMRP behaves effectively even in highly dynamic conditions. It shows constant throughput characteristics throughout the range of simulated mobility. The variation is within few percents for a certain amount of senders. The mesh topology of the forwarding group provides multiple paths. The chances of packet delivery remain high even when the primary links are broken.

Furthermore, the performance of the protocol degrades as the number of multicast sources increases. The number of transmitted control packets within the network is proportionally increased with the number of senders and the amount of forwarded data packets is also increased. This causes packet loss due to network congestion and channel contention. The performance degradation is more severe for the lower bitrates since the probabilities of collisions and congestion are higher. Important factor is the delay caused by the data processing in a node. It is known that the delay is significantly longer with 2 Mbps than with 11 Mbps. Therefore in Figure 19 for the datarate of 2 Mbps, the processing delay per node is actually considerably affecting the performance and on the other hand in Figure 20 for the datarate of 11 Mbps, the effects of mobility are more apparent. In particular, the simultaneous action of five senders at 2 Mbps reduces the average packet delivery ratio to 80 % even if the network traffic load is small.

### 6.7.2 Effectiveness versus Scalability

In this experiment, the number of multicast senders was fixed to 2, the network traffic load is kept light (10 packets/s) and the size of multicast group varies from 5 to 40 nodes. Three scenarios with low, moderate and high mobility were executed. It is expected that the mobility level does not affect the results significantly and that the size of multicast group does not impact the performance. The packet delivery ratio versus the size of the multicast group is depicted for both bitrates in Figures 21 and 22. Each point in the following graphs is an average over ten runs.

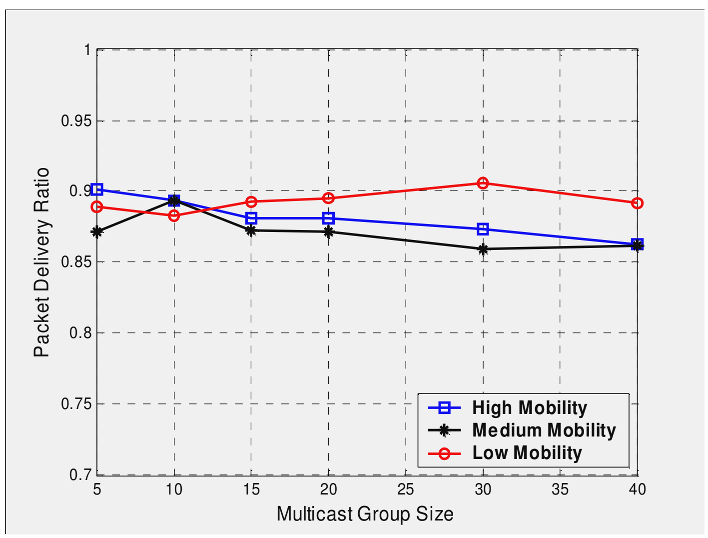


**Figure 21:** Packet delivery ratio for the 40-node model and different mobility versus multicast group size at 2 Mbps

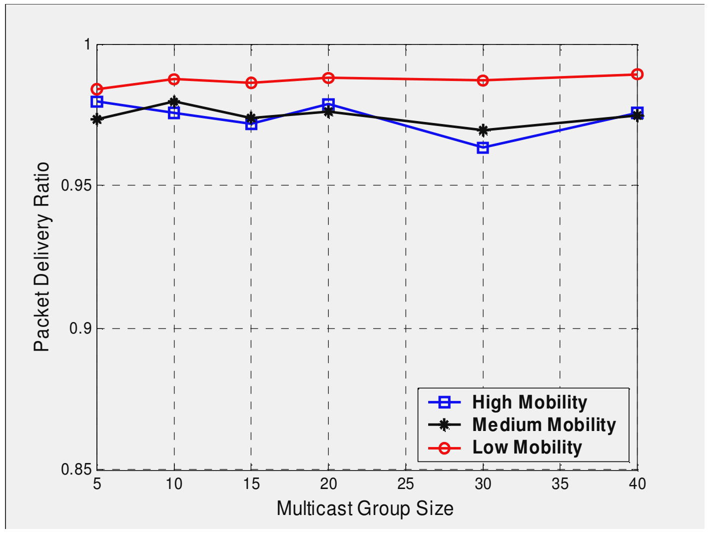


**Figure 22:** Packet delivery ratio for the 40-node model and different mobility versus multicast group size at 11 Mbps

The simulations indicate the robustness of the protocol against the size of the multicast group for different mobility levels. The performance remains almost the same with mobility, but like in the previous chapter 6.7.1, the affect of mobility can be observed more clearly with the higher datarate since the processing delay per node is considerably lower. The differences are not very significant and only 3% performance improvement is achieved for the low mobile network. As the number of group members increase the amount of forwarders increases as well. Since the mesh becomes denser, more redundant paths are created and packets are delivered even to receivers located far away. The formation of redundant paths provides resilience to mobility for the network as well. The fact that the results for medium and high mobility are very similar and the difference is greater from the low mobility is caused by the characteristics of random waypoint model and chosen pause time (50 s) for the medium mobility. With the pause time of 50 s, the expected number of neighbors and average velocity are clearly closer to the case of the high mobility (0 s pause time) compared to the low mobility where the nodes don't move anymore after stopping the first time during the simulation run.

### 6.7.3 Effectiveness versus Network Traffic Load

The protocol effectiveness under variable traffic load is tested next. The number of multicast senders is fixed to 2 and the multicast group is fixed to 10 nodes. The network traffic load varies between 5 packets/s and 66 packets/s. This corresponds to a load fluctuation between 2.5 packets/s/sender and 33 packets/s/sender. The heaviest traffic load corresponds to two video communication flows among the multicast group. The averaged results for packet delivery ratio are plotted below. The effectiveness was tested also for different mobility levels.

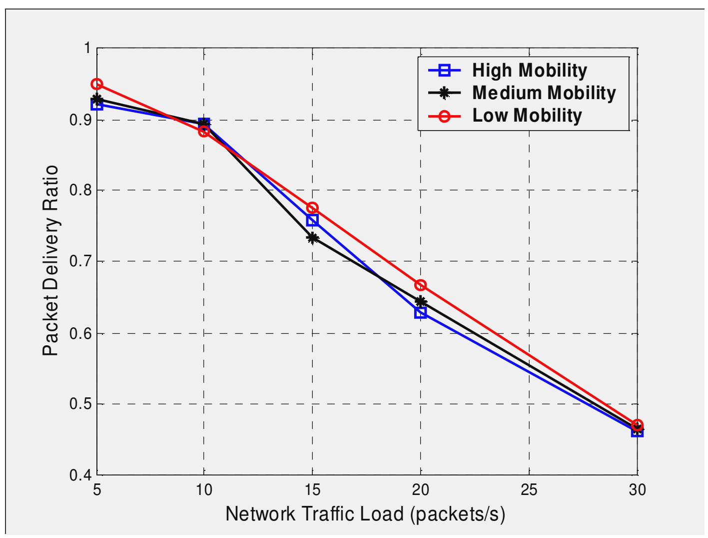


**Figure 23:** Packet delivery ratio for the 40-node model and different mobility vs. network load at 2 Mbps

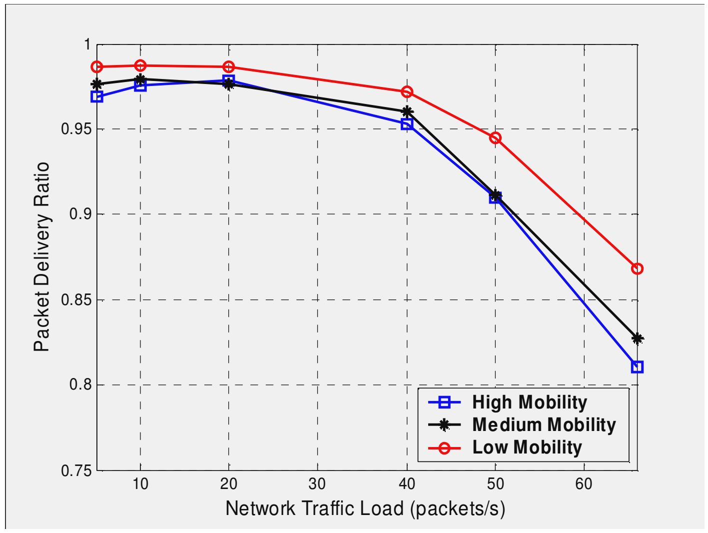


**Figure 24:** Packet delivery ratio for the 40-node model and different mobility vs. network load at 11 Mbps

The Figures 23 and 24 demonstrate that the packet loss remains almost the same for different mobility schemes indicating that the increase in load affects the number of collisions and congestion in the network. Although the number of senders is only 2, the degradation in protocol effectiveness is very severe for the lowest bitrate of 2 Mbps. The packet delivery drops below 50 % even for a network load of 30 packets/s. On the other hand it is remarkable that at 11 Mbps the video transmission is successful from packet delivery point of view.

### 6.7.4 Average End-to-End Delay with Scalability and Network Load under Mobility

The results of the average end-to-end delay of data packets for all the simulation scenarios developed from 6.7.1–6.7.3 are presented and discussed. The superiority of the 11 Mbps data rate in all test cases is worth pointing out.

Table 31 indicates that the increasing the number of multicast senders deteriorates the average delay. It has already been discussed in 6.7.1 that the overhead of control and data forwarding is increased with the number of senders. The performance degradation is more severe for the 2 Mbps bitrate.

The scalability of multicast receivers may not affect the packet delivery ratio but it has an impact on the average delay of the packets instead. When the group size changes from 5 to 40 nodes Table 32 shows that the delay grows from 65.81 ms to 144.33 ms. The average delay is proportional to the average number of hops. In a larger multicast group there are more nodes located many hops away from the senders resulting in higher delay values.

Table 33 indicates that the simultaneous transmission of two video flows within a multicast group of ten nodes at 11 Mbps meets the delay constraints. On the other hand the average end-to-end delay of data packets at 2 Mbps is growing enormously as the network load increases.

**Table 31:** Average end-to-end delay for different number of multicast senders and mobility

| Data Rate | Senders | Pause Time (s) | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|
| | | 0 | 5 | 10 | 20 | 50 | 100 | 250 | 500 |
| | | Average End-to-End Delay (ms) | | | | | | | |
| 2 Mbps | 1 | 27.3 | 28.8 | 31 | 28.6 | 29.6 | 31.1 | 34.4 | 36.6 |
| | 2 | 79.1 | 74.1 | 74.2 | 67.5 | 69.6 | 75.3 | 73.3 | 77.9 |
| | 5 | 457 | 360 | 339 | 300 | 280 | 250 | 199 | 221 |
| 11 Mbps | 1 | 11.6 | 11.4 | 11.6 | 11.8 | 12.3 | 12.4 | 13.2 | 12.8 |
| | 2 | 16.7 | 16.2 | 16.6 | 16.9 | 18.8 | 18.2 | 19.5 | 18.8 |
| | 5 | 39.3 | 39 | 39.7 | 38.9 | 39.4 | 40.2 | 40.9 | 40.9 |

**Table 32:** Average end-to-end delay for different multicast group size

| Mobility | Data Rate | Multicast Group Size (Nodes) | | | | | |
|---|---|---|---|---|---|---|---|
| | | 5 | 10 | 15 | 20 | 30 | 40 |
| | | Average End-to-End Delay (ms) | | | | | |
| Low | 2 Mbps | 65.81 | 77.86 | 85.69 | 94.32 | 116.23 | 144.33 |
| | 11 Mbps | 19.72 | 18.85 | 19.73 | 19.42 | 21 | 20.59 |
| Medium | 2 Mbps | 59.14 | 69.64 | 88.01 | 100.9 | 159.38 | 288.6 |
| | 11 Mbps | 16.26 | 18.81 | 18.89 | 18.61 | 19.97 | 19.85 |
| High | 2 Mbps | 56.67 | 79.1 | 99.1 | 145.4 | 217.8 | 451.4 |
| | 11 Mbps | 14.71 | 16.77 | 17.2 | 17.75 | 18.44 | 18.65 |

**Table 33:** Average end-to-end delay for different network traffic loads

| Mobility | Data Rate | Network Traffic Load (packets/s) | | | | | |
|---|---|---|---|---|---|---|---|
| | | 5 | 10 | 20 | 40 | 50 | 66 |
| | | Average End-to-End Delay (ms) | | | | | |
| Low | 2 Mbps | 66.24 | 77.86 | 1460 | × | × | × |
| | 11 Mbps | 19.23 | 18.85 | 19.46 | 23.97 | 61.9 | 377.12 |
| Medium | 2 Mbps | 60.52 | 69.64 | 1735 | × | × | × |
| | 11 Mbps | 17.35 | 18.81 | 18.32 | 20.92 | 66.32 | 437.7 |
| High | 2 Mbps | 62 | 79.1 | 1960 | × | × | × |
| | 11 Mbps | 16.84 | 16.77 | 17.1 | 21.59 | 60.71 | 415 |

### 6.7.5 Control Overhead with Mobility, Number of Senders, and Multicast Group Size

The amount of control overhead to data payload is estimated for different number of multicast senders and multicast group members. In control overhead the JREQs (44 bytes), the transmissions and retransmissions of JREPs (36 bytes), the active ACKs (32 bytes), the MAC layer ACKs (20 bytes) and the headers (62 bytes) of all the data packets transmitted and forwarded are considered. The control byte overhead divided by the total number of delivered data bytes shows how efficiently control packets are utilized in delivering data. Figure 25 and Figure 26 below present the dependence of control overhead on mobility and different number of senders. The simulation setup is similar to the one presented in 6.7.1.

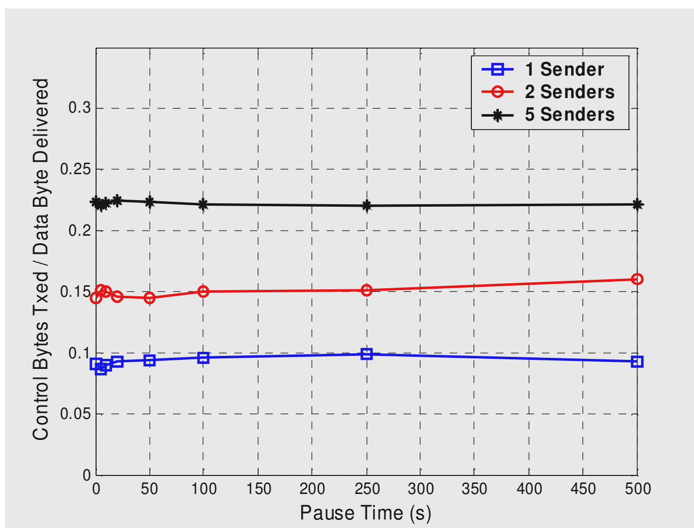


**Figure 25:** Control bytes transmitted per data byte delivered versus mobility for different number of senders, 2 Mbps

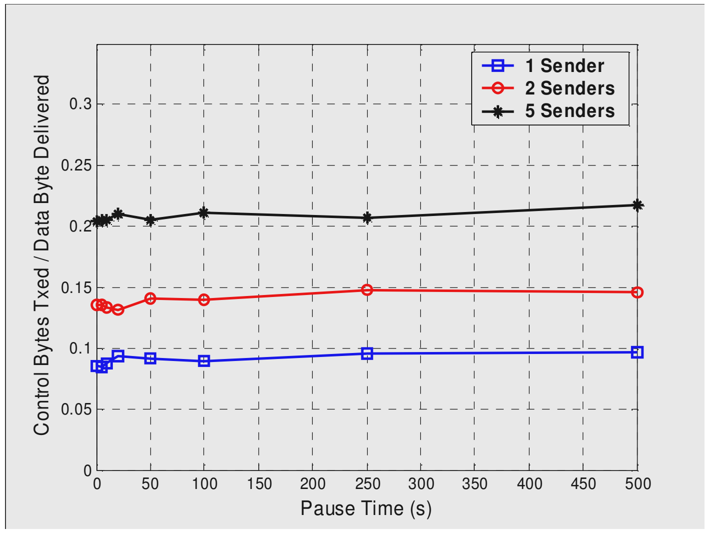


**Figure 26:** Control bytes transmitted per data byte delivered versus mobility for different number of senders, 11 Mbps

The results show that the control overhead remains almost constant with mobility since no updates or very few are caused by link breaks. The characteristic redundancy of routes with ODMRP renders new route updates unnecessary most of the time. The JREQ interval is kept constant at 3 s in all simulations and thus no additional overhead is produced as mobility increases. On the other hand it is evident that the control overhead scale with the number of senders since ODMRP builds per source mesh. If the number of senders increases, more JREQs are broadcasted into the network, more data packets are forwarded and the control overhead grows accordingly [23]. It can now be assumed that in a network with many multicast sources ODMRP becomes less effective, less efficient and introduces higher delays due to the augmented overhead produced. As the bitrate increases the effect of large overhead becomes less severe.

Now the simulation tests presented in 6.7.2 are considered and the behavior of control overhead as the number of group members increases is investigated. This is depicted in Figure 27 and Figure 28 below. It is expected that the scheme becomes more effective as the multicast group size grows since fewer data transmissions ends to be redundant.

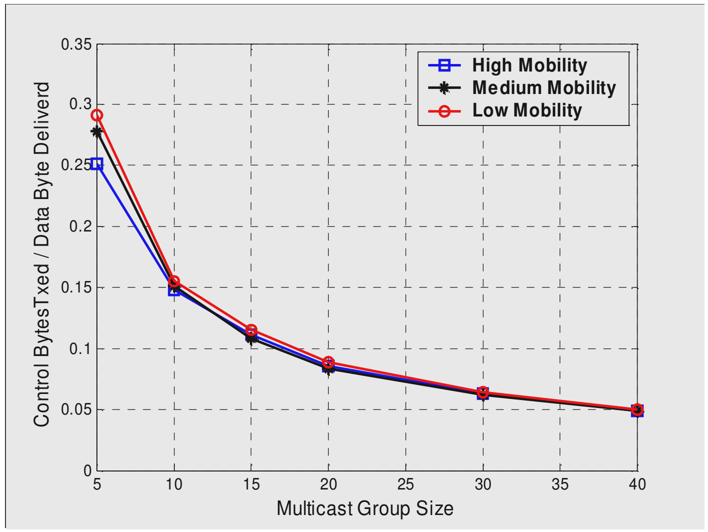


**Figure 27:** Control bytes transmitted per data byte delivered versus group size for different mobility, 2 Mbps

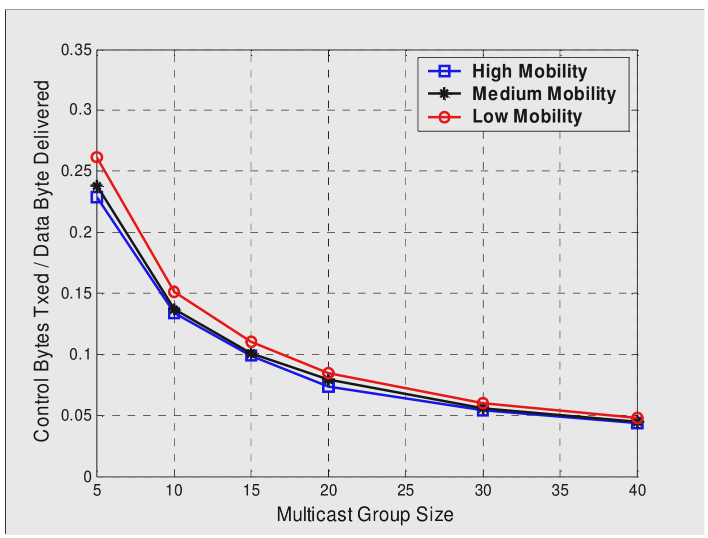


**Figure 28:** Control bytes transmitted per data byte delivered versus group size for different mobility, 11 Mbps

The results confirm that the efficiency of the protocol increases as the multicast group enlarges. ODMRP appears to be suitable in large multicast groups with few multicast senders. Another remark is that the protocol seems less efficient under low mobility conditions. This is the case because less links are broken under low mobility and the nodes remains forwarders during the specified group timeout interval. Therefore, they contribute to redundant forwarding of data packets. As the group size grows the difference in efficiency between different mobility schemes decreases.

So far in control overhead calculation the bytes of control packets and the headers of data packets are included. For clarification purposes, two more metrics are defined to assess the protocol efficiency. They allow evaluating separately the efficiency of the protocol at the control and data phase in terms of packets.

The control phase is observed every 3 s in the simulations and it serves for route discovery and route update. In control phase only the JREQ, JREP, passive ACK and MAC layer ACK are considered. The normalized control overhead is defined as the number of control packets issued divided by the total number of delivered data packets.

On the other hand the data phase covers every individual transmission of data throughout the simulation time. The normalized forwarding overhead is defined as the number of data packets transmitted per data packets delivered [23,31]. It is noted that the value of normalized forwarding can be less than one because a simple transmission in multicast can lead to multiple receptions. On the other hand, this metric is always equal or grater than one in unicast transmissions.

The dependence of the two metrics to multicast group size at 2 Mbps is plotted in the Figures 29 and 30 below.

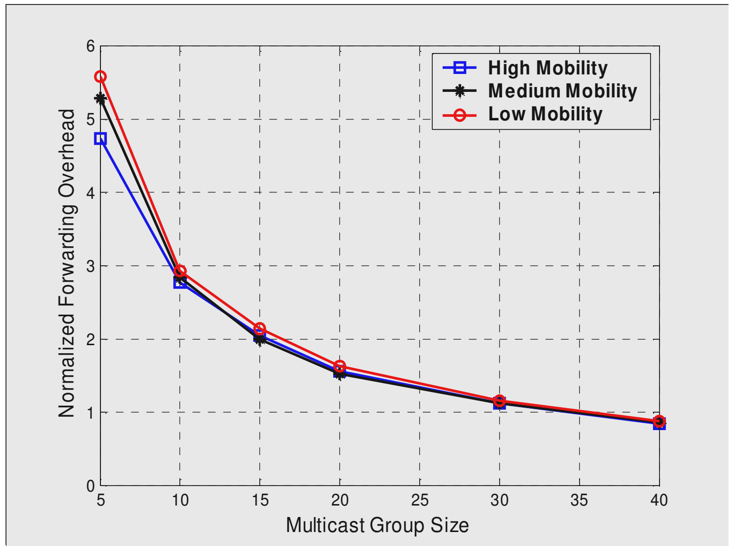


**Figure 29:** Normalized forwarding overhead versus group size, 2 Mbps

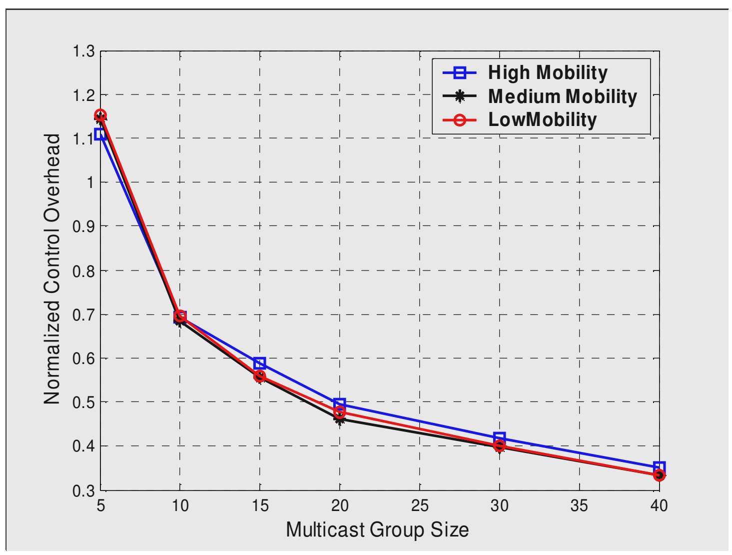


**Figure 30:** Normalized control overhead versus group size, 2 Mbps

It is observed that the system is more efficient during data phase and less efficient during control phase with high mobility. In a highly dynamic network more links are broken and more nodes stop forwarding data packets before the protocol timer is expired. This leads to smaller forwarding overhead and better efficiency in the data phase. On the other hand, more JREPs need to be retransmitted since the number of passive and active ACKs is reduced due to same above mentioned broken links. This leads to larger overhead and worse efficiency in the control phase.

# 7. CONCLUSIONS

Test bed experiments and simulation studies for multicast applications are the main topics in this thesis. The PHY and MAC layer parameters of the test bed had to be derived experimentally. For this purpose the maximum throughput of the test bed was measured for different bitrates of IEEE 802.11b/g. Then the maximum throughput with the same PHY and MAC layer parameters were measured in QualNet. The throughput results indicated that QualNet performs very close to the expected results. However, the test bed was found to outperform the simulator. The reason for this behavior was not cleared since any official documentation to confirm the assumptions was not found.

As the next step the performance of SMF was investigated in small scale scenarios. It was found out that choosing the algorithm for MPR set has significant effect on performance. S-MPR algorithm seemed to function best in small-scale scenarios. Nevertheless, it can be debated whether the more simple and robust classical flooding is actually a better choice, since it is not dependable on unicast routing at all and in small-scale scenarios the congestion might not become a problem. This is further studied in real field trial among other tests.

Then the performance of a typical mesh based routing protocol ODMRP was tested with simulations. The simulations were executed for a large-scale scenario of forty nodes and for IEEE 802.11b. The simulation results pointed out the good performance of ODMRP as the mobility level and the size of multicast group increases. On the other hand it was illustrated that ODMRP is not that efficient in networks where a large number of nodes are multicast senders. The performance of ODMRP was also found to be dependent on the bitrate. By using 11 Mbps the transmission of two simultaneous video flows within the same multicast group met the basic QoS constraints.

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## Appendix 1. Test Bed Throughput Results

**Table 1.1:** Transmitter throughput for IEEE 802.11b@11 Mbps without AP

| **UDP Payload (Bytes)** | **6** | **50** | **100** | **151** | **199** | **300** |
|---|---|---|---|---|---|---|
| **Packets/sec** | 4900 | 4610 | 4450 | 4298 | 4170 | 3932 |
| **UDP Payload (Bytes)** | **400** | **500** | **600** | **700** | **800** | **900** |
| **Packets/sec** | 3704 | 3517 | 3301 | 3181 | 3033 | 2903 |
| **UDP Payload (Bytes)** | **1000** | **1099** | **1200** | **1300** | **1400** | **1472** |
| **Packets/sec** | 2774 | 2670 | 2556 | 2458 | 2370 | 2308 |

**Table 1.2:** Transmitter throughput for IEEE 802.11g@54 Mbps without AP

| **UDP Payload (Bytes)** | **6** | **50** | **100** | **150** | **200** | **300** |
|---|---|---|---|---|---|---|
| **Packets/sec** | 1700 | 1627 | 1537 | 1443 | 1380 | 1225 |
| **UDP Payload (Bytes)** | **400** | **500** | **601** | **700** | **800** | **900** |
| **Packets/sec** | 1146 | 1057 | 983 | 914 | 859 | 807 |
| **UDP Payload (Bytes)** | **1000** | **1100** | **1200** | **1301** | **1400** | **1472** |
| **Packets/sec** | 757 | 721 | 685 | 651 | 620 | 600 |

**Table 1.3:** Transmitter throughput for IEEE 802.11b@1 Mbps with AP

| **UDP Payload (Bytes)** | **16** | **50** | **100** | **150** | **200** | **300** |
|---|---|---|---|---|---|---|
| **Packets/sec** | 780 | 642 | 508 | 422 | 357 | 277 |
| **UDP Payload (Bytes)** | **400** | **500** | **601** | **700** | **800** | **900** |
| **Packets/sec** | 224 | 190 | 165 | 145 | 130 | 116 |
| **UDP Payload (Bytes)** | **1000** | **1100** | **1200** | **1301** | **1400** | **1472** |
| **Packets/sec** | 107 | 98 | 91 | 84 | 79 | 75 |

**Table 1.4:** Transmitter throughput for IEEE 802.11b@11 Mbps with AP

| **UDP Payload (Bytes)** | **16** | **50** | **100** | **150** | **200** | **300** |
|---|---|---|---|---|---|---|
| **Packets/sec** | 1675 | 1605 | 1514 | 1428 | 1361 | 1238 |
| **UDP Payload (Bytes)** | **400** | **500** | **601** | **700** | **800** | **900** |
| **Packets/sec** | 1121 | 1046 | 970 | 890 | 848 | 796 |
| **UDP Payload (Bytes)** | **1000** | **1100** | **1200** | **1301** | **1400** | **1472** |
| **Packets/sec** | 749 | 706 | 673 | 641 | 610 | 591 |

**Table 1.5:** Transmitter throughput for IEEE 802.11g@54 Mbps with AP

| **UDP Payload (Bytes)** | **16** | **50** | **100** | **151** | **199** | **300** |
|---|---|---|---|---|---|---|
| **Packets/sec** | 6500 | 6420 | 6090 | 5710 | 5480 | 5215 |
| **UDP Payload (Bytes)** | **400** | **500** | **600** | **700** | **800** | **900** |
| **Packets/sec** | 4730 | 4460 | 4140 | 3890 | 3650 | 3410 |
| **UDP Payload (Bytes)** | **1000** | **1099** | **1200** | **1300** | **1400** | **1472** |
| **Packets/sec** | 3270 | 3090 | 2960 | 2820 | 2685 | 2610 |

## Appendix 2. QualNet Simulator Throughput Results

**Table 2.1:** Transmitter throughput for IEEE 802.11b@1 Mbps without AP

| **UDP Payload (Bytes)** | **16** | **50** | **100** | **151** | **199** | **300** |
|---|---|---|---|---|---|---|
| **Packets/sec** | 798.4 | 655.7 | 519.7 | 428.5 | 368.1 | 283.7 |
| **UDP Payload (Bytes)** | **400** | **500** | **600** | **700** | **800** | **900** |
| **Packets/sec** | 231.3 | 195 | 168.7 | 148.6 | 132.8 | 120 |
| **UDP Payload (Bytes)** | **1000** | **1099** | **1200** | **1300** | **1400** | **1472** |
| **Packets/sec** | 109.4 | 100.7 | 93.1 | 86.7 | 81 | 77.4 |

**Table 2.2:** Transmitter throughput for IEEE 802.11b@11 Mbps without AP

| **UDP Payload (Bytes)** | **16** | **50** | **100** | **151** | **199** | **300** |
|---|---|---|---|---|---|---|
| **Packets/sec** | 1589.2 | 1528.9 | 1449.7 | 1375.9 | 1311.7 | 1198.1 |
| **UDP Payload (Bytes)** | **400** | **500** | **601** | **700** | **800** | **900** |
| **Packets/sec** | 1101.8 | 1020.2 | 949.6 | 889 | 834.6 | 786.8 |
| **UDP Payload (Bytes)** | **1000** | **1100** | **1200** | **1300** | **1400** | **1472** |
| **Packets/sec** | 744.5 | 706.5 | 671.3 | 639.8 | 611.7 | 592.5 |

**Table 2.3:** Transmitter throughput for IEEE 802.11g@54 Mbps without AP

| **UDP Payload (Bytes)** | **16** | **50** | **100** | **151** | **199** | **300** |
|---|---|---|---|---|---|---|
| **Packets/sec** | 5744 | 5493 | 5376.3 | 5154.8 | 4949.9 | 4587 |
| **UDP Payload (Bytes)** | **400** | **500** | **600** | **700** | **800** | **900** |
| **Packets/sec** | 4346.3 | 4064 | 3815.4 | 3595.8 | 3446.1 | 3265.7 |
| **UDP Payload (Bytes)** | **1000** | **1099** | **1200** | **1300** | **1400** | **1472** |
| **Packets/sec** | 3104 | 2992.1 | 2855.7 | 2730.9 | 2616.1 | 2563.2 |

**Table 2.4:** Transmitter throughput for IEEE 802.11b@1 Mbps with AP

| **UDP Payload (Bytes)** | **16** | **50** | **100** | **150** | **200** | **300** |
|---|---|---|---|---|---|---|
| **Packets/sec** | 791.5 | 650.1 | 514.7 | 424.7 | 364.8 | 281.1 |
| **UDP Payload (Bytes)** | **400** | **500** | **601** | **700** | **800** | **900** |
| **Packets/sec** | 229.2 | 193.3 | 167.1 | 147.2 | 131.5 | 118.9 |
| **UDP Payload (Bytes)** | **1000** | **1100** | **1200** | **1301** | **1400** | **1472** |
| **Packets/sec** | 108.4 | 99.8 | 92.3 | 85.9 | 80.3 | 76.7 |

**Table 2.5:** Transmitter throughput for IEEE 802.11b@11 Mbps with AP

| **UDP Payload (Bytes)** | **16** | **50** | **100** | **151** | **199** | **300** |
|---|---|---|---|---|---|---|
| **Packets/sec** | 1584.8 | 1524.2 | 1445.3 | 1371.6 | 1307.9 | 1194.6 |
| **UDP Payload (Bytes)** | **400** | **500** | **601** | **700** | **800** | **900** |
| **Packets/sec** | 1098.5 | 1016.9 | 946.7 | 886.2 | 831.9 | 784.1 |
| **UDP Payload (Bytes)** | **1000** | **1100** | **1200** | **1300** | **1400** | **1472** |
| **Packets/sec** | 742.2 | 704.1 | 669.1 | 637.7 | 609.7 | 590.5 |

**Table 2.6:** Transmitter throughput for IEEE 802.11g@54 Mbps with AP

| **UDP Payload (Bytes)** | **16** | **50** | **100** | **151** | **199** | **300** |
|---|---|---|---|---|---|---|
| **Packets/sec** | 5722.1 | 5471.9 | 5355.2 | 5135.3 | 4931.5 | 4570.3 |
| **UDP Payload (Bytes)** | **400** | **500** | **600** | **700** | **800** | **900** |
| **Packets/sec** | 4330.9 | 4050.1 | 3803.1 | 3584 | 3434.8 | 3255.7 |
| **UDP Payload (Bytes)** | **1000** | **1099** | **1200** | **1300** | **1400** | **1472** |
| **Packets/sec** | 3094.5 | 2983.2 | 2847.3 | 2723 | 2608.9 | 2555.8 |

# Appendix 3. Simulation Results for VoIP Performance with ODMRP

**Table 3.1:** VoIP 64 Kbps Performance metrics for linear configuration of nodes for IEEE 802.11b/g. G711 codec and 172 bytes per packet

| Multicast Receivers | $N_2$ | $N_3$ | $N_4$ | $N_5$ | $N_6$ | $N_7$ | $N_8$ | $N_9$ | $N_{10}$ |
|---|---|---|---|---|---|---|---|---|---|
| **Data Rate: 2 Mbps** | | | | | | | | | |
| **Packet Loss (%)** | 6.25 | 11.43 | 15.71 | 19.26 | 22.33 | 25.05 | 27.48 | 27.51 | 27.55 |
| **Delay(s)** | 0.0062 | 0.0125 | 0.0188 | 0.025 | 0.031 | 0.038 | 0.044 | 0.05 | 0.056 |
| **Data Rate: 5.5 Mbps** | | | | | | | | | |
| **Packet Loss (%)** | 2.11 | 4.32 | 6.45 | 8.434 | 10.3 | 12.11 | 13.83 | 13.86 | 13.86 |
| **Delay(s)** | 0.0055 | 0.0111 | 0.0167 | 0.0223 | 0.028 | 0.0334 | 0.039 | 0.0445 | 0.05 |
| **Data Rate: 11 Mbps** | | | | | | | | | |
| **Packet Loss (%)** | 1.19 | 2.53 | 3.876 | 5.196 | 6.454 | 7.7 | 8.9 | 8.93 | 8.93 |
| **Delay(s)** | 0.00535 | 0.0107 | 0.016 | 0.021 | 0.027 | 0.032 | 0.037 | 0.043 | 0.048 |
| **Data Rate: 54 Mbps** | | | | | | | | | |
| **Packet Loss (%)** | 0.2195 | 0.498 | 0.788 | 1.0845 | 1.383 | 1.681 | 1.9725 | 1.9895 | 1.991 |
| **Delay(s)** | 0.0051 | 0.0102 | 0.0153 | 0.0204 | 0.0255 | 0.0306 | 0.0357 | 0.0408 | 0.0459 |

**Table 3.2:** VoIP 64 Kbps Performance metrics for linear configuration of nodes for IEEE 802.11b/g. G711 codec and 252 bytes per packet

| Multicast Receivers | $N_2$ | $N_3$ | $N_4$ | $N_5$ | $N_6$ | $N_7$ | $N_8$ | $N_9$ | $N_{10}$ |
|---|---|---|---|---|---|---|---|---|---|
| **Data Rate: 2 Mbps** | | | | | | | | | |
| **Packet Loss (%)** | 0.636 | 1.8 | 3.39 | 5.23 | 7.13 | 9.05 | 10.9 | 10.933 | 10.935 |
| **Delay(s)** | 0.0064 | 0.0129 | 0.0194 | 0.026 | 0.0325 | 0.0391 | 0.0457 | 0.052 | 0.0586 |
| **Data Rate: 5.5 Mbps** | | | | | | | | | |
| **Packet Loss (%)** | 0.206 | 0.544 | 1.053 | 1.732 | 2.497 | 3.34 | 4.218 | 4.24 | 4.24 |
| **Delay(s)** | 0.0056 | 0.0112 | 0.0168 | 0.0225 | 0.0282 | 0.0338 | 0.0395 | 0.045 | 0.051 |
| **Data Rate: 11 Mbps** | | | | | | | | | |
| **Packet Loss (%)** | 0.13 | 0.32 | 0.612 | 0.99 | 1.44 | 1.93 | 2.457 | 2.47 | 2.474 |
| **Delay(s)** | 0.0054 | 0.0107 | 0.0161 | 0.0215 | 0.027 | 0.032 | 0.038 | 0.043 | 0.0486 |
| **Data Rate: 54 Mbps** | | | | | | | | | |
| **Packet Loss (%)** | 0.0315 | 0.0717 | 0.1246 | 0.2076 | 0.3012 | 0.4087 | 0.523 | 0.5269 | 0.5281 |
| **Delay(s)** | 0.0051 | 0.0102 | 0.0153 | 0.0204 | 0.0255 | 0.0306 | 0.0357 | 0.0408 | 0.0459 |

**Table 3.3:** VoIP 64 Kbps Performance metrics for linear configuration of nodes for IEEE 802.11b/g. G711 codec and 332 bytes per packet

| Multicast Receivers | $N_2$ | $N_3$ | $N_4$ | $N_5$ | $N_6$ | $N_7$ | $N_8$ | $N_9$ | $N_{10}$ |
|---|---|---|---|---|---|---|---|---|---|
| **Data Rate: 2 Mbps** | | | | | | | | | |
| Packet Loss (%) | 0.62 | 1.1 | 1.51 | 1.95 | 2.44 | 3.01 | 3.65 | 3.67 | 3.67 |
| Delay(s) | 0.00675 | 0.0135 | 0.0202 | 0.027 | 0.034 | 0.0406 | 0.047 | 0.054 | 0.0609 |
| **Data Rate: 5.5 Mbps** | | | | | | | | | |
| Packet Loss (%) | 0.308 | 0.55 | 0.741 | 0.914 | 1.088 | 1.28 | 1.5 | 1.51 | 1.51 |
| Delay(s) | 0.0057 | 0.0114 | 0.017 | 0.023 | 0.0286 | 0.0344 | 0.04 | 0.046 | 0.051 |
| **Data Rate: 11 Mbps** | | | | | | | | | |
| Packet Loss (%) | 0.198 | 0.355 | 0.486 | 0.6 | 0.697 | 0.813 | 0.94 | 0.946 | 0.948 |
| Delay(s) | 0.0054 | 0.0109 | 0.016 | 0.021 | 0.027 | 0.032 | 0.038 | 0.043 | 0.049 |
| **Data Rate: 54 Mbps** | | | | | | | | | |
| Packet Loss (%) | 0.04 | 0.0772 | 0.103 | 0.13 | 0.152 | 0.1794 | 0.2044 | 0.2074 | 0.209 |
| Delay(s) | 0.0051 | 0.0102 | 0.0153 | 0.0204 | 0.0256 | 0.0306 | 0.0358 | 0.0409 | 0.046 |

## Appendix 4. Simulation Results for Packet Loss and Delay with ODMRP

**Table 4.1:** Packet Loss and end-to-end Delay at 2 Mbps (FG-Time-Out: 12 s)

| Experiment | Feature | $N_2$ | $N_3$ | $N_4$ |
|---|---|---|---|---|
| **Type 1** | **Packet Loss (%)** | 0 | 64.97 | 0 |
| **Type 1** | **Delay (sec)** | 0.010963 | 0.021827 | 0.010963 |
| **Type 2** | **Packet Loss (%)** | X | 54.89 | X |
| **Type 2** | **Delay (sec)** | X | 0.021582 | X |

**Table 4.2:** Packet Loss and end-to-end Delay at 2 Mbps (FG-Time-Out: 500 s)

| Experiment | Feature | $N_2$ | $N_3$ | $N_4$ |
|---|---|---|---|---|
| **Type 1** | **Packet Loss (%)** | 0 | 76.068 | 0 |
| **Type 1** | **Delay (sec)** | 0.011082 | 0.019984 | 0.011082 |
| **Type 2** | **Packet Loss (%)** | X | 77.081 | X |
| **Type 2** | **Delay (sec)** | X | 0.019384 | X |

**Table 4.3:** Packet Loss and end-to-end Delay at 36 Mbps (FW-Time-Out: 12 s)

| Experiment | Feature | $N_2$ | $N_3$ | $N_4$ |
|---|---|---|---|---|
| **Type 1** | **Packet Loss (%)** | 0 | 5.475 | 0 |
| **Type 1** | **Delay (sec)** | 0.005379 | 0.009393 | 0.005379 |
| **Type 2** | **Packet Loss (%)** | X | 5.402 | X |
| **Type 2** | **Delay (sec)** | X | 0.009351 | X |

**Table 4.4:** Packet Loss and end-to-end Delay at 36 Mbps (FG-Time-Out: 500 s)

| Experiment | Feature | $N_2$ | $N_3$ | $N_4$ |
|---|---|---|---|---|
| **Type 1** | **Packet Loss (%)** | 0 | 6.725 | 0 |
| **Type 1** | **Delay (sec)** | 0.005386 | 0.009031 | 0.005386 |
| **Type 2** | **Packet Loss (%)** | X | 6.658 | X |
| **Type 2** | **Delay (sec)** | X | 0.009 | X |

**Table 4.5:** Packet Loss and end-to-end Delay at 18 Mbps (FG-Time-Out: 12 s)

| Experiment | Feature | $N_2$ | $N_3$ | $N_4$ |
|---|---|---|---|---|
| **Type 1** | **Packet Loss (%)** | 0 | 9.685 | 0 |
| **Type 1** | **Delay (sec)** | 0.005714 | 0.010109 | 0.005714 |
| **Type 2** | **Packet Loss (%)** | X | 9.786 | X |
| **Type 2** | **Delay (sec)** | X | 0.010038 | X |

**Table 4.6:** Packet Loss and end-to-end Delay at 18 Mbps (FG-Time-Out: 500 s)

| Experiment | Feature | $N_2$ | $N_3$ | $N_4$ |
|---|---|---|---|---|
| **Type 1** | **Packet Loss (%)** | 0 | 12.635 | 0 |
| **Type 1** | **Delay (sec)** | 0.005715 | 0.009607 | 0.005715 |
| **Type 2** | **Packet Loss (%)** | X | 12.65 | X |
| **Type 2** | **Delay (sec)** | X | 0.009556 | X |

**Table 4.7:** Packet Loss and end-to-end Delay at 9 Mbps (FG-Time-Out: 12 s)

| Experiment | Feature | $N_2$ | $N_3$ | $N_4$ |
|---|---|---|---|---|
| **Type 1** | **Packet Loss (%)** | 0 | 17.055 | 0 |
| **Type 1** | **Delay (sec)** | 0.00635 | 0.01154 | 0.00635 |
| **Type 2** | **Packet Loss (%)** | X | 17.052 | X |
| **Type 2** | **Delay (sec)** | X | 0.011448 | X |

**Table 4.8:** Packet Loss and end-to-end Delay at 9 Mbps (FG-Time-Out: 500 s)

| Experiment | Feature | $N_2$ | $N_3$ | $N_4$ |
|---|---|---|---|---|
| **Type 1** | **Packet Loss (%)** | 0 | 24.52 | 0 |
| **Type 1** | **Delay (sec)** | 0.006351 | 0.010726 | 0.006351 |
| **Type 2** | **Packet Loss (%)** | X | 23.897 | X |
| **Type 2** | **Delay (sec)** | X | 0.010655 | X |