Routing Protocols Performance and Intelligent Quality of Service applied to MANETs

Aïcha Saïd

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DECLARATION

This is to certify that I am responsible for the work submitted in the thesis, and that neither the thesis nor the original work contained therein has been submitted to this or any institution for a higher degree.

Signature: ……………………………………………..

Name : **Aïcha Saïd**

Date : **November 2008**

Abstract

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The wireless revolution prompted by the success of IEEE 802.11 standard has pressed the research community to deal with requirements of new wireless networks. In particular, wireless ad-hoc networks which are, specifically, a collection of wireless mobile nodes dynamically forming a temporary network without the use of any pre-existing infrastructure or centralised administration. Routing protocols used in ad-hoc networks must automatically and continually adjust to environments. Most emerging network services require specialised Quality-of-Service (QoS) functionalities that cannot be provided by the current QoS-unaware routing protocols.

Despite the large amount of research in these areas, several issues still need further investigation. The following points have become main concerns: i) traditional use of the hop count metric does not capture the very nature of wireless paths, resulting in poor performance of wireless networks; ii) the lack of comprehensive simulation methods to effectively observed performance of networks in various conditions and iii) the complexity of multi-constraint routing decisions, resulting in poor service quality in the end-user's point of view.

This study takes an experimental approach to the evaluation of ad-hoc routing protocols and focuses on routing parameters as well as multimedia application QoS performance. In this thesis, we tackle the above mentioned issues and implement an efficient solution for the multi-constraint problem based on network measurements of valid experiments set-up. This study is exclusively based on simulations using NS-2 network simulator. In order to obtain an overview of the limitations of current conventional routing protocols, AODV and DSR protocols are used and their limitations in terms of QoS are measured and discussed. Operating conditions vary greatly from a static, lightly loaded network to constantly moving nodes with up to 10 simultaneous transmission connections. The results show that network performance degrades quickly and that QoS requirement was hardly met by any of these protocols.

To evaluate the overall network performance, a new fuzzy logic assessment approach was developed taking into account the QoS parameters requirement of the transmitted application. Critical parameters were obtained through detailed simulation experiments under demanding operating conditions. These parameters were used as input to the fuzzy logic system to allow the computation of a single metric to represent the input variables (i.e. delay, jitter and throughput). The end results show that without a complicated mathematical model, a QoS value can be computed. This study addresses both theoretical aspects of QoS performance and routing progress in ad-hoc networks as well as practical issues in the set-up of simulation based studies.

Finally, this study indicated that intelligent techniques can be effective for processing multiple QoS metrics to obtain an overall parameter that represents the application QoS. They can be adapted, not only to QoS routing, but to various aspects of QoS provisioning techniques.

LIST OF PUBLICATIONS

In the course of completing this study, the following articles were published:

A. Said, R. Saatchi and R. Strachan. *"Intelligent Quality of Service Routing for Supporting Multimedia Applications in Ad-Hoc Wireless Networks".* IEE/EPSRC Postgraduate Research Conference in Electronics, Photonics, Communications & Networks, and Computing Science (PREP 2004), University of Hertfordshire, UK, 5-7 April 2004.

A. Said, R. Saatchi and R. Strachan. *"Ad-Hoc Wireless Network Evaluation And a Proposal for Intelligent Quality of Service Routing".* In Proceedings of EPSRC Fifth Annual Postgraduate Symposium on the Convergence of Telecommunications, Networking and Broadcasting (PGNet 2004), Liverpool John Moores University, Liverpool, U.K, 28-29 June 2004. ISBN:1-9025-6010-8.

A. Said, R. Saatchi and R. Strachan. *"Implementing QoS Routing in Ad-Hoc Wireless Networks: Practical Issues and Some Proposed Solutions".* In Proceedings of EPSRC Sixth Annual Postgraduate Symposium on the Convergence of Telecommunications, Networking and Broadcasting (PGNet 2005), Liverpool John Moores University, Liverpool, U.K, 27-28 June 2005. ISBN: 1-902-56011-6.

A. Said, R. Saatchi. 2009. *"A Performance Analysis of AODV and DSR Routing Protocols in Mobile Wireless Networks".* In The Mediterranean Journal of Computers and Networks, **5**(1), ISSN: 1744-2397.

DEDICATION

This thesis is dedicated with a lot of love to:

My mother, Moina Amade Said, for being strong when I left home, supportive when I was away and always been there for me in moments of weaknesses. Je t'aime maman,

My late father, Mohamed Said, peace be upon you. I love you and I miss you papa,

My little brothers, Salim, Ali, Hachim, Allaoui and Nassure, who have seen their only sister leave home when they needed her the most.

Aïcha

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Table of Content

LIST OF FIGURES

LIST OF TABLES

List of Abbreviations

ABR	Adaptive Routing Protocol
ACK	Acknowledgement
AODV	Ad Hoc On-demand Distance Vector
BGP	Border Gateway Protocol
CBR	Constant Bit Rate
CEDAR	Core Extraction Distributed Ad-hoc Routing
CGSR	Cluster-head Gateway Switch Routing Protocol
CTS	Clear to Send
CSMA/CD	Carrier Sense Multiple Access with Collision Detection
CSMA/CA	Carrier Sense Multiple Access with Collision Avoidance
DSDV	Destination Sequenced Distance Vector
DSSS	Direct Sequence Spread Spectrum
DSR	Dynamic Source Routing
FHSS	Frequency Hopping Spread Spectrum
FSR	Fisheye State Routing
FTP	File Transfer Protocol
GSR	Global State Routing
HSR	Hierarchical State Routing
IEEE	Institute of Electrical and Electronics Engineers
IETF	Internet Engineering Task Force
IP	Internet Protocol
ISM	Industrial Scientific Medical
IFQ	Interface Queue
LAN	Local Area Network
LL	Link Layer
MAC	Medium Access Control
MACA	Medium Access with Collision Avoidance
MACA-BI	Medium Access with Collision Avoidance By Invitation
MANET	Mobile Ad Hoc Networks
NS	Network Simulator
OSPF	Open Shortest Path First
PHY	Physical Layer
QoS	Quality of Service
RF	Radio Frequency
RFC	Request For Comments
RREQ	Route Request
RREP	Route Reply
RTS	Request to Send
TCL	Tool Command Language
TCP	Transmission Control Protocol
TORA	Temporally Ordered Routing Algorithm
UDP	User Datagram Protocol
WLAN	Wireless Local Area Network
WMA	Windows Media Audio
WRP	Wireless Routing Protocol
ZRP	Zone Routing Protocol

Introduction

Interest in wireless technologies has grown considerably over the past few years. Various multimedia applications services are now available on the move and portable devices such as laptops are becoming more available and affordable. These applications have stringent requirements, and a network is expected to comply with them in order to limit information degradation and loss. A considerable amount of work has been done within the field of Quality-of-Service (QoS) in telecommunication network. QoS routing plays an important role to achieve an acceptable level of service. Researchers have been developing different techniques to improve network performance, however many issues remain to be solved. QoS routing in mobile ad-hoc networks is challenging because the network topology may change constantly and the available state information for routing are inherently imprecise. Thus techniques to efficiently determine network states and make a decision based on those information are still undergoing research.

In this chapter, the motivations of this research in wireless ad-hoc networks and QoS routing are introduced. The research problems and the present contributions are discussed. Finally, the outline of the rest of the thesis is presented.

1.1 Research overview and Motivations

Nowadays, the use of handheld devices is greatly increasing. Laptops, mobile phones and Personal Digital Assistants (PDA) take an important place in the everyday life. Hence, the challenge is now to make all these devices communicate together in order to build a network. Obviously, this kind of network has to be wireless. Indeed, wireless technology allows flexibility and mobility. In this context, the idea of ad-hoc networks was developed.

A Mobile Ad-hoc NETwork (MANET) (Hännikäinen et al., 2002; Corson and Macker, 1999) is a network architecture that can be rapidly deployed without relying on a pre-existing fixed network infrastructure so that a network can be created as soon as the nodes are within a transmission range of each other. Each node has a wireless access interface (Bluetooth, WLAN, HiperLAN2, UWB, Infra-Red, etc.) and is free to enter and leave the network, frequently, often without warning, and possibly without disruption to other nodes' communication at any time. Also known as infrastructureless mobile network which means that mobile nodes are not centralised. All nodes can function as routers and thus they discover and maintain routes to other nodes. They are also characterised by multi-hop wireless connectivity. Finally, the nodes in the network can be highly mobile, thus rapidly changing the network topology and the presence or absence of links which emphasise the need for efficient dynamic routing protocols. In spite of such volatility, the MANET is expected to deliver diverse traffic types, ranging from pure voice to integrated voice and image, and even possibly some limited video.

Techniques for achieving QoS routing in wireless networks have recently been the subject of intense investigations. Emerging new technologies, such as handheld PDA or laptop computers, allow users to run diverse multimedia applications while on the move. As ad-hoc networks do not have fixed infrastructures, they cannot rely on a centralised node to allow several users to establish communication. A network can emerge as soon as a set of devices is within the same sensing range. Routing, more precisely QoS routing, is therefore becoming more and more important to allow information to be forwarded in the most efficient way. More often, routing protocols normally use single objective optimisation algorithms, where route are selected based on a single metric and packet updates are broadcast to monitor network changes. Shortest-path algorithms are then used in the network path computation. The diverse QoS requirements of multimedia traffic make the definition of a single routing metric very difficult. Furthermore, since each flow has its own characteristics, the same metric is not universally applicable. Therefore, new routing techniques that improve the path computation process for finding a route that guarantee the QoS of the traffic, are needed. QoS-based routing has to guarantee an appropriate performance not just for a single hop but also for the overall network.

QoS parameters must represent the network properties, but the challenge is to find a way to measure and collect this network information in such a dynamic environment. Moreover, the complexity of the path must be considered, which means that a path computation based on a single metric or a combination of metric must not be too complex. Unfortunately, path computation of QoS routing based on two or more additive or multiplicative QoS parameters is known to be NP-complete meaning that there is no solution known using polynomial time algorithms, more details on the NP-complete problem can be found in Wang and Crowcroft (1996). Several heuristics algorithms have been proposed to solve the problem. Sequential filtering is a common one under which a combination of metrics is ordered in some fashion, reflecting the importance of different metrics. A primary metric (e.g., bandwidth) is computed first, paths that are above the threshold required on this metric are eliminated then a subset is eliminated based on the second metric and so forth until a single path is found. There are no routing methods that use a complete set of QoS parameters to determine a route for multimedia data flows (Vogel *et al.*, 1996). Therefore, a novel approach to assess QoS metrics in wireless ad-hoc networks based on fuzzy logic can be valuable as it will allow the computation of a single QoS metric considering diverse multimedia traffic requirements.

The main area of concern in this study is to investigate the limitations of current Ad-hoc On-demand Distance Vector (AODV) and Dynamic Source Routing (DSR) routing protocols, to identify the circumstances for degradation and to provide a method to evaluate and provide QoS when transmitting various applications. This can be achieved by incorporating a novel analysis technique that will solve the NP- problem affecting QoS routing.

In this thesis, investigations, qualitative analysis and proposed methods concerning QoS routing ad-hoc networks are covered. Baselines and methods in conducting and making efficient use of simulation results are presented. This dissertation involves identifying the relevant QoS parameters from the point of view of the network designer and end-user but also highlighting the necessary steps to improve the way current routing protocols can be manipulated in terms of QoS performance.

1.2 Research aims and objective

The overall aim of this research is to analyse the QoS in wireless computer networks for multimedia transmission under various operating conditions. Several techniques were utilised so that the complex task of assessing and quantifying the QoS can be achieved effectively and efficiently. This study evaluates existing methods and devises new methods for measuring and quantifying the overall QoS of wireless networks for transmitting multimedia applications. The study uses the IEEE 802.11 protocol for performing and validating concepts.

The objectives of this study were to:

(i) Investigate techniques, which enable the QoS performance of wireless networks for transmission of multimedia applications to be assessed and quantified.

(ii) Study the effects of operational conditions and resource availability for providing the required QoS.

(iii) Quantitatively evaluate and analyse the QoS performance of wireless networks for transmission of multimedia applications.

(iv) Investigate the advantages of fuzzy logic for QoS performance analysis and provision in MANETs.

(v) Explore how the findings of the proposed method can be used as part of an efficient ad-hoc QoS-aware routing mechanism.

1.3 Thesis contributions

This thesis highlights relevant parameters in the set-up of sound simulation based experiments and demonstrates that multi-constraints QoS routing provisioning solutions can be developed based on intelligent approaches. Firstly, a short survey on current QoS solutions and their applications to routing in wireless ad-hoc networks is provided, stating their limitations. Additionally, multi-constraints route selection challenges in such a network are discussed. Secondly, a study of two popular on-demand routing techniques is carried out and their behaviour is discussed based on simulation experiments. Several QoS metrics are defined and the QoS routing problem is formulated. The inability of these protocols to meet the QoS requirements of multimedia applications is validated against the literature. Therefore, an intelligent model based on fuzzy logic is developed demonstrating that complex mathematical models are not essentially needed to provide a good solution to the selection criteria problem. Thirdly, a proposed approach to assess routing QoS performance is formulated leading to a new cost function which is able to mirror the performance of the routing protocol ability to meet applications' requirements. This metric is more efficient and can be generalised to be used within the current single constraint routing protocols.

1.4 Outline of thesis

The remainder of the thesis is organised as follows. Chapter 2 covers the theory of wireless technologies, the development of Wireless Local Area Networks (WLANs) technology and describes the mode of operations and characteristics of Mobile Ad-hoc Networks (MANETs). The IEEE 802.11 standards are briefly outlined including a description of the physical (PHY) layer and the Media Access Control (MAC) protocols. It also includes an overview of routing protocols in ad-hoc networks and more precisely, a detailed description of the protocols considered for this study. This identifies the main drawbacks of the existing routing mechanisms. This chapter also outlines the QoS aspects from the perspective of routing. This is then followed by reviewing the current state-of-the-art in the area of QoS based routing protocols including their limitations and how these can be minimised by the use of a new efficient metric described in this study. Chapter 3 provides an introduction to fuzzy logic theory. Chapter 4 outlines the experiment methodology used in this research study. The simulation overview, topology, traffic and measurement models that are used throughout this thesis are explained. Also, included in this chapter is the work related to this study. The limitations and performance AODV and DSR protocols in terms of QoS are discussed in Chapter 5 and 6, for various mobility levels and variable network load respectively. These chapters investigate the limitation of the protocols on multi-hop networks through simulation experiments under different network condition operations. The performance of the routing protocol under different operating conditions and the impact on the overall QoS parameters are investigated. These chapters represent the baseline of the proposed approach for this study. In Chapter 6, a fuzzy logic approach is used to assess QoS for various multimedia applications and an overview of the benefits of using such an approach in ad-hoc networking is provided. The assessment and measurement approach based on fuzzy logic are also discussed and tested through extensive simulation experiments. Finally, Chapter 7 concludes this thesis and highlights future research directions.

Wireless Networks, Routing Protocols and Quality-of-Service

2.1 Introduction

There is a growing demand to improve the transmission of multimedia information over wireless networks. The quality of the transmission is particularly important for multimedia type applications. Mechanisms which improve the network's ability to meet the QoS requirements for these multimedia applications are of particular importance in current and future wireless networks. QoS routing allows for these QoS requirements to be met. Conventional routing protocols do not consider the application needs in terms of QoS therefore many efforts have been made to incorporate QoS requirements into route selection algorithms. The focus of this study is based around ad-hoc networking and QoS routing.

The rest of this chapter is organised as follows: The next section provides a general description of wireless technologies and standards. An introduction to wireless technologies is provided in section 2.2. Section 2.3 covers Mobile Ad-hoc Networks (MANETs) characteristics and applications. The various routing protocols developed for MANETs and more precisely the routing protocols used in this study are described in section 2.4. An overview of QoS architectures and parameters are discussed in section 2.5. The definition of QoS-based routing is outlined in section 2.6, as well as the current research around providing QoS routing based on multiple QoS parameters in MANETs. In section 2.7, issues and research challenges are pointed out which lead this study onto the areas that need further investigations. Finally, a summary of all topics covered in this chapter is provided in section 2.8.

2.2 Wireless Networks

2.2.1 Overview

IEEE 802.11 is a standard for Wireless Local Area Networks (WLAN) (Crow *et al.*, 1997) developed by the Institute of Electrical and Electronic Engineer (IEEE) task group (IEEE, 1997). This task group focuses its effort to produce standards for high-speed WLAN. WLAN are an essential part of wireless communication and are based on standards which are provided mostly by two big standardisation parties: the European

Telecommunications Standard Institute (ETSI) and Institute of Electrical and Electronics Engineers (IEEE).

In every country, the use of the radio spectrum is regulated by an organisation: The Federal Communications Commission (FCC) for North America and the European Telecommunications Standard Institute (ETSI) for Europe. These regulators define the allocation of each radio frequency bandwidth. The oldest and most common ones are located at 900MHz and 2.4GHz and are called the ISM bands (Industrial, Scientific and Medical). The main characteristic of these bands is that they are unlicensed; this means that the user is free to use them without having to register or to pay anything (apart from the radio hardware).

As wireless LANs became more popular, the demands on them have also increased for greater bandwidth and to provide better services in crowded networks. Currently the most widely used wireless protocols are IEEE 802.11 b and g with data rates up to 54Mbps. Wireless technologies have shown a rapid growth during recent years. They include Wireless Wide Area Network (*WWAN*), Wireless Metropolitan Area Network (*WMAN*), *WLAN* and Wireless Personal Area Network (*WPAN*). This classification of wireless technologies is based on the coverage area and the data rate (Hännikäinen *et al.,* 2002). New standards have been developed and there are summarised in Table 2.1 (Philip, 2005) and the performance of some of these is reviewed in (Syrjälä, 2003).

Among the most popular and widely used wireless technologies, one can cite WiFi (802.11), Bluetooth (802.15.1) and Zigbee (802.15.4). While technologies such as Bluetooth have been quite a success story, so far Wireless Personal Area Networking products have not been able to make a significant impact on the market. Bluetooth is a short range communication technology, intended to replace cables connecting portable and/or fixed electronic devices. Its key features being robustness, low complexity, low power and low cost technology (Bluetooth, 2006) and (Giovino, 2004). A small network (called piconet) can be formed with as many as seven slaves and a master coordinator. Typical applications include intelligent devices (e.g. PDAs, mobile phones, PCs), data peripherals (e.g. mouse, keyboard), audio peripherals (e.g. headsets), etc.

Table 2.1: Performance overview of IEEE & ETSI Wireless LAN standards

Standard	Frequency (GHz)	Physical Speed Mbps	Modulation Technique
IEEE 802.11 (WiFi)	2.4	2	DSSS[1], FHSS[2]
IEEE 802.11b	2.4	11	DSSS
IEEE 802.11a	5	54	OFDM[3]
IEEE 802.11g	2.4	54	DSSS, OFDM
IEEE 802.15.1 (Bluetooth)	2.4	1	FHSS
IEEE 802.15.4 (Zigbee)	2.4	0.250	DSSS
ETSI HIPERLAN/1	5	24	CSMA/CA[4]
ETSI HIPERLAN/2	5	54	OFDM
ETSI HIPERACCESS	40-43.5	25	< 5 km
ETSI HIPERLINK	17	155	150 m

Bluetooth and Zigbee technologies are guided by IEEE 802.15 Personal Area Networks group which defines the PHYsical (PHY) and Medium Access Control (MAC) layers. Above these two layers of the Open System Interconnect (OSI) model, Zigbee define applications, security and enable interoperability from different manufacturers (Giovino, 2004). The limitation of Bluetooth in the automation arena has led to the development of the wireless low data rate personal networking technology Zigbee (802.15.4), for the home/industrial automation. It has received a tremendous boost among the embedded system and wireless mesh network industries. Most technologies designed so far, primarily focus either directly or indirectly on the ability to support higher data rates, with a wider operating space (range), which has a direct impact on the power requirements (Marandin, 2006). Zigbee has been adopted for applications in mesh networks. Target applications include remote control, monitoring and sensor applications. It is low cost, low speed and can be used for very diverse applications. It is targeted at transmission speed of 20-250 KBps with a transmission range of well over 50 meters and an excellent battery life (Giovino, 2004).

The IEEE 802.11 standard specifies the MAC protocol procedures and three types of PHY layers. Two of them use radio and one uses infrared. Here, the focus will be on the radio type interface.

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[1] Direct Sequence Spread Spectrum
[2] Frequency Hopping Spread Spectrum
[3] Orthogonal Frequency Divisional Multiplexing
[4] Carrier Sense Multiple Access/Collision Avoidance

2.2.2 The Radio Physical Layer (PHY)

The ISM band specify that Spread Spectrum has to be used (either Direct Sequence or Frequency Hopping). Spread spectrum is a technique trading bandwidth for reliability (Tourrilhes, 2000). The goal is to use more bandwidth than the system really needs for transmission to reduce the impact of localised interferences.

Direct Sequence Spread Spectrum (DSSS): DSSS spreads the transmission signal over an allowed band (for example, 25MHz). A random binary string which is the spreading code, is used to modulate the transmitted signal. The higher the spreading ratio, the more the signal is resistant to interference. IEEE 802.11 standard requires spreading ratio of 11 (Van Nee, 2000). Recovery from interference is fast in DSSS, because of the ability to spread the signal over a wider band. The radio interface Lucent WaveLAN^{TM} DSSS is an example of a commercial product using this modulation technique (Tuch, 1993).

Frequency Hopping Spread Spectrum (FHSS): FHSS uses a set of narrow channels and go through all of them in a sequence, for example, the 2.4GHz ISM band is divided in 79 channels of 1MHz (Tourrilhes, 2000). The system avoids interferences by never staying on the same channel. Frequency hopping is less vulnerable to interference than direct sequence, because frequency is always shifting. It is also difficult to intercept a frequency hopping communication. An example for this technology is Wave-Access Jaguar (Tuch, 1993).

2.2.3 The Media Access Control (MAC) Protocol

The main function of the MAC protocol is to regulate the use of the medium, and this is performed through a channel access mechanism. The MAC protocol is concerned with per-link communications, not end-to-end. Most wired LANs use Carrier Sense Medium Access with Collision Detection (CSMA/CD) (Haas *et al,*. 1999) as the MAC protocol. Carrier sense means that the station will listen before it transmits. If someone is already transmitting, the sender waits and tries again later. When two stations send at the same time, transmissions collide and information is lost. Collision detection handles this situation, by listening to the signal it is transmitting to ensure everything is going right. Whenever a collision occurs, nodes stop and try again at a later time, which is determined by the backoff algorithm.

Applicability of the existing MAC-layer protocol, in particular the family of the Carrier Sense Multiple Access (CSMA), to the radio environment is limited by the following two interference mechanisms: the hidden terminal (Figure 2.1) and the exposed terminal problems (Figure 2.2) (Tobagi and Kleinrock, 1975; Hsieh and Sivakumar, 2002; Shugong Xu and Saadawi, 2001 and Haas *et al.,* 1999).

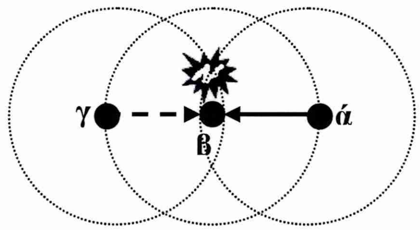


Figure 2.1: The hidden terminal problem

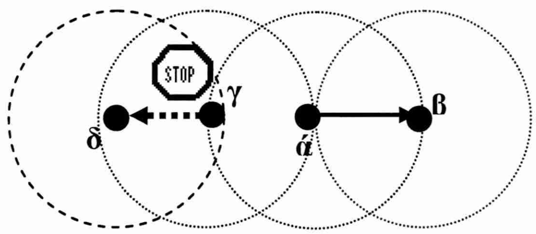


Figure 2.2: The exposed terminal problem

In the hidden terminal problem, two nodes are said to be hidden from one another (out of signal range), when both attempt to send information to the same receiving node, this result in a collision of data at the receiver node as illustrated in Figure 2.1. In the exposed terminal problem illustrated by Figure 2.2, node α is currently transmitting to node β. Node γ listens to the medium, but since there is an obstruction between the nodes α and γ, no transmission can be carried out between γ and δ, causing the bandwidth to be wasted. In general, the hidden terminal problem reduces the capacity of a network due to increasing the number of collisions, while the exposed terminal problem reduces the network capacity due to the unnecessarily deferring nodes from transmitting. Some experiments have been carried out to point out the consequences of these problems in multi-hop wireless ad-hoc networks performance. These can be found in (Hsieh and Sivakumar, 2002_a and Shugong Xu and Saadawi, 2001). The title of one of these papers is very explicit: does the IEEE 802.11 MAC protocol work well in multi-hop wireless ad-hoc networks? The conclusion of this paper is: no, it does not (Shugong Xu and Saadawi, 2001).

Several attempts have been made to reduce the effect of the hidden terminal and the exposed terminal problems. Progress has been made in designing random access MAC protocols based on Carrier Sense Multiple Access (CSMA). For example, the IEEE standard 802.11 (Crow *et al.* 1997) specifies physical and MAC layer protocols for wireless LANs. It uses a CSMA/CA (Carrier Sense Multiple Access/Collision Avoidance) protocol that persists on a busy channel and employs a random backoff after the channel switches to idle to avoid the possibility of collision at the receiver (Joa-Ng and Lu, 1999_b). The IEEE 802.11 standard optionally uses RTS/CTS (request-to-send/clear-to-send) packets to destination, to avoid the classical hidden terminal problem.

RTS/CTS Mechanism

The necessity of a dialogue between the transmitting and the receiving nodes that precede the actual transmission, referred to as the RTS/CTS dialogue, has been generally accepted. The RTS/CTS dialogue is described in Figure 2.3. A node ready to transmit a packet sends a short control packet, the Request To Send (RTS). All nodes that hear the RTS defer from accessing the channel for the duration of the RTS/CTS dialogue. The destination, upon reception of the RTS responds with another short control packet, the Clear To Send (CTS). This mechanism limits the effect of the hidden terminal problem, but the exposed terminal problem remains. Studies to limit the effect of these two problems are still ongoing.

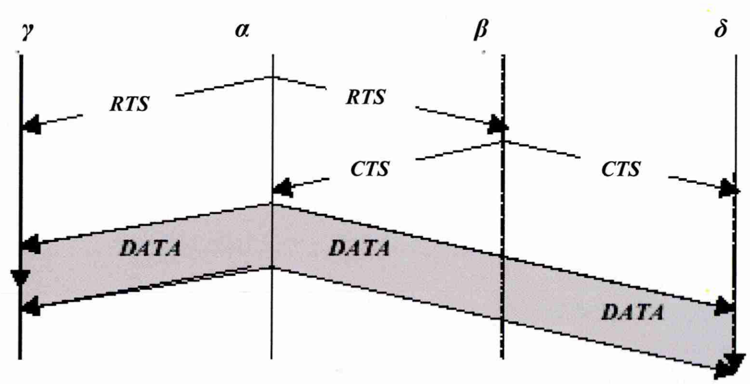


Figure 2.3: RTS-CTS mechanism (Haas *et al.* 1999)

2.3 Mobile Ad-Hoc Networks (MANETs)

With the large increase in the use of handheld devices, laptops, mobile phones and PDAs play an important part in our everyday life. The challenge is now to make all these devices communicate together in order to build a network. Obviously, this kind of networks has to be wireless. Indeed, wireless technology allows flexibility and mobility. In this context, the idea of mobile ad-hoc networks was developed. This section introduces MANETs concepts, characteristics and applications.

2.3.1 Introduction

A Mobile Ad-hoc NETwork (MANET) (Hännikäinen *et al.,* 2002 and Corson and Macker, 1999) is a network architecture that can be rapidly deployed without relying on pre-existing fixed network infrastructure. Mobile nodes come together for a period of time to exchange information while continuing to move and so the network must be prepared to adapt continuously.

Nodes in the MANET exhibit nomadic behaviour by freely migrating within some area, dynamically creating and tearing down associations with other nodes. Groups of nodes that have a common goal can create formations (clusters) and migrate together, similarly to military units on missions or to guided tours on excursions. Nodes can communicate with each other at any time and without restrictions, except for connectivity limitations and subject to security provisions (Frodigh *et al.*, 2000). Although an autonomous MANET is useful in many scenarios, the integration of MANET with the Internet is desirable due to the growing interest in the Internet and associated technologies. Thus, a MANET may be connected at the edges of the fixed, wired Internet. A mobile Internet router is one of the main requirements for an ad-hoc network to gain access to the Internet. Under this scenario, the mobile terminals in ad-hoc networks might dynamically obtain and lose Internet connectivity through interfaces not participating in the MANET routing. Among mobile terminals, some of them can directly connect to the Internet and serve as Access Points for the rest of the mobile terminals in the Internet mobile ad-hoc network. Therefore, an Access Point will provide a gateway for the Internet, and is assumed to have access to any information. A typical scenario would be a laptop that might be connected to the Internet through an Ethernet link for a limited time while participating in a MANET through a wireless interface. As MANETs gain in popularity, their need to support real-time and multimedia applications is growing as well. Such applications have stringent QoS requirements such as bandwidth, delay, and delay jitter. QoS is more difficult to achieve in ad-hoc networks, because the wireless bandwidth is shared among adjacent nodes and the network topology changes as the nodes move thus making the state information inherently imprecise.

2.3.2 Multi-hop Communication Network

A MANET is a *peer-to-peer* network that allows direct communication between any two nodes, when adequate radio propagation conditions exist between these two nodes and subject to transmission power limitations of the nodes. If there is no direct link between the source and the destination nodes, *multi-hop* routing is used. Multi-hopping is a technique which allows the virtual extension of the transmission range for each node. A node can send a packet to another one even if this destination node is not in the radio range from the source. A sent packet crosses several wireless links to reach the destination. But, to be efficient, it is necessary that the different nodes which participate to the transmitting process are well distributed in space (Frodigh *et al.,* 2000).

Multi-hop communication has three main performance advantages compared with single hop communication solution:

1. Adaptability. By deploying a multi- hop data forwarding, packets can be routed around obstructions or areas captured by enemy units, which is very crucial for the battlefield scenario.
2. Reduction of interference. Packet forwarding over multiple hops via small radio transmissions will exploit spatial reuse, by allowing multiple concurrent packet transmissions in different regions of the network, and maximize throughput (Keinrock and Silvester, 1987).
3. Energy consumption efficiency. Packet forwarding via multiple small radio transmissions as opposed to a single large radius transmission will improve the throughput per unit energy (Hsieh and Sivakumar, 2002_b).

In this type of wireless network, potentially every node becomes a router; it must be able to forward traffic on behalf of others. Each node in the network maintains a routing table containing each destination with a corresponding next hop node and link cost. Packets are forwarded by consulting the routing table for the next-hop node leading to the shortest path to the destination. In MANETs, the routing table at each node can be thought of as a description of the network topology. The goal of the routing protocol is to ensure that the overall data structure contains a consistent and correct view of the actual network topology. All nodes need to be within the transmission range of one another to be able to establish a transmission. Two nodes in a Mobile Ad-hoc NETwork (MANET) can communicate if the distance between them is less than the minimum of their two broadcast ranges (Bose *et al.,* 1999). Furthermore, transmission is not necessarily made via a straight path but it can also involve intermediate nodes and in this case multi-hop routing protocols mechanisms are needed. Figure 2.4 depicts an example of MANET including four mobile nodes. In this example, nodes S, B, and C are within the transmission ranges of each other. Thus, nodes S, B, and C are named neighbours, and they can communicate directly with each other. However, node D does not reside in the transmission range of node S. If node S wants to send data to node D, the data must be routed through the intermediate node, such as node C, which acts as the router between node S and node D.

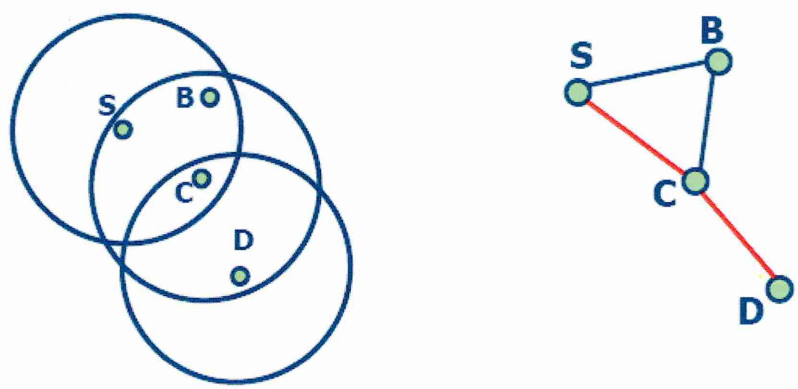


Figure 2.4: An example of multi-hop mobile ad-hoc network

2.3.3 Applications of MANET

In a region where there is no fixed infrastructure or it is costly and time-consuming to build one, ad-hoc wireless networks can provide an effective network for communication. They have numerous applications. Below are a few examples of their use (Frodigh *et al.,* 2000; Qin and Kunz, 2004; Larsson and Hedman, 1998):

- tactical operation - for fast establishment of military communication during the deployment of forces in unknown and hostile terrain;
- rescue missions - for communication in areas without adequate wireless coverage;
- national security - for communication in times of national crisis, where the existing communication infrastructure is non-operational due to a natural disaster or a global war;
- law enforcement - for fast establishment of communication infrastructure during law enforcement operations;

- commercial use - for setting up communication in exhibitions, conferences, or sales presentations;
- education - for operation of virtual classrooms and videoconferencing;
- sensor networks - for communication between intelligent sensors (e.g. robotics) mounted on mobile platforms.

The future for this type of network seems to be very clear; the market for wireless communications expands every year. Ad-hoc will be needed everywhere, people will want to communicate by deploying a cheap or temporary network. This will lead to lower prices providing a further incentive to use computing.

2.3.4 Operational Challenges

Because of the lack of central elements in a MANET, ad-hoc networks require distributed protocols. The main challenges in the design and operation of MANETs, compared to more traditional wireless networks, stem from this lack of centralised entity, the potential for rapid node movement, and the fact that all communication is carried over the wireless medium. In standard cellular wireless networks there are a number of centralised entities (e.g., the Base Stations, the Mobile Switching Centres (MSCs), etc.). In ad-hoc networks, there is no pre-existing infrastructure, and these centralised entities do not exist. The centralised entities in the cellular networks perform the function of coordination. The lack of these entities in the MANETs requires distributed algorithms to perform these functions. All communications between all network entities in ad-hoc networks are carried over the wireless medium. Due to the radio communications being vulnerable to propagation impairments, connectivity between network nodes is not guaranteed. In fact, intermittent and sporadic connectivity may be quite common. Additionally, as the wireless bandwidth is limited, its use should be minimized. Finally, as some of the mobile devices are expected to be handheld with limited power sources, the required transmission power should be minimised as well. Therefore, the transmission radius of each mobile is limited, and channels assigned to mobiles are typically spatially reused (Frodigh *et al.*, 2000). Consequently, since the transmission radius is much smaller than the network span, communication between two nodes often needs to be relayed through intermediate nodes; i.e., multi-hop routing is used.

Because of the possible rapid movement of the nodes and variable propagation conditions, network information, such as a route table, becomes obsolete quickly. Frequent network reconfiguration may trigger frequent exchanges of control information to reflect the current state of the network. However, the short lifetime of this information means that a large portion of this information may never be used. Thus, the bandwidth used for distribution of the routing update information is wasted. In spite of these attributes, the design of a MANETs still needs to allow for a high degree of reliability, survivability, availability, and manageability of the network.

There are a number of issues to consider when deploying MANETs. The following are some of the main issues exposed by Chen (1999) and Liu (2005):

1. **Unpredictability of environment**: Ad-hoc networks may be deployed in unknown terrains, hazardous conditions, and even hostile environments where tampering or the actual destruction of a node may be imminent. Depending on the environment, node failures may occur frequently.
2. **Unreliability of wireless medium**: Communication through the wireless medium is unreliable and subject to errors. Also, due to varying environmental conditions such as high levels of electro-magnetic interference (EMI) or inclement weather, the quality of the wireless link may be unpredictable. Furthermore, the wireless medium for communication makes ad-hoc networks extremely vulnerable to security attacks (Zhou and Haas, 1999). It opens another venue for initiating link level attacks ranging from passive eavesdropping to message replay and message distortion in hostile environment such as military battle site. Therefore, security is one of the major challenges in using ad-hoc networks.
3. **Resource-constrained nodes**: Nodes in a MANET are typically battery-powered as well as limited in storage and processing capabilities. Moreover, they may be situated in areas where it is not possible to re-charge and thus have limited lifetimes. Because of these limitations, they must have algorithms which are energy-efficient as well as operating with limited processing and memory resources. The available bandwidth of the wireless medium may also be limited because nodes may not be able to sacrifice the energy consumed when operating at full link speed.

4. **Dynamic topology**: The topology in an ad-hoc network may change constantly due to the mobility of nodes. As nodes move in and out of range of each other, some links break while new links between nodes are created. In ad-hoc networks, most nodes are expected to route packets for other nodes in the network, while they themselves may also be a source or destination for one or more application flows. Keeping current knowledge of the network topology is an important requirement in any network management system. In a wireless ad-hoc environment, it is crucial that the management system keeps up with the frequent topology changes. However, the frequent exchange of topology information may lead to considerable signalling overhead, congesting low bandwidth wireless links, and possibly depleting the limited battery life of the nodes involved. Hence, the choice of mechanism used to collect or manage topology information is critical.

As a result of these issues, MANETs are prone to numerous types of faults including,

1. **Transmission errors**: The unreliability of the wireless medium and the unpredictability of the environment may lead to transmitted packets being garbled and thus received in error.
2. **Node failures**: Nodes may fail at any time due to different types of hazardous conditions in the environment. They may also drop out of the network either voluntarily or when their energy supply is depleted.
3. **Link failures**: Node failures as well as changing environmental conditions (e.g., increased levels of EMI) may cause links between nodes to break.
4. **Route breakages**: When the network topology changes due to node/link failures and/or node/link additions to the network, routes become out-of-date and thus incorrect. Depending upon the network transport protocol, packets forwarded through stale routes may either eventually be dropped or be delayed; packets may take a circuitous route before eventually arriving at the destination node.
5. **Congested nodes or links**: Due to the topology of the network and the nature of the routing protocol, certain nodes or links may become over utilised, i.e., congested. This will lead to either larger delays or packet loss.

Based on the above, some required features of MANETs have been outlined in the literature. These include:

- **Robust routing and mobility management algorithms** to increase the network's reliability and availability; e.g., to reduce the chances that any network component is isolated from the rest of the network;
- **Adaptive algorithms and protocols** to adjust to frequently changing, network, and traffic conditions;
- **Low-overhead algorithms and protocols** to preserve the radio communication resource;
- **Multiple (distinct) routes,** between a source and a destination, to reduce congestion in the vicinity of certain nodes, and to increase reliability and survivability;
- **Robust network architecture** to avoid susceptibility to network failures, congestion around high-level nodes, and the penalty due to inefficient routing.

Routing protocols for MANETs must deal with these issues to be effective. These key features will be used to evaluate the performance of the routing protocols used in this study. In the following section, we present an overview of some of the key unipath routing protocols for MANETs.

2.4 Routing Protocols in MANET

In recent years several routing protocols have been proposed for use in ad-hoc networks, Royer and Toh (1999) provide a survey of these. The routing protocol is the primary issue and has to be supported before any applications can be deployed. Before presenting the current approaches for design and implementation of QoS routing protocols, it is important to briefly discuss the existing best-effort routing protocols for MANETs. This section will include an overview of the different routing mechanisms including a detailed description of the protocols considered for this study.

2.4.1 Overview

Typically, every node in an ad-hoc network serves as a router for other nodes and paths from source to destination often requiring multiple hops. Compared to wired networks, wireless ad-hoc networks have less bandwidth, longer paths and less stable connectivity, all of which render routing protocols from wired networks less suitable for the wireless world. The main group of proposals comes from the work of IETF's MANET group (Corson and Macker, 1999). Many routing protocols have been designed to discover and maintain routes between source and destination nodes.

Among the most important and classic routing algorithms for MANETs, two categories have evolved. Each of these types has its own advantages, disadvantages, and appropriateness of use in certain types of ad-hoc networks depending on the mobility, number of nodes involved, node density, underlying link layer technology, and general characteristics of the environment and applications being supported. These routing protocols are designed for Internet Protocols (IP) based homogenous, mobile ad-hoc networks under the following assumptions: each node in the network has identical capability and has a unique address (IP address for example). Number of hops is used as the only route selection criteria. Other parameters, like route delay, energy usage, load balancing or quality of service are not considered. These protocols focus on fast route establishment and re-establishment and route maintenance with minimal overhead.

Those routing protocols are classified into two main categories: topology-based and position-based (c.f. Figure 2.5). In position-based routing protocols, mobile nodes know physical position information by geographic location techniques such as GPS (e.g., the Distance Routing Effect Algorithm for Mobility (DREAM) (Basagni *et al.,* 1998), Location-Aided Routing algorithm (LAR) (Ko and Vaidya, 1998), the Geographical Routing Algorithm (GRA) (Jain *et al.,* 2001) and Landmark Ad-hoc Routing (LANMAR) (Pei *et al,.* 2000). Topology-based routing protocols are based on the information concerning links. This study is considering only topology-based protocols since they have been widely used in the literature. These can be generally grouped into three routing strategy categories. Despite being designed for the same type of network, the characteristics of each of these routing protocols are quite distinct;

- Proactive or table-driven protocols.
- Reactive or on-demand protocols.
- Centralised or distributed

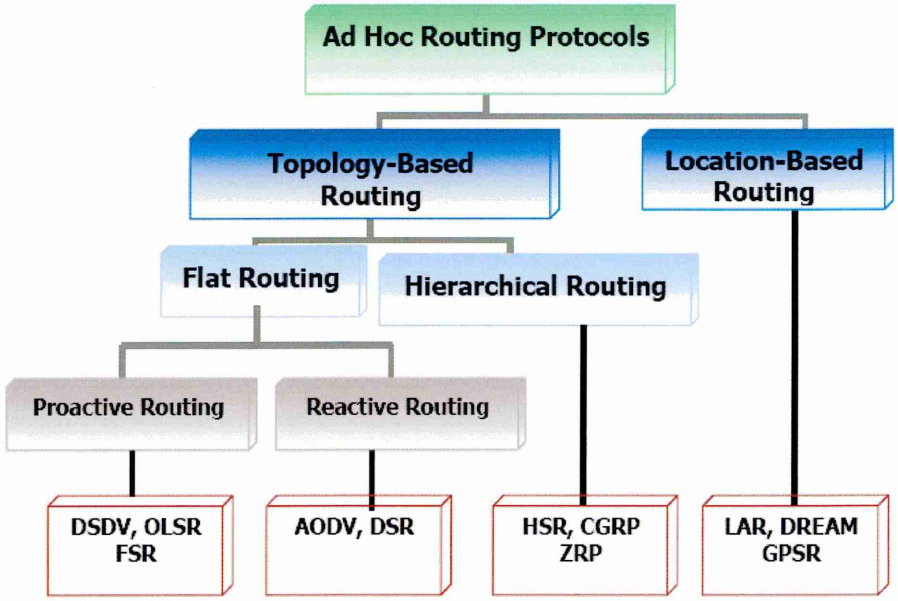


Figure 2.5: Taxonomy of Mobile Ad-Hoc Routing Protocols

2.4.2 Proactive (Table-driven) Protocols

A proactive routing protocol maintains network topology through the periodic exchange of control information. Event driven protocols will not send any routing update packets, if no change in topology occurs. Only if a node detects a change of the topology (usually a node moves out of reach of this node, or a new node comes close enough), this is reported to other nodes, according to the strategy of the routing protocol. A proactive protocol might be appropriate in a network in which non-local communications are normal and route maintenance must be rapid.

Table-driven protocols continuously learn the topology of the network by exchanging topological information among the network nodes. These protocols require each node to maintain one or more tables to store routing information. Thus, when there is a need for a route to a destination, such route information is available immediately. The early protocols that were proposed for routing in ad-hoc networks were proactive Distance Vector protocols based on the Distributed Bellman-Ford (DBF) algorithm (Bertsekas and Gallager, 1992; Bellman, 1957; Ford and Fulkerson, 1962) and Routing Internet Protocol (RIP) (Malkin, 1998). The problems of the DBF algorithm i.e. exponential growth of processing time with respect to the graph size, convergence and excessive control traffic, which are especially an issue in limited resources ad-hoc networks, were outlined in (Perkins and Royer, 1999; Yuan, 1999). Proactive routing protocols of this type include the Destination-Sequenced Distance Vector (DSDV) Routing (Perkins and Bhagwat, 1994), Wireless Routing Protocol (WRP) (Murthy and Garcia-Luna-Aceves, 1996) and Least Resistance Routing (LRR) (Pursley and Russell, 1993). An approach was taken to address the convergence problem by applying link state protocol (Moy, 1998) to the ad-hoc environment. Examples of the latter are the Optimised Link State Routing protocol (OLSR) (Clausen *et al.,* 2001) and Source Tree Adaptive Routing (STAR) (Garcia-Luna-Aceves and Spohn, 1999).

The proactive protocols generally provide a source node with readily available routes to all other nodes. They incur no routing delay or query traffic. Although proactive protocols can produce the required route immediately, they may waste too much of the network resources by unnecessary control traffic in the attempt to always maintain the updated network topology, whether that information is needed for routing or not.

2.4.3 Reactive (On-demand) Protocols

The first type of topology based routing is a reactive routing protocol, which does not require maintaining a route to each destination of the network on a continual basis. Instead, routes are established on demand by the source. This source-initiated strategy creates routes only when desired by the source. Routing protocols do not continuously maintain an up-to-date topology of the network. But, when the need arises, a reactive protocol invokes a procedure to find a route to the destination; such a procedure involves a mechanism of flooding a message of route request. The destination selects the best route based on route selection metrics (e.g. number of hops).The process is complete once a route is found or all possible route permutations have been examined (Royer and Toh, 1999). Once a route is established, it is maintained by a route maintenance procedure until either the destination becomes inaccessible along every path from the source or until route is no longer desired. In reactive routing protocols, control communication overhead is greatly reduced compared with proactive routing protocols since no effect is made to maintain the total network topology. Numerous protocols of this type have been proposed, such as Dynamic Source Routing (DSR) (Johnson and Maltz, 1996; Johnson *et al.,* 2001_a and Johnson *et al.,* 2001_b), Ad-hoc On Demand Distance Vector (AODV) routing (Perkins and Royer, 1999 and Perkins *et al.,* 2000), Associativity-Based Routing (ABR) (Toh, 1996), Ad-hoc On-demand Multipath Distance Vector (AOMDV) algorithm (Marina and Das, 2001), Multipath Dynamic Source Routing (MDSR) (Nasipuri and Das, 1999), etc.

The main advantage of the source-initiated protocols is that no routing table updating is required unless a route is used or broken. Therefore, the battery power and wireless bandwidth can be conserved efficiently. However, when a route is needed, the source node needs to initiate a route query by flooding the network with query packets which can lead to excessive routing delay. Furthermore, an efficient route finding mechanism is required in order to prevent overloading the network with query packets.

The initial approach for routing in MANETs was proactive, i.e. the protocol keeps track of routes in the network and this requires the protocol to exchange control messages at a regular time interval. However, channel bandwidth was wasted during these exchanges and this main issue raised the need to prioritise the use of reactive protocol, where routing is performed only on-demand. Various simulations have shown that on-demand protocols perform better in ad-hoc networks than table-driven protocols (Broch *et al.,* 1996; Jiang and Garcia-Luna-Aceves, 2001; Das *et al.,* 2000_b; Chin *et al.,* 2002; Celebi, 2001). The first real world implementations of these protocols were deployed recently (Maltz *et al.* 1999_a; Maltz, 2001; Desilva and Das, 2000; Dupcinov *et al.,* 2002). This led to use on-demand routing technique in our study. Ad-hoc On-demand Distance Vector (AODV) and Dynamic Source Routing (DSR) protocols are such algorithms. Hence the need to describe these basic mechanisms in the following sub-sections.

2.4.3.1 Ad-hoc On-demand Distance Vector (AODV)

AODV (Perkins and Royer, 1999; Perkins *et al.,* 2000) is an on-demand routing protocol i.e. routes to the destination are only discovered when required thus avoiding memory overhead and consuming less power. Moreover, a node using AODV does not have to discover and maintain a route to another node until the two nodes need to communicate with one another. AODV uses destination sequence number which is generated by the destination itself for each route entry. The destination sequence number avoids loops and if two similar routes to a destination exist, then the node chooses the one with the highest sequence number. AODV uses *Route Request (RREQ)*, *Route Reply (RREP)*, and *Route Error (RERR)* messages for route discovery and maintenance. The functioning of AODV is composed of *Route discovery* and *Route maintenance*.

Route discovery

Route discovery allows any host in the ad-hoc network to dynamically discover a route to any other host in the ad-hoc network, whether directly reachable within wireless transmission range or reachable through one or more intermediate network hops through other hosts. When a source wants to send information to a destination and does not have a route it generates a RREQ packet and broadcasts the packet to its neighbours. The RREQ uses the following fields in its packet: *Hop Count*, *RREQ ID*, *Destination IP Address*, *Destination Sequence Number*, *Originator IP Address*, and *Originator Sequence Number*. The hop count is the number of hops from the source to the node handling the RREQ. Thus, when node receives a RREQ, if it is not the destination and nor does it have path to the destination it increments the hop count by 1 and rebroadcasts the packet to its neighbours. The destination IP address and the originator IP address are the addresses of the destination, and source generating the RREQ respectively. RREQ ID is a number that uniquely identifies the RREQ. If the RREQ ID in the RREQ packet matches the RREQ ID in the nodes route entry table the RREQ will be dropped. Destination sequence number is the greatest sequence number received in the past by the originator for any route towards the destination and it indicates the freshness of a route.

Figure 2.6 illustrates an example of a mobile ad-hoc network and how RREQ are flooded in the network. In this example, source node 1 wants to send data packets to destination node 3, the routing process using AODV is described in Figure 2.7.

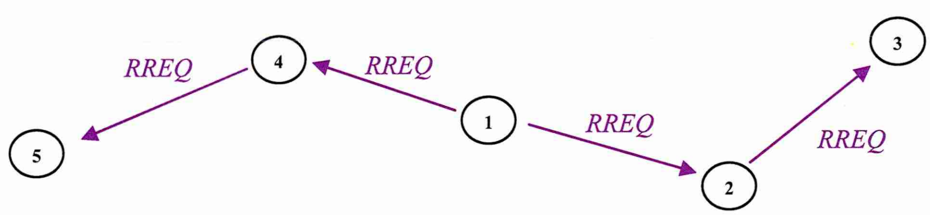


Figure 2.6: Propagation of RREQ Packet

The route table maintains entries for each destination node it is interacting. The routing table has the following fields: *Destination IP address*, *Active neighbours*, *Number of hops*, *Next hop*, *Destination Sequence Number*, and *Expiration time for the routing table entry*. They help the node to maintain the connectivity of the network. The expiration time associated with the route depends on the size of the ad-hoc network and indicates the time after which the route to that associated destination in the route table is to be removed. The node maintains the list active neighbours that are the next hop to the destination associated in the route table, thus if a link to this active neighbour is broken then the node can immediately broadcast RERR messages.

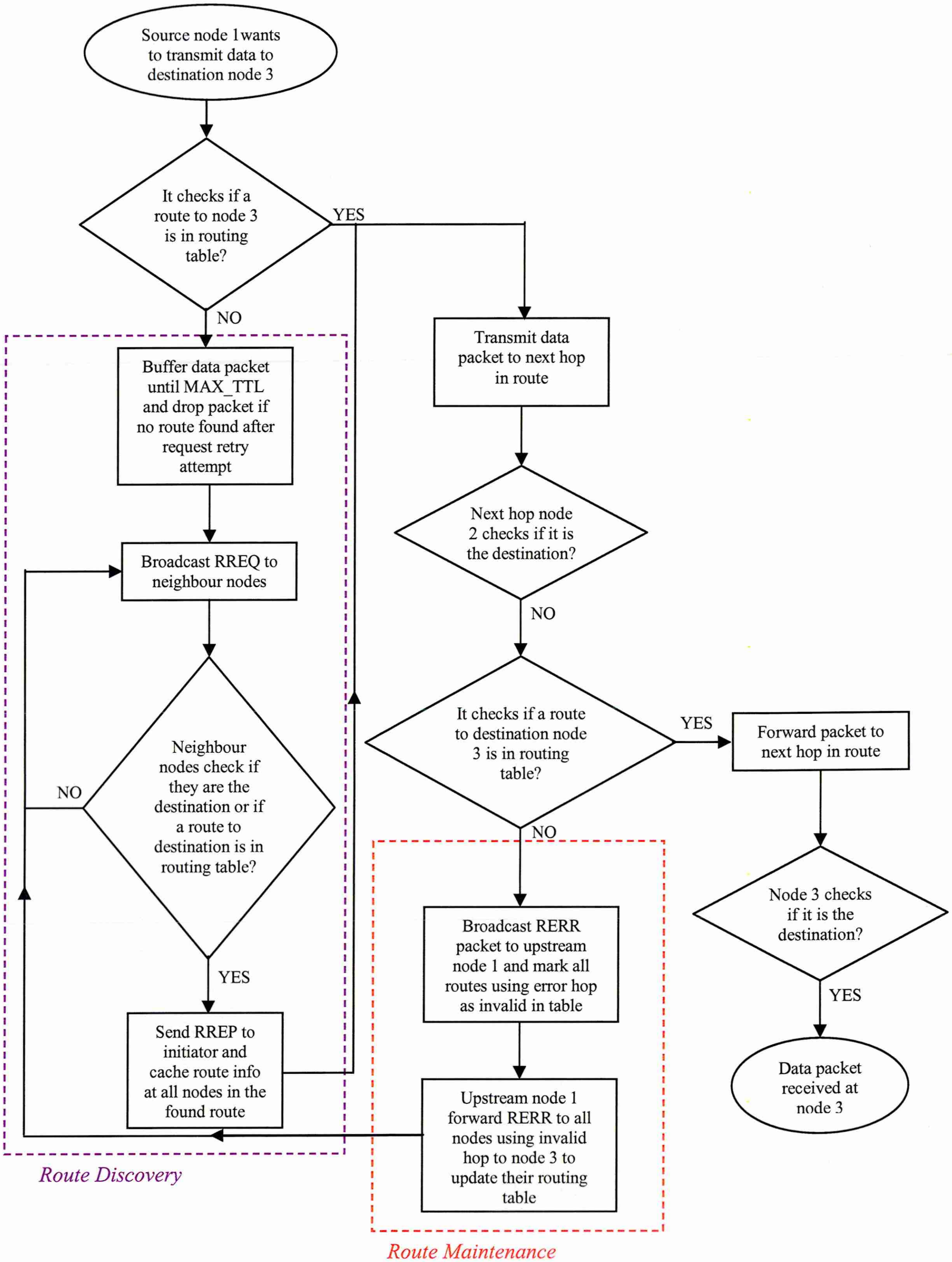


Figure 2.7: Routing process used in AODV

Route Maintenance

Route maintenance is the mechanism by which the source node is able to detect, while using a route to the destination, if the network topology has changed such that it can no longer use its route to the destination node because a link along the route is no longer available. When *route maintenance* indicates a source route is broken, it can invoke the route discovery to find a new route. A node broadcasts RERR packets when a link to the next hop is broken. This is how AODV reacts to link failures. Thus, the node adds the destination addresses that are unreachable due to the link failure in the RERR packet and broadcasts it to its neighbours. A RERR message is processed only when a node detects a link break for the next hop for which it has an active route in its routing table or when it receives a RERR from the neighbour for an active route it has in its routing table or when it gets a data packet for a destination and it does not have an active route to the destination. Under these circumstances the node sends out RERR messages to its neighbours. A node keeps track of its neighbour through *hello messages* that each node broadcast at set intervals. These messages contain the nodes identity and sequence number to its neighbours so that its neighbours can update their local connectivity to the node that broadcast the *hello message*. It can assume that the link is broken and can broadcast a RERR packet to its neighbours regarding the link failure. Other methods to maintain link connectivity are used, like physical and link layer methods to detect link breakages to nodes that it considers neighbours (Perkins and Royer, 1999).

2.4.3.2 Dynamic Source Routing (DSR)

Dynamic Source Routing (Johnson and Maltz, 1996; Johnson *et al.,* 2001_a) is another conventional on-demand routing protocol. DSR is one of the most well-known routing algorithms for ad-hoc wireless networks. It was originally developed by Johnson, Maltz, and Broch (Johnson *et al.,* 2001_b). DSR uses source routing, which allows packet routing to be loop free. It increases its efficiency by allowing nodes that are either forwarding route discovery requests or overhearing packets through promiscuous listening mode to cache the routing information for future use. DSR is also on demand, which reduces the bandwidth usage especially in situations where the mobility is low. It is a simple and efficient routing protocol for use in ad-hoc networks. Similar to AODV, it is composed of two mechanisms, which are requested only when two nodes want to communicate with each other, namely, *Route discovery* and *Route maintenance*. Along with those mechanisms, DSR allows multiple routes to any destination, thus can lead easily to load balancing or increase robustness.

Route discovery

Using the example of Figure 2.6, the routing process used in DSR is described in Figure 2.8. If the source wants to transmit a packet, it has to check its "route cache" to determine whether a suitable route for the destination, also called target, exists. If no route is found, it will have to start a route discovery to find a route to the target. The route discovery itself consists of a chain of locally broadcasted Route Request (RREQ) packets. The data structure of RREQ consists of two fields: IP fields and route request fields. IP fields contains *source address*, *destination address* and *hop limit*. Route request fields contain *option type, option data length, identification, target address, and route record*. Each route request packet contains a unique *request id*, set by the initiator from a locally-maintained sequence number. In order to detect duplicate route requests received, each host in the ad-hoc network maintains a list of the initiator address, request id pairs that it has recently received on any route request. The host may buffer the original packet in order to transmit it once the route is learned from route discovery, or it may discard the packet, relying on higher-layer protocol to retransmit the packet if needed. Each entry in the route cache has associated with it an expiration period, after which the entry is deleted from the cache.

Route Maintenance

The route maintenance is performed using route error packets (RERR) and acknowledgments (ACK). RERR is sent whenever a fatal transmission problem occurs. The nodes receiving RERR will delete the entry of the error hop in their route caches. ACK is used to verify the availability of route links. The result of ACK will be used to update route caches in order to reflect the current topology. If the data link level reports a transmission problem for which it cannot recover (for example, because the maximum number of retransmissions it is willing to attempt has been exceeded), this host sends a *route error* packet to the original sender of the packet encountering the error. The route error packet contains the addresses of the hosts at both ends of the hop in error: the host that detected the error and the host to which it was attempting to transmit the packet on this hop. When a route error packet is received, the hop in error is removed from this host's route cache, and all routes which contain this hop must be truncated at that point. A node may salvage a packet if it knows another route to the packet's destination. However, a count is maintained in the packet of the number of times that it has been salvaged, to prevent a single packet from being salvaged endlessly (Johnson and Maltz, 1996).

Route maintenance is also illustrated in Figure 2.8.

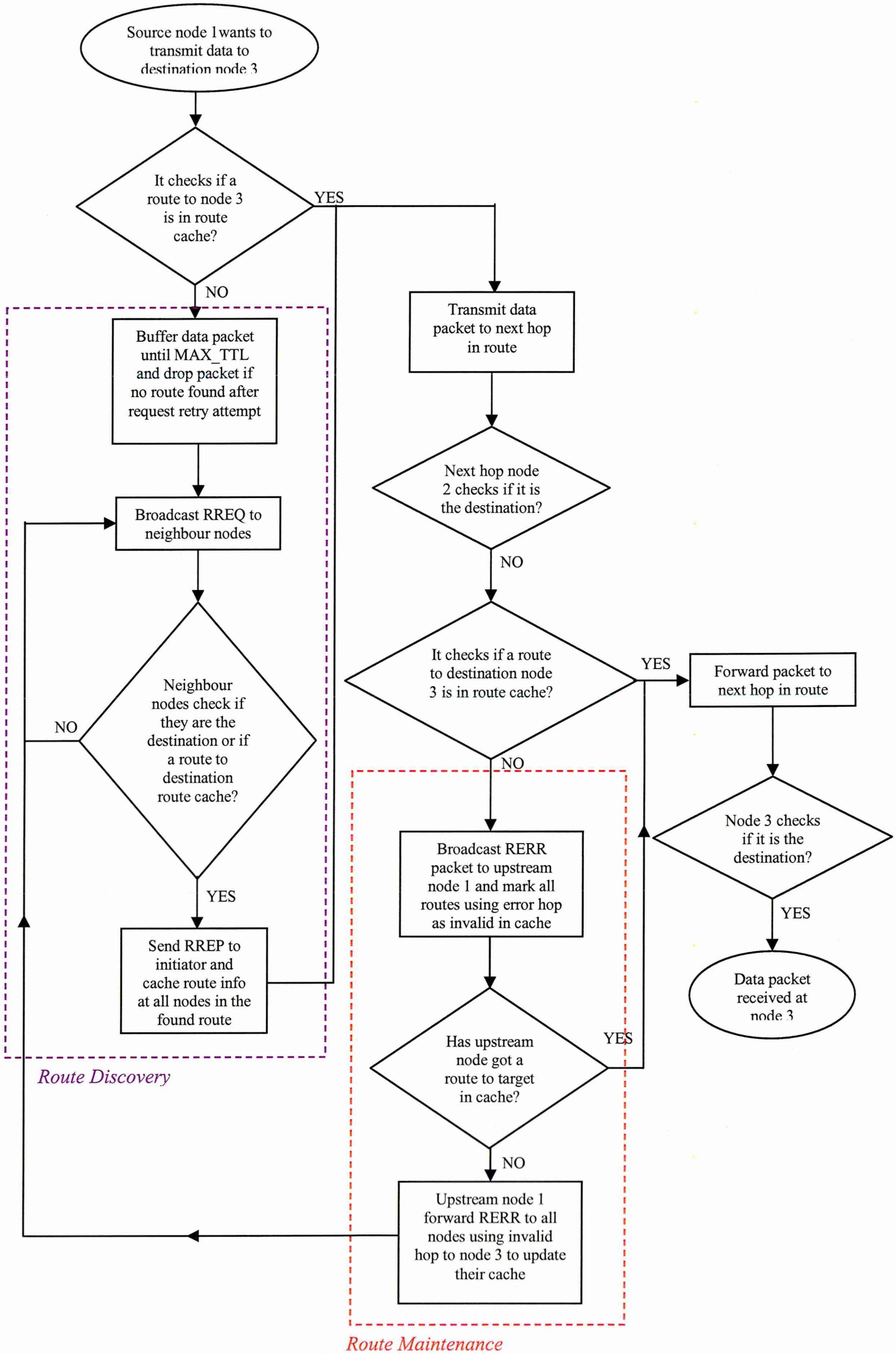


Figure 2.8: Routing process used in DSR

2.4.4 Hybrid or Hierarchical Protocols

The hybrid or hierarchical topology method incorporates some aspects of proactive and reactive protocols. As an example of a hybrid MANET routing protocol, ZPR (Joa-Ng and Lu, 1999_a) defines a zone around each node where the local topology is proactively maintained via the Intrazone Routing Protocol (IARP) (Haas *et al.,* 2001_b). When routes are required outside the local zone, a reactive route discovery mechanism is used via the Interzone Routing Protocol (IERP) (Haas *et al.,* 2001_a). In hierarchical routing algorithms, a set of nodes is divided into clusters. Each cluster has a node, which is designated as the cluster-head. So, every node is either a cluster head or one wireless hop away from the cluster head. Here, each cluster head maintains information about other nodes in its cluster, and from time to time, this information is exchanged between cluster heads over the network. Thus, the cluster heads gather network topology information. A node that has a packet to send to another node can obtain routing information from its cluster head.

Other examples of hybrid protocols are the Temporally Ordered Routing Algorithm (TORA) (Park and Corson, 1997), and the Landmark Routing Protocol (LAN-MAR) (Pei *et al.,* 2000).

2.5 Quality-of-Service (QoS)

This section gives a definition of QoS, how it will be considered in this study, also included is a description of current QoS protocols. Finally we define some intrinsic QoS parameters and their importance for QoS provisioning during multimedia applications transmissions.

2.5.1 Definition

The Internet Engineering Task Force (IETF) (Crawley *et al.,* 1998) defines QoS as 'a set of service requirements to be met by the network while transporting a flow'. Here a flow is a packet stream from a source to a destination (unicast or multicast) with an associated QoS. It is a quantitatively defined performance between the service provider and user applications based on the connection requirements. Franken (1996) defines it as a control mechanisms that can provide different priority to different users or data flows, or guarantee a certain level of performance to a data flow in accordance with requests from the application program. In this study, QoS refers to the ability of the network to provide the required QoS level in terms of delay, jitter, loss, and packet delivery fraction (i.e. throughput) for the transmitted applications.

Depending on the field of study concerned, the term *quality* has various different meanings. In the telecommunication area, it is employed to assess whether the service satisfies the end-user's expectations (Gozdecki *et al.,* 2003). However, it is dependant on who is judging the level of service: an end user may give an opinion of the service based on his experience, whereas the network designer may evaluate the service according to a number of technical factors. Consequently, there are many meanings for QoS, which can lead to misunderstanding. In (Hardy, 2001), three concepts of QoS are presented which can be used to clarify this confusion: intrinsic QoS, perceived QoS and assessed QoS. From a technical perspective, intrinsic QoS includes service characteristics. It is a technical measure considered by engineers, designers and network providers. Moreover, intrinsic QoS concerns the network architecture and its development, dependability, and effectiveness (Hardy, 2001). Hence, the performance is measured and compared to an expected level of service without considering the end-user opinions. Perceived QoS reflects the end user's opinion about a service. It is evaluated by making a comparison between the end-user's expectations and the practical performance. As a result, perceived QoS is subjective to the user's experience, a service with the same intrinsic QoS characteristics may have different perceived QoS with different clients (Zhang, 2004). The end-user may make a decision on whether to keep using a service on not based on the assessed QoS. This choice is made based on the perceived quality, service price, and how effective issues and complaints have been dealt with by the provider. In this study only intrinsic QoS is considered.

2.5.2 Quality of Service Protocols in Communication Networks

At present, information is most often forwarded using the best effort IP (Internet Protocol). All traffic will be delivered as quickly as possible regardless of requirements on packet loss, throughput or delay. While the best effort forwarding may be sufficient for most applications, QoS support is required to assure the increasing requirements of multimedia applications, e.g. video on-demand or IP telephony. Over time, "the variation in ad-hoc resources (e.g., bandwidth and battery power) is omnipresent" which makes current wired networks' QoS inappropriate for a MANET, hence new QoS models must be defined. Existing QoS models can be classified into two types according to their fundamental operation; the Integrated Services (IntServ) architecture

(Braden *et al.,* 1994) and the Differentiated Services (DiffServ) framework (Blake *et al.,* 1998). Specifically, IntServ is based on resource reservation and admission control, it offers per flow end-to-end reservations on the opposite DiffServ provides hop-by-hop traffic packet differentiation by categorising flows into a number of service levels. Both of these protocols do not perform very well in ad-hoc networks, as they both need precise information on the topology and link states (e.g., delay, available bandwidth). Table 2.2 presents a summary of pros and cons of IntServ and DiffServ.

Table 2.2: Comparison of IntServ and DiffServ in MANETs

QoS Models	Pros	Cons
IntServ	• Connection oriented • Enable resource guarantees • End-to-end per-flow QoS guarantees	• Every router must maintain specific state info for every flow • If insufficient resources are available, flow is denied admission in network • Not scalable • Hard QoS is not possible in MANETS since flow states change over time
DiffServ	• Packets are marked and classify by edge routers • Intermediate routers just forward based on marking • No state information to be maintained by routers • More scalable	• Limited resources availability • Imprecise state information due to link state changes causing data loss and delays

Due to the limitations highlighted in Table 2.2, novel mechanisms were needed to enhance QoS in MANETS, below a few examples are presented.

IEEE 802.11e: The IEEE 802.11 standard defines two channel access mechanisms, called coordination functions (IEEE, 1997). These coordination functions determine when a station is permitted to transmit, and when it must be prepared to receive data. The mandatory function is the DCF which adopts *CSMA/CA* mechanism to provide services for asynchronous data transmission. The alternative function is the PCF (Point Coordination Function) which is proposed for use with real time traffic, since IEEE 802.11 is not able to maintain QoS in ad-hoc networks. To provide MAC-level QoS assurance at the link layer, the IEEE 802.11 Working Group is currently developing the IEEE 802.11e standard (Xiao, 2004 and Sung and Yun, 2006). This standard provides QoS features to the existing 802.11b and 802.11a standards and it still retains interoperability with these standards (BreezeCOM, 1997). The IEEE 802.11e MAC establishes two new coordination functions: Enhanced DCF (EDCF) and Hybrid CF (HCF). The EDCF works in Contention Period (CP) only while the HCF works in both Contention Free Period (CFP) and CP. EDCF is an enhanced version of DCF access method to provide service differentiation. In IEEE 802.11e, traffic is categorised into service classes based on various QoS parameters such as initial window sizes, maximum window sizes, and interframe spaces. For example, high priority classes are forwarded with a short Contention Window (CW) to make sure that they are transmitted before the lower priority classes (Grilo and Nunes, 2002). With this method, high priority traffic benefits from a better service whilst maintaining a minimum service for low priority traffic.

FQMM: Flexible QoS Model for MANETs (FQMM) (Xiao *et al.,* 2000) was proposed to define a MANET QoS model based on both IntServ and Diffserv functions and mechanisms. FQMM defines three types of nodes: an ingress node which sends data, an interior mode which forwards data to other nodes, and an egress node which is a destination. Each node may have multiple roles. This model selectively uses the per-flow state property of IntServ, and the service differentiation of DiffServ. That is to say, for applications with high priority, per-flow QoS guarantees of IntServ are provided. On the other hand, applications with lower priorities are given per-class differentiation of DiffServ. Therefore, FQMM applies a hybrid provisioning where both IntServ and DiffServ scheme are used separately.

2.5.3 Multimedia Applications QoS

The growth of multimedia applications over wide area networks has increased research interest in QoS. An overview of providing QoS over the internet is available in (Xiao and Ni, 1999). The communication delay and synchronization needed for voice, data and images are major concerns. Internet telephony (Voice over IP) and other multimedia applications such as video conferencing, video-on-demand, and media streaming require service guarantees and are very time-sensitive. Just increasing the amount of resources such as available bandwidth to avoid congestion does not provide proper resource utilisation. This over-provisioning solution has been effective in applications such as file transfer (FTP), web browsing (HTTP), and email. However, the nature of traffic over the internet has changed in its characteristics. The Internet is becoming the backbone of future communications in an entertainment centre. Multimedia applications require network services beyond what IP delivers, hence the need for new solutions.

In order to facilitate QoS support in MANETs, the most important aspect is to identify the QoS requirements and measure them. In communication networks, transmitted traffic characteristics are represented, in a very general way, by four primary parameters (or metrics): loss (unreliability), delay, jitter (delay variation) and bandwidth. Together, these determine the QoS requirements of the traffic (Farkas *et al.,* 2006). For instance, jitter is an important QoS parameter for IP telephony, which can tolerate a certain percentage of packet loss without any degradation of quality. For data transfer, loss is a crucial QoS parameter.

In general, data can be classified as follows (Tanenbaum, 2003; Abdullah *et al.,* 2003):

- *Delay-sensitive traffic:* Most real time applications (video, audio or voice) fall into this category. Some real time traffics are more sensitive to both end-to-end delay and delay variations (jitter) such as telephony and video conferencing while others tolerate the former but not the latter such as audio and video on demand.
- *Loss-sensitive traffic.* File transfer and email fall into this category. In this type of traffic, no bits may be delivered incorrectly. If a packet is damaged during transmission, it is not acknowledged and will be retransmitted.
- *Security-sensitive traffic.* Examples are: money transactions and confidential applications.
- *Bandwidth-sensitive traffic.* Examples are: video on demand (VoD).
- *Multi-sensitive traffic.* This type of traffic is associated with certain multimedia applications or when certain traffic is sensitive to more than one metric. For example, confidential email can be considered as loss-security- sensitive, confidential video conferencing can be considered as delay-security sensitive.

Traffics can be described as time-based and non-time-based information (Kwok, 1995), time-based (i.e. real-time applications) information is sensitive to time varying as video, and voice, non-time-based (i.e. non-real-time application) is stored (perhaps temporarily) at the receiving points for later consumption, thus is insensitive to time varying as image, and data. Its traffic characteristics and the corresponding communication requirements can characterise an application. Constant Bit Rate (CBR) is intended for real-time applications requiring tight constraints on delay and delay variation (Crowcroft *et al.,* 1995; Garret, 1996), such as voice and video applications which are expected to transmit at a continuous rate.

Table 2.3: Example of common applications and the sensitivity of their QoS requirements

Applications	QoS requirements			
	Loss	Delay	Jitter	Bandwidth
Email	High	Low	Low	Low
File transfer	High	Low	Low	Low, Medium, High
Telephone	Low	High	High	Low
Video-on-demand	Low	Low	High	High
Videoconference	Low	High	High	High

In this study, the characteristics and requirements of voice and video are considered. For each application, the characteristics in terms of data rate and packet size are described during the simulation process. Additionally, the requirement outlined in Table 2.2, in terms of delay, jitter, throughput and packet loss, are considered for the evaluation of the QoS.

2.5.4 QoS Parameters

A definition of the different QoS parameters considered in this thesis is given in this section and a summary of QoS requirements for these applications is provided in Table 4.4 (ITU, 2001; Fluckiger, 1995). These have to be well understood in order to determine the appropriate treatment to give to each application in the network.

Table 2.4: QoS requirements for audio and video applications

TYPE	APPLICATION	TYPICAL DATA RATES	PARAMETERS		
			DELAY	JITTER	PACKET LOSS
Audio	IP telephony	4-64 kbps	Less than 150 ms preferred Less than 400 ms tolerable	Less than 1 ms preferred	Less than 3%
Video	Video on-demand	16-384 kbps	Less than 150 ms preferred Less than 400 ms tolerable	Less than 50 ms preferred	Less than 3%

Packet delivery fraction of data packets

Packet delivery ratio is defined as the ratio of the data packets delivered to the destination to those generated by the Constant Bit Rate (CBR) sources. Packet delivery ratio is a very important metric since it describes the loss rate that will be seen by the transport protocols, which in turn affects the maximum throughput of the network. It is computed according to Equation 2.1 (Wang *et al.,* 2000).

$$Pdf = \frac{R}{S} * 100 \qquad 2.1$$

Where,

Pdf = the total packet delivery fraction in percent

S = total number of generated packets

R = the total number of successfully received packets.

Average end-to-end delay of data packets

The average end-to-end delay includes all possible delays from the moment the packet is generated to the moment it is received by the destination node. Generally, there are three factors affecting end-to-end delay of a packet: (1) Route discovery time, which causes packets to wait in the queue before a route path is found; (2) Buffering waiting time, which causes packets to wait in the queue before they can be transmitted; (3) The length of routing path. The more number of hops a data packet has to go through, the more time it takes to reach its destination node. In this study, end-to-end delay and average end-to-end delay are computed according to Equations 2.2 and 2.3, respectively (Michaut and Lepage, 2005).

$$\textit{End-to-end delay} = (tr_i - ts_i) \qquad 2.2$$

$$\textit{Average end-to-end delay} = \frac{1}{R}\sum_{i=1}^{R}(tr_i - ts_i) \qquad 2.3$$

Where,

R = Number of successfully received packets

i = Unique packet Identifier

tr_i = Time at which a packet with unique identifier i is received

ts_i = Time at which a packet with unique identifier i is sent

Jitter or delay variation of data packets

While network latency effects how much time a real-time packet spends in the network, jitter controls the regularity in which real-time packets arrive. Typical real-time sources generate packets at a constant rate. The destination expects incoming real-time packets to arrive at a constant rate. However, the transmission delay by the hop-by-hop network may be different for each packet. The result is that packets that are sent with equal intervals from a source node arrive with irregular intervals at a destination node. Jitter is calculated based on the delay difference of successive packets for each flow. For many multimedia applications a high amount of jitter is worse than a high amount of delay because it introduces some degree of unpredictability into the data. It has a significant effect on real-time or delay-sensitive applications such as voice and video applications. In this study, jitter and average jitter are computed according to Equations 2.4 and 2.5 (Michaut and Lepage, 2005), respectively.

$$J_i = |D_i - D_{i-1}| , \qquad i \rangle 0 \tag{2.4}$$

$$Average \;\; jitter = \frac{1}{R}\sum_{i=1}^{R} J_i \tag{2.5}$$

Where,

J_i = the absolute values of jitter in (second) of the i^{th} packet,

D_i-D_{i-1} = difference in delays of two consecutive i and i-1 packets, D_i is obtained from Equation 2.2

R = is the number of successfully received packets.

2.6 QoS-Based Routing

Extensive work has been carried-out over QoS-based routing. The development and evaluation of various QoS routing protocols have been explored in details. However, there are still many issues in this area that need further investigations, such as how to consider multiple QoS constraint to find a route that would improve the QoS delivery of various multimedia applications, especially real-time applications which are time-bounded. As the main focus of this study is based on network performance for time sensitive applications such as audio or video, it is essential to review the previous efforts that have been made in the research areas of QoS routing protocols in MANETs.

2.6.1 Definition

A definition of QoS based routing is outlined in (Sun, 2000) as:" A routing mechanism under which paths for flows are determined based on some knowledge of resource availability in the network as well as the QoS requirement of the flows". In short it is a dynamic routing scheme with QoS consideration. Routing consists of two basic tasks. The first task is to collect the state information and keep it up to date. The second task is to find a feasible path for a new connection based on the collected information. The QoS requirement of a connection is given as a set of constraints, which can be *link constraints* or *path constraints* (Lee *et al.,* 1995). A *link constraint* specifies a restriction on the use of links. A bandwidth constraint of a unicast connection requires, for instance, that links composing the path must have a certain amount of free bandwidth available. A *path constraint* specifies end-to-end QoS requirements on a single path. The challenge is to guarantee the QoS not only over a single hop but also over an entire wireless multi-hop path. This requires the propagation of QoS information within the network. Three types of constraints on the path can be identified (Wang and Hou, 2000; Chen and Wu, 2003):

Additive metrics: an additive metric has the following form

$$m(p) = \sum_{i=1}^{K} m(k_i) \tag{2.6}$$

Where $m(p)$ is the total of metric m of route p, k_i is the *ith* link in the route p, and K is the number of links in route p. The link metric $m(k_i)$ is determined based on the QoS parameters, such as delay, delay variation (jitter) and cost. The end-to-end delay (delay variation) is the accumulation of all delays of the links along the path.

Concave metrics: a concave metric has the following form

$$m(p) = \min(m(k_i)) \tag{2.7}$$

Bandwidth is the most common example of this type of metric. Bandwidth is concave in the sense that the end-to-end bandwidth is the minimum of all the links along the path.

Multiplicative metrics: a multiplicative metric has the form

$$m(p) = \prod_{i=1}^{k} m(k_i) \tag{2.8}$$

Loss probability is an example of this metric.

However, Wang and Crowcroft (1996) proved that multi-constraint based routing with two or more additive (e.g. delay) or multiplicative (e.g. bandwidth) QoS metrics is a Non-deterministic-Polynomial (NP) problem (also refereed as a NP-complete), therefore, they are considered to be intractable for large networks. Accordingly, many heuristics solutions have been proposed for these problems. Wang and Hou (2000) provide a list of twelve combinations with multiple constraints. Approximation methods exist for QoS constraints that are NP-complete (e.g. *sequential filtering*) (Chen and Nahrstedt, 1998_a).

In considering QoS routing, it is important to distinguish the following two concepts (Chen, 1999):

- **Routing information**: the mobility of the node leads to a more complex task for routing. Indeed, in most cases, nodes will move frequently. The changes of topology imply that a dynamic routing protocol needs to maintain routes between sources and destination. It is important to make the routing decision based on correct and updated information about the topology and the states of the links of the network.
- **Routing algorithm**: the algorithm used to make a routing decision in order to find a feasible path for a new connection (or an existing one in case of link failure problems) based on the collected information. The routing algorithm has to manage the mobility of the nodes which move randomly and unpredictably in the network. In order to achieve this task, nodes might store information concerning the topology of the network in their routing table. Most routing protocols for MANETs, such as AODV and DSR are designed without explicitly considering QoS of the routes they generate and are based on a single metric such as shorter number of hops.

Hence, the success of the ad-hoc routing protocol depends on both the quality of the information collected by the node and the efficiency of the routing protocol. The performance of any routing algorithm directly depends on how well the first task is solved (Chen and Nahrstedt 1998_c). Even after establishing a route that satisfies the QoS requirements, this route is hard to guarantee due to frequent change in topology. The size of an ad hoc network is also a problem if it is large, because the computational load will be high, and it will be difficult to propagate network updates within given time bounds.

Unlike the protocols discussed in the section 2.4, constraint-based routing protocols use metrics other than shortest-path for finding a suitable and feasible route. Figure 2.9 shows a wireless topology (Li, 2006). The mobile nodes are labelled as A, B, C,..., and K. The numbers beside each edge represent the available bandwidths of the wireless links. Assume, A is the source node and G the destination node, for conventional routing using shortest path (in terms of the number of hops) as metric, the route A-B-H-G will be chosen. It is quite different in QoS route selection. Consider the bandwidth as the QoS metric and the desire to find a route from A to G with a minimum bandwidth of 4. Now the feasible route will be A-B-C-D-E-F-G. The shortest path route A-B-H-G will not be suitable for providing the required QoS.

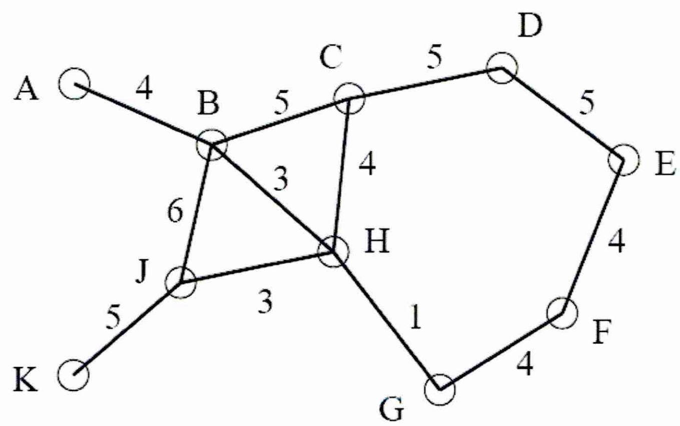


Figure 2.9: An example of QoS routing in ad hoc networks

The characteristics of good QoS routing algorithms include the following aspects (Li, 2006):

- The routing algorithm should compute an efficient route that can satisfy the QoS requirements with a very high probability, if such a route exists.
- The QoS route computation algorithm should be simple and robust.
- With the change in network dynamics, the propagation and updates in state information should be kept to a minimum.

Various QoS routing algorithms have been proposed to resolve the QoS provisioning problem in ad-hoc networks. A detail introduction to QoS routing in ad-hoc networks is available in (Chakrabarti and Mishra, 2001; Pragyansmita and Raghavan, 2002; Chen and Nahrstedt, 1998_a). Some QoS extensions have been implemented based on AODV and DSR:

- In (De Renesse *et al.,* 2004) , QoS fields have been added to the route discovery packet (RREQ) to enable QoS route selection based on minimum, maximum delay and bandwidth requirement. This indicated that average packet delay and bandwidth utilisation are improved compared to the standard AODV. However, on-going traffic sessions were sometimes paused, thus reducing packet delivery ratio in the search of optimal paths.
- Multipath is another method used to improve bandwidth utilisation which is based on finding multiple paths that satisfy the demand on the selected QoS

metric (Marina and Das 2001; Abidi and Erfani, 2006; Leung *et al.,* 2001). This method provides load balancing which is a very important feature especially in MANETs, where bandwidth is limited, thus reducing congestion and bottlenecks. Although multipath reduces overhead during link breaks, it can incur an overhead if route selection is based on several metrics (Mueller *et al.,* 2004)

2.6.2 The Multi-Constraints Routing Problem (MCP)

2.6.2.1 Definition

Multi-constraints QoS routing finds a path that satisfies multiple independent path constraints. One example is the delay-cost-constrained routing, i.e., finding a route in the network with bounded end-to-end delay and bounded end-to-end cost. The delay-cost-constrained routing is an example of a 2-constrained routing problem. However, the algorithms to solve such a problem have been shown to have, in general, high computational complexity. Several approaches have been proposed to address the complexity of multi-constrained path computation problem. As mentioned in section 2.5.1, multi-constrained QoS routing is known to be NP-complete (Jaffe 1984) and (Wang and Crowcroft, 1996). Previous work (Chen and Nahrstedt, 1998_a; Chen and Nahrstedt, 1998_b; Jaffe, 1984; Korkmaz *et al.,* 2000) have focused on developing heuristic algorithms to solve 2-constrained problems. In the worst case, the time complexity of the algorithm may grow exponentially with respect to the network size. Algorithms in (Chen and Nahrstedt, 1998_b) find approximate solutions in polynomial time.

Many QoS routing algorithms incorporating a variety of constraints have been proposed in the past few years (Kuipers *et al.,* 2002; Chen and Nahrstedt, 1998_a; Curado and Monteiro, 2004). For unicast routing, the Multi-Constraint Optimal Path (MCOP) (also known as PCPO) and the MultiConstraint Path (MCP) problems are the most notorious ones for their NP-complete property. MCOP routing is to find a path satisfying the path constraint meanwhile the found path is optimised on another QoS metric. An example of MCOP is the Delay-Constrained Least Cost (DCLC) which consists of finding a least cost path with bounded delay (Liu *et al.,* 2005). An example of MCP is to find a path satisfying multiple constraint which maybe simple than MCOP since it does not optimise on any metric instead it only finds a path that meet all the constraints.

The selection of QoS paths subject to multiple constraints can be defined as the Multi-Constrained Path (MCP) problem. In order to present the MCP problem, some other useful definitions are introduced: as is common practice in the literature, a network is modelled as a connected directed graph (digraph) $G(V,E)$ composed of a set of vertices (V) which represent the set of network nodes and a set of edges (E) representing physical or logical connectivity between nodes. The number of vertices of G is given by $n = |V|$ and the number of edges is given by $m = |E|$. Each edge, is represented by the link between two vertices $e = (u,v)$ and has associated q weights corresponding to QoS metrics such that $w_i(u,v) >= 0$, and $i = 1,2,...,q$. The constraint for each QoS metric is L_i. The *Multi-Constrained Path problem* is to find a path P from a source s to a destination d such that all the QoS constraints are met, as depicted in Equation 2.9:

$$w_i(P) \leq L_i, \qquad i=1,2,...,q \tag{2.9}$$

The paths that satisfy these constraints are called *feasible paths* (Kuipers *et al.,* 2002) and a definition and a survey of the problem and some proposed solution are available by the same reference. The solution of the MCP problem requires a path computation algorithm that finds paths that satisfy all the constraints as expressed in Equation 2.9. Since the optimal solution of this type of problems for multiple additive and independent metrics is NP-complete, general approaches tackling this problem are approximation or heuristic algorithms that guarantees to find a near optimal solution with polynomial complexity. Algorithms that solve the multi-constrained path problem using the above strategies are described in the following section.

2.6.2.2 Related Work

Sequential filtering is a commonly used approach to deal with MCP. When a route is needed the algorithm will first eliminate all the routes that do not comply with the first constraint (e.g. bandwidth) and the remaining pool of path will then be submitted to the second metric requirement (e.g. minimum delay) based on the Dijkstra's algorithm (i.e. shortest path). Wang and Crowcroft (1996) dealt with the NP-problem using sequential filtering considering bottleneck bandwidth and propagation delay and they argued that the only feasible combination to avoid the NP-completeness, are: bandwidth and one of the four (delay, delay jitter, cost and loss probability). A similar approach used heuristics methods based on Metric Ordering (MO), where two metrics are considered based on their priority (Curado ad Monteiro, 2004). For example, the Shortest-Widest Path (SWP) is based on finding the path with the highest available bandwidth then selects the shortest path according to the second metric (e.g. number of hops or end-to-end delay). Ma and Steenkiste (1998) proposed extensions to Dijkstra and Bellman-Ford to solve this problem based on sequential filtering. (Salama *et al.,* 1997) implemented a simplification to solve the Delay-Constrained Least Cost (DCLC) which is also known to have no tractable solutions (Garey and Johnson, 1979). However, an assumption is made that first, a selection of route is based on the least-delay value and the subset is then selected based on either least-delay or least-cost. This technique has proven to be an efficient mean to deal with the multi-constraint NP-complete problem and it also improved the performance of the network. However, it does not consider the QoS requirements simultaneously and it requires a good mathematical knowledge of graph theory. This technique present another disadvantage, as the number of metrics increases the computation overhead will increase in consequence. Dealing with mobile hosts means that the information on best routes becomes quickly outdated, thus the selected route after filtering might not be good enough or may no longer be available when the connection is established.

Metrics Combination (MC) is another approach for the solution of the MCP problem (Jüttner *et al.* 2001; Korkmaz and Krunz, 2001; Feng *et al.,* 2002). By combining a set of QoS metrics in a single metric, it is possible to use existing path computation algorithms, such as Bellman-Ford or Dijkstra. Jüttner *et al.* (2001) proposed a Lagrange Relaxation based Aggregated Cost (LARAC), it is a polynomial time solution for the DCLC routing problem (Salama *et al.,* 1997). The basic idea is first to construct an aggregated weight (λ) with a linear or non linear function using the Lagrange Relaxation technique, and then use the Dijkstra repeatedly to find a feasible path. Delay *d(P)*, weight λ (where, $\lambda \geq 0$ is also referred as the Lagrange Relaxation coefficient) and number of hops *c(P)* are the metrics combined in a single cost function shown in Equation 2.10:

$$c_{\lambda}(P) = c(P) + \lambda d(P) \qquad \textbf{2.10}$$

Even though metrics combination contributes to the simplification of path computation algorithms, it does not guarantee the QoS provisioning for each one metrics involved. To overcome this problem, there is a need to define the appropriate weight , also know as the Lagrange Relaxation coefficient used in the combination rule of metrics.

Resource Reservation is another method used to tackle the MCP problem. Zhu and Corson (2002) and Lin and Liu (1999) have implemented a solution based on Time

Division Multiple Access (TDMA) for communications. Their algorithm establishes the a guaranteed bandwidth through bandwidth calculation and allocation. QoS routes using resource reservation in small networks whose topologies change at low to medium rate. The protocol is based on on-demand routing, and builds QoS routes only as needed. In TDMA systems time is divided into slots, bandwidth is represented by the number of slots needed to be reserved in the TDMA frames. A session specifies its QoS requirements as the number of transmission time slots it needs on its route from a source to a destination, which then allows for a route to be reserved and ultimately meet the bandwidth constraint. Du and Pomala-Raez (2004) and Xue and Ganz (2002) introduced a similar method. They used resource reservation based routing and signalling protocol. Xue and Ganz (2002) introduce Ad-hoc QoS on-demand routing (AQOR) which integrates: on-demand route discovery between source and destination, signalling functions for resource reservation and maintenance, and hop-by-hop routing. The best route available in terms of the smallest delay with a bandwidth guarantee is chosen. Resource reservation is used to guarantee the availability of the resources to the requesting flow, at each node along the path. If end-to-end delay violation is detected at the destination, the QoS recovery will be triggered and the flow is then automatically adjusted without notice of the higher layer application. Du and Pomalaza-Raez (2004) presented an algorithm to calculate minimum end-to-end delay and used it as route selection criteria. Associated with node location information only routes that provide the required resources are selected. They demonstrated that resource consumption is more efficient by minimising the unnecessary signalling and stopping the sessions that cannot meet the demanded QoS requirement.

In conclusion, these protocols produced higher throughput and lower delay than the best effort protocol. However, resource reservation results in limited use of the network for other flows, which could reduce overall QoS, especially to traffic which are time-sensitive rather than bandwidth bounded. Moreover deciding slot assignment at the same time as available bandwidth calculation is searched along the path is a NP-complete problem, so heuristic approaches are needed to resolve the issue (Lin and Liu 1999).

Finally, other approaches that deal with the NP problem worth mentioning are based on ***imprecise network state model.*** The dynamic nature of an ad-hoc network makes the available state information inherently imprecise. Though some algorithms were required to work with imprecise information, they require precise information about the network topology, which is not available in an ad-hoc network (Guerin and Orda, 1999). Therefore, the selection of a route will typically be performed based only on partial or approximate information. Similarly, the suitability of a given link or node to accommodate a complex connection request, e.g., with requirements for bandwidth, end-to-end delay, loss probability, etc (Lorenz and Orda, 1998 and Guerin and Orda, 1999).

Chen (1999) and Chen and Nahrstedt (1999) introduced the imprecise network state model namely, the ticket-based probing algorithm. The model provides a cost-effective method for providing QoS support based on imprecise network information. The majority of QoS routing protocols are reservation-based. Probe messages are sent through the network from the source to the destination in order to discover and reserve paths which satisfy a given QoS requirement. Multiple paths are searched in parallel to find a QoS route. Each probe from the source toward the destination carries at least one ticket to control how many alternate paths to be searched, thus minimising the routing overhead. The lower the probability of finding a route with the desired QoS requirements, the larger the number of tickets carried by the probe (i.e. more paths need to be searched). In order to maximize the chance of finding a QoS route, the state information at the intermediate nodes are collected to make hop-by-hop route decisions. Thus, in case of link breaks and unlike the re-routing technique, path-repairing technique does not find a completely new path. Instead, it tries to repair the path with local reconstructions. This hop-by-hop technique reduces routing overhead during congestion by searching multiple paths however the drawback is that the overhead will increase when more paths need to be searched to meet the QoS requirements as the number of tickets carried by the probe increases.

Guerin and Orda (1999) focused on connections with QoS requirements, and considered the two cases of bandwidth requirements and end-to-end delay guarantees. To find a path that has the highest probability to satisfy a given end-to-end delay bound (i.e. delay constraint routing) is NP-complete but various special cases can be solved in polynomial time. Heuristics algorithms were proposed to solve this problem. The idea is to transform a global constraint into local constraints by splitting end-to-end delay constraint among the intermediate links in such a way that every link in the path has an equal probability of satisfying its local constraints (Chen and Nahrstedt, 1998_c). Guerin and Orda (1999) showed that in the case of connections with only bandwidth requirements, the impact of inaccuracies is relatively minimal, i.e., "good" paths can be identified using essentially a shortest path algorithm. The same could not be said for connections with end-to-end delay guarantees, for which they found that inaccuracies had a major impact on the complexity of the path selection process that typically became intractable. Therefore, end-to-end solutions are needed to provide a QoS in terms of delay with inaccurate information.

2.7 Issues with Current QoS Routing Protocols

MANETs are likely to increase their presence in the future communication environments. The support for QoS services is thus an important and desirable component of MANETs. One challenge in creating a routing protocol for ad-hoc networks is to design a single protocol that can adapt to the wide variety of conditions that can be present in any ad-hoc network over time. Additionally to the potential variability in bandwidth, nodes in an ad-hoc network may alternate between periods during which they are stationary with respect to each other and periods during which they change topology rapidly. Conditions across a single network may also vary, so while some nodes are slowly moving, others change location rapidly. Another challenge for routing is that mobility causes the next-hop node to be disconnected as nodes move in and out of transmission range. The result is that routes are frequently broken causing extra network traffic to reconstruct the routing table. If there is a high frequency of broken links, the overhead cost of routing can dominate the traffic load causing congestion and consuming precious energy in an attempt to discover unstable pathways. Many efforts have been directed towards providing QoS routing, particularly by using variant of IntServ and DiffServ to fit into MANET environment. Resource reservation technique along with bandwidth calculation, heuristic algorithms and multipath routing have helped improving network performance compared to conventional routing protocols such AODV or DSR. However, QoS parameters were not considered with equal priority (i.e. simultaneously) due to the NP-complete problem which makes it difficult or impossible to find feasible paths.

Although significant research efforts were made to support QoS routing in MANETs by selecting or reserving routes with available resources, several issues have not yet been considered. These include:

- Most of the proposed approaches use resource reservation which might not be the most suitable solution in wireless ad-hoc networks where resource is limited.
- Most proposed approaches use approximation algorithms and sequential filtering which can increase complexity and overhead according to the number of selection metric and the size of the network.
- Most of the proposed algorithms only considered one or two QoS metrics such as delay or bandwidth.
- Various studies performed evaluation without considering a particular traffic type.
- None of the proposed algorithms considered the QoS of multimedia applications simultaneously to optimise the route selection.

In this thesis the following issues are considered to provide a sound baseline for the implementation of QoS routing in ad-hoc networks:

- Intrinsic QoS parameters are evaluated for audio and video traffic models. The QoS metrics that are considered include: end-to-end delay, jitter, packet delivery fraction, normalised routing load and hop count.
- Popular routing protocols are critically evaluated based network conditions.
- QoS parameters are assessed using fuzzy logic according to video or audio application QoS requirement and the measured state information.
- In this study, in order to deal with the NP-complete problem, the most pertinent parameters are computed into a single metric which qualitatively represented the QoS of the selected route in relation with the application requirement.
- The computed QoS metric is used to highlight possible solutions for QoS routing.

2.8 Summary

A thorough review of the current state of wireless network technologies, the various routing protocols implemented in ad-hoc networks, QoS and QoS routing provisioning in mobile ad-hoc networks has been presented. Initially, this chapter gave a general overview of wireless technologies, WLAN standards architecture and the current trends. Section 2.3 described the characteristics and applications of mobile ad-hoc networks. Multimedia applications are readily available via small affordable and portable devices thus it is even more interesting and challenging to come up with some novel ideas on how we can take advantage of all these applications available to us on the go and still benefit from the best QoS possible. Routing protocols are one of the main features from the network layer which would help on achieving this goal. Thus, Section 2.4 introduced the different current MANETs routing protocols and their characteristics and also focused on the two on demand routing technique used in this study and gave a detailed description of their mechanism. Current Internet routing protocols only provide best effort based on shortest path (i.e. routing that is optimised on a single metric, priority or hop count) which is insufficient for multimedia application transmission. The term QoS, some QoS protocols and the parameters that defined QoS for this study were discussed in section 2.5. Furthermore, in this chapter, an extensive literature review on previous studies in the area of multi-constraint QoS-based routing in MANETs has been included.

Due to the limitations of MANETs and the growth of multimedia applications, QoS routing is needed. This demand is widely recognised through the extensive studies and conferences worldwide in mobile networking by industry and academy institute. Current QoS models such as AQOR uses reservation oriented method to decide admission control and allocate bandwidth for each flow. The shortcomings are that bandwidth is considered but some other parameters such as delay and jitter are not, which are more important for time-sensitive applications. Although, AQOR improves QoS performance in terms of delivery, it would need to prove itself with real-time multimedia applications. Finally, a discussion on the issues related to QoS-based routing and the direction of this study is described in Section 2.7. As a result, this chapter has provided sufficient motivation and justification for investigating routing limitations and improvement techniques to overcome some of these drawbacks and to improve the application QoS in MANETs.

Fuzzy Logic Theory

3.1 Introduction

Fuzzy Inference System (FIS) are popular computing frameworks based on the concepts of fuzzy set theory, which have been applied successfully in several areas such as control systems, decision making, optimisation, evaluation of systems, etc. (Ross, 1997). Zadeh (1965) introduced the term fuzzy logic in his work "Fuzzy sets", which described the mathematics of fuzzy set theory. It was initially presented as a way of processing data by allowing partial set membership rather than a crisp set member or non-membership. Professor Lofti Zadeh used the fact that people do not require precise, numerical information and yet people are capable of highly adaptive control. Fuzzy logic provides a simple way to arrive at a definite conclusion based upon vague, ambiguous, imprecise, noisy or missing input information (Hudson and Cohen, 2000). Instead of an element being 100% true or false, fuzzy logic deals with degrees of membership and degree of truth instead of using Boolean logic where a value is either 0 or 1. Fuzzy logic has the capability to mimic the human mind to effectively employ modes of reasoning that are appropriate rather than exact. Fuzzy logic can model non-linear function of arbitrary complexity to a desired degree of accuracy that would be difficult or impossible to model mathematically. In most applications, a fuzzy logic solution is a translation for a human solution in which an input space can be mapped to an output space using a set of rules specified in terms of words rather than numbers.

Conventional system theory relies on crisp mathematical models of systems, such as algebraic or differential equations. The large number of practical problems present in computer networks makes the gathering of acceptable degree of knowledge needed for physical modelling difficult, time-consuming and expensive or sometimes an impossible task. In those types of systems, only a partial understanding of the underlying phenomena is available and crisp mathematical models cannot be derived or are too complex to be useful. A significant portion of information about these systems is available as the knowledge of human experts, technical administrators and designers. This knowledge may be too vague and uncertain to be expressed by mathematical functions since those models would only make sense if the inputs data and variables were accurately known (Babuska, 1998). Fuzzy logic and set theory is a modelling framework which can adequately process not only the given data, but also the associated uncertainty (Smets, *et al.* 1988). Fuzzy rule-based systems can be used as knowledge-based model constructed by using knowledge of experts in the given field of interest (Pedrycz, 1993; Yager and Filev, 1994). From the input-output view, fuzzy systems are flexible mathematical functions which can approximate other functions or just data (e.g. measurement) with a desired accuracy (Kosko, 1994; Wang, 1994). In recent years, it has been used as an evaluation and optimisation tool in the domain of networking systems, Saraireh *et al.* (2004) and Aboelela (1998)'s work are closely related to this area of study.

3.2 Fuzzy Inference Systems

A Fuzzy Inference System (FIS) essentially defines a non-linear mapping of the input data vector into a scalar output using fuzzy rules. A general view of a fuzzy logic based inference system is illustrated in Figure 3.1 (Jantzen, 1998 and Franklyn *et al.*, 1998). It can be seen that a FIS is composed of four block functions: fuzzifier, inference engine, rule base and defuzzyfier. The role of each block will be outlined in the following sections.

3.2.1 Fuzzification

As illustrated in Figure 3.1, the first block in a FIS is fuzzification. This is the process of computing crisp values into fuzzy inputs in terms of fuzzy sets using one or various membership functions. It is the process of taking real data (such as delay) and converting it into fuzzy input value such as the value "low" (Terano, 1992; Jantzen, 1998). This is called a label, and the conversion process is performed by a membership function. The goal of fuzzification is to produce fuzzy inputs that can be processed by the second block which is the inference engine, with the rule evaluation step. The fuzzy inference system consists of if-then rules that specify a relationship between the input and the output of fuzzy sets. The third block implements the defuzzyfication stage, if necessary. It provides a crisp value from the rule aggregation result.

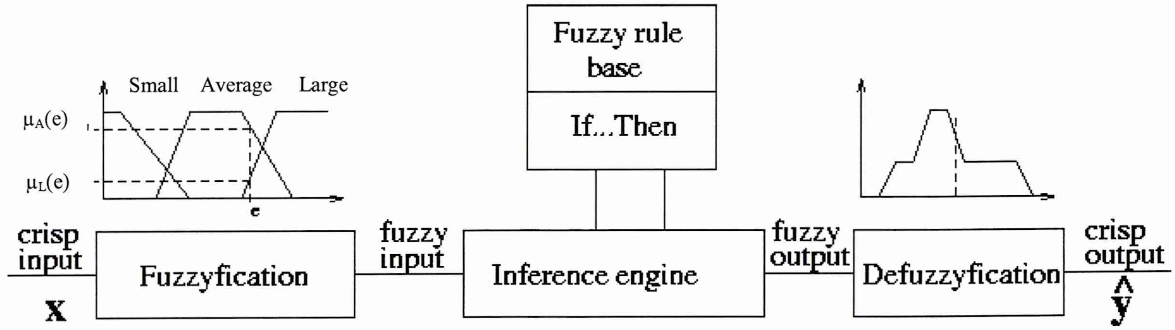


Figure 3.1: Fuzzy inference system block diagram

3.2.2 Rule Base

The rule base contains linguistic rules that are provided by experts. Once the rules have been established, the fuzzy inference system (FIS) can be viewed as a system that maps an input vector to an output vector. A fuzzy rule base contains a set of fuzzy rules. A single if-then rule generally assumes the general form:

IF (antecedent 1) AND (antecedent 2) THEN (consequent)

An example of a rule might be "IF *packet loss* is *high* and *delay* is *high* THEN *QoS* is *poor*". The linguistic rules describing the control system consist of two parts: an antecedent block and a consequent block. The antecedent are the inputs that are used in the decision making process or the IF part of the rule (e.g. IF *packet loss* is *high*). The consequent are the implications of the rule or the THEN part (e.g. THEN *QoS* is *poor*) (Raju *et al.,* 1991). Fuzzy operators allow the combination of antecedents into premises. The fuzzy logic operators are AND, OR and NOT. Given two antecedents a and b, the operators are defined as given in Equation 3.1 (National, 1997):

$$\begin{bmatrix} \text{AND: } \mu A \bullet B = \min(\mu A, \mu B) \\ \text{OR: } \mu A + B = \max(\mu A, \mu B) \end{bmatrix} \quad \textbf{3.1}$$
$$\text{NOT: } \mu\neg A = 1 - \mu A$$

The rules in a fuzzy system are represented in linguistic variables as a way to capture available semi-qualitative knowledge. The definition of rules relies on the designer's experience and knowledge of how the system should behave (Pitsillides and Sekercioglu, 1994). Depending on the system, it may not be necessary to evaluate every possible input combination, since some may rarely or never occur (Hellmann, 2001).

3.2.3 Membership functions

The meaning of linguistic variables is defined by their membership functions. A membership function $\mu(x)$ is a general representation of the magnitude of the participation of each input. It associates a weighting with each inputs that are processed and defines functional overlaps between them. The rules use the input membership values as weighting factors to determine their influence on the final output conclusion. There are different memberships functions associated with each input and output response. Different shapes of membership functions can be employed, the most commonly used shapes for membership functions are triangular, trapezoidal, Bell-shaped and Gaussian (Mendel, 1995) Figure 3.2 illustrates a Gaussian membership function example of the output variable *QoS*.

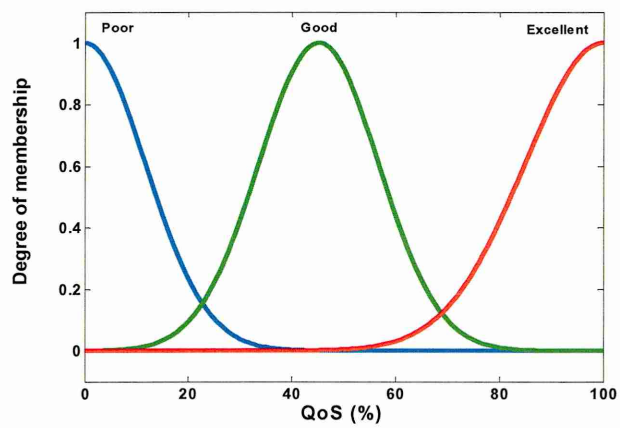


Figure 3.2: Gaussian membership functions for the output variable QoS

Every element in the universe is a member of a fuzzy set with a degree of membership between zero and one. The degree of membership for all its members describes a fuzzy set, such as poor label of the output variable QoS. In fuzzy sets, each element of the universe of discourse is assigned a degree of membership (Jantzen, 1998).

The membership function $\mu(x)$ allows gradual transitions from one fuzzy set to the other, with intermediate value presenting degree of membership to the fuzzy set. For example, the transition from the membership function *low* to the membership function *medium* occurs gradually according to the overlap between these functions. Fuzzy sets have a more flexible membership requirement that allow partial membership in a set. The degree to which an object is a member of a fuzzy set can be any value between 0 and 1, rather than strictly 0 or 1 as in traditional set.

Membership functions do not have to overlap; but, one of the great strengths of fuzzy logic is that membership functions can be made to overlap. This expresses the fact that "the glass can be partially full and partially empty at the same time". Another important factor to consider is the universe of discourse. It contains all elements belonging to an input or output and should be considered before setting up the membership functions. For example, in the following rule: "IF *packet loss* is *high* AND *delay* is *low* THEN *QoS* is *poor*", the membership functions for high and medium have to be defined for all possible values of throughput and delay, and a standard universe may be suitable.

As an example, Figure 3.3 illustrates two sets of membership functions for the set of terms: short men, medium men, tall men. Clearly, these terms would have a very different meaning for a professional basketball player than they would have for most other people. This illustrates the fact that memberships functions can be quite context dependent (Mendel, 1995).

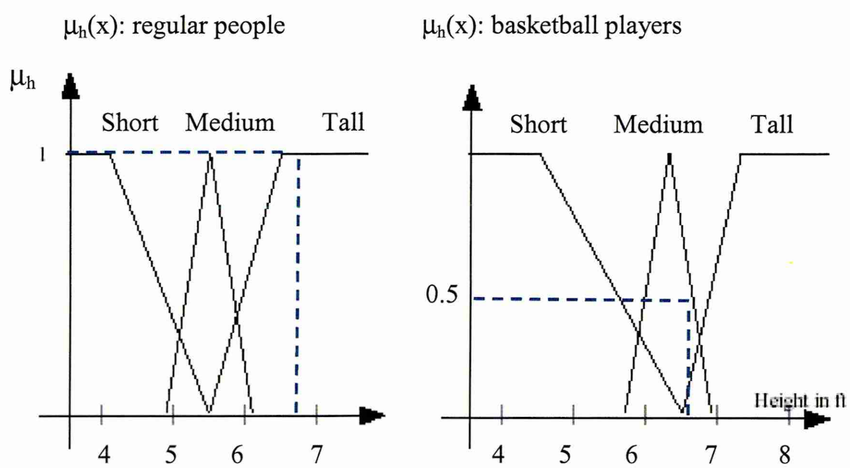


Figure 3.3: Membership functions for T(height) for two different observers

The terms short, medium, tall are employed to identify the different height categories. These terms are named linguistics terms and correspond to values of the linguistic variable height. Each term is represented by a fuzzy set defined by a membership function $\mu_h(x)$, the y-axis is the degree of membership and the x-axis is known as the universe of discourse (U) and is defined as the whole range of the fuzzy set input variable (Jantzen, 1998). The linguistic variable permits the translation of crisp measure value into its linguistic description. This process is called fuzzification.

The linguistic variable in Figure 3.3 allows the fuzzification of the crisp value of height, given in foot, into its linguistic description. For instance, for a basketball player, a height of 6 ft 5 might be evaluated as a medium height with a degree of membership equal to 0.5 whereas for most other people, this will be viewed as a tall height with a degree of membership equal to 1. The degree of membership to the fuzzy set medium height can be interpreted as degree of truth associated to the statement "measuring 6 ft 5 is medium for a basketball player".

3.3 Fuzzy inference engine

The FIS uses the fuzzified inputs together with the rules to perform inferencing. The fuzzy inference consists of two components: firstly, aggregation which evaluates the antecedents (IF part) of the rule and secondly composition which evaluates the consequent part (THEN part) of the rule. In this step, there are now rules to be processed and the end result is a set of rule strength. When a real-world system inputs is fuzzified by its corresponding membership function, it will produce multiple fuzzy inputs each with a corresponding numeric value (Aboelela, 1998). The operators AND and OR can be used to shape the membership function of the output fuzzy sets according to Equation 3.1. Hence, the process of rule evaluation produces a rule strength which is defined as the minimum (or smaller) numeric value of both antecedent and finally the rule strength is fed into the optional defuzzification step.

The type of fuzzy control method used in this study has been proposed by Mamdani and it is the most widely used approach in fuzzy control (Lee, 1990_a; Lee, 1990_b). The main feature of this type of FIS is that both the antecedent and consequent of the rule are expressed as linguistics constraints (Zadeh, 1965). As a consequence, Mamdani FIS can provide a highly intuitive knowledge base that is easy to understand and maintain.

3.4 Defuzzification

This process is characterised by the combination of the various rule strength to produce an output. It is optionally used when it is useful to convert the fuzzy output set to a crisp number. The goal of defuzzification step is to produce a real-world system output. Given a fuzzy set that encompasses a range of output values, the defuzzifier returns one number, thereby moving from a fuzzy set to a crisp number. Several methods for defuzzification are used in practice, including the centroid, maximum, mean of maxima and height (Fuzzy logic fundamentals, 2001). The most popular defuzzification method is the centroid expressed in Equation 3.2, which calculates and returns the centre of gravity of the aggregated fuzzy set (Ross, 2004).

$$Y = \frac{\sum_{i=1}^{m} y_i \times \mu_i}{\sum_{i=1}^{m} \mu_i} \quad \mathbf{3.2}$$

where,

m = number of output fuzzy sets obtained after implication,

y_i = centroid of fuzzy region i (i.e., the output universe of discourse)

μ_i = output membership value.

Simulation Experiments Methodology

4.1 Introduction

The aim of this chapter is to provide the details of procedures carried-out in most of the experiments in this study. The procedures that are unique to a particular study are explained in their relevant chapters. This chapter also presents the network simulation environment used to carry out network performance analysis in relation to various routing scenarios such as pause time or traffic load. A detailed description of the methodology, simulations tools and data processing methods used in this research is discussed. A justification of the network simulation environments, experimental design and the measurement procedure is provided.

4.2 Simulation overview

In this study due to the nature of wireless networks and the practical issues involved in implementing real systems, simulation experiments have been carried out under various scenarios. Generally, this approach is considered as an appropriate method for performance evaluation as opposed to analytical modelling or measurement (Jain, 1991). A typical simulation procedure involved network measurements and quantitative post-processing analysis. The investigation and critical evaluation of the data were based on two criteria: (i) the results collected from the routing experiments and (ii) the findings from recent studies related to this study.

In order to simulate wireless networks with realistic topologies a simulation tool was required. In a recent study by Kurkowski *et al.* (2005), a survey showed that 44.4% of current MANET simulation studies used NS-2 (Network Simulator 2) (NS, 2006) and (Fall and Varadhan, 2006), 11.1% used GloMoSim (Global Mobile Information Systems Information Library) (GloMoSim, 2006; Zeng et *al.*, 1998), and 6.3% used OPNET (Optimised Network Engineering Tools) (OPNET, 2006). These are the most popular network simulation tools used throughout this research field. NS-2 is a discrete event simulator targeted at networking research (NS, 2006). It is a widely used simulation tool for simulating inter-network topologies to test and evaluate various networking protocols. NS-2 is open source and freely available. It runs on Linux operating system or on Windows with the installation of a Linux platform. It provides a comprehensive platform and can support a large number of network components such as different applications, protocols, and traffic models. It can be extended either by modifying the OTcl or C++ code. As stated, NS-2 is written in the object oriented language, *C++*, with an Object Tool Command Language (OTcl) interpreter. The simulator supports a class hierarchy in *C++*, and a similar class hierarchy within the *OTcl* interpreter. These two hierarchies are closely related to each other; from the user's view, via a one-to-one correspondence between a class in the interpreted hierarchy and one in the compiled hierarchy.

A simulation task is specified in NS-2 by a simulation script that specifies the network topology (node configurations, locations, and interconnection), the transport protocol used (e.g., UDP), the source application (e.g., CBR), and the events (send data, movement, etc...). A typical NS-2 TCL script includes the following:

(i) The node configurations, e.g. medium type, interface queue, routing protocol.

(ii) Traffic details (i.e. transport protocol, application, and communication pairs) and movement scenarios (i.e. node location).

(iii) Schedule events, such as start and end times of data packet transmission and when the simulation should terminate.

(iv) Simulation activation command.

The simulation results required post-processing to extract the desired information. The process used is illustrated in Figure 4.1. It consisted of generating the simulation script i.e. a TCL script that contained the network components, the traffic and mobility model. The simulation generated two trace files, the event file and the visualisation file. The event file, most commonly known as the trace file, contained a list of the different events that occurred during the simulation (i.e. the event, the time and location it occurred, etc...). The visualisation file (*NAM*) showed a graphical representation of the network topology as well as the node movement, packets sent, packets received and packets drops.

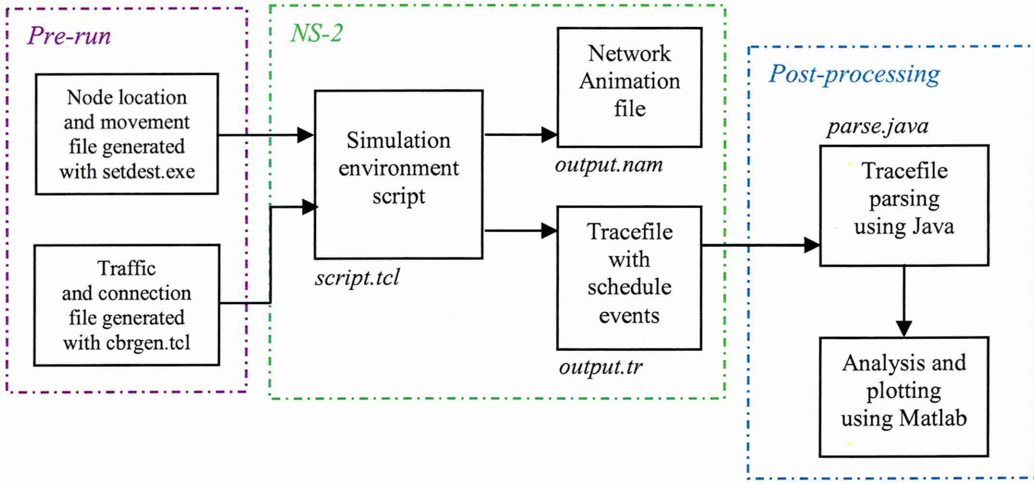


Figure 4.1: Simulation process for NS-2

The tracefile is a mixture of text and figures therefore Java was used to parse the required information to calculate the performance metrics, as it is a powerful tool for text parsing. Other tools such as Perl (Wall and Schwatz, 1993) and *MATLAB* (MATLAB, 2004) were also used for data processing, analysis, plotting and evaluation.

In order to make accurate and effective comparisons between routing protocols, it is critical to set up the simulations environment with identical loads and traffic conditions. Each simulation accepts as input: a movement and traffic file that describes:

- the direction and speed of movement of each node
- the sequence of packet originated by each node
- the time at which changes in movement or packet sent occurred

4.2.1 Network Topology

The wireless simulation model was composed of 50 nodes forming a mobile ad-hoc network. Throughout this study, a node can be a source, a destination or a forwarding agent. These nodes moved about over a rectangular x-y plane (flat space) of dimension 1000x500m, during 900 seconds. A rectangular space was used in order to have longer routes (multi-hop) between nodes in comparison with square space. The number of nodes in the scenarios was kept at 50. The reason to fix this value was that 50 is used in most ad-hoc networks studies (Kurkowski *et al.*, 2005). More nodes would have resulted in more increased processing time.

Mobility scenarios files were pre-generated with varying movement patterns based on random motion movement with various velocities. These pre-generated files were obtained by executing a pre-compiled program based on the Random Waypoint algorithm (RWP) (Bettstetter *et al.* 2003; Broch *et al.* 1998). The RWP model has been used in this study because its pattern is close to the behaviour of a mobile user used in an ad-hoc network, it is implemented and widely distributed with NS-2 and various studies on performance evaluation are based on this model (Broch *et al.,* 1996; Perkins *et al.,* 2000; Johansson *et al.,* 1999; Das *et al.* 2000_a). In this model, each node is initially stationary for pause time interval. Then, every node in the network selects a random destination point "waypoint" in the 1000x500m space and moves to that destination at a speed distributed uniformly between $[V_{\min}, V_{\max}]$. Upon reaching the destination the node pauses again for a predefined time interval, selects another destination, and proceeds as previously described, repeating this behaviour for the duration of the simulation (Bettstetter et al, 2003).

Recent work by Yoon *et al.* (2003_a and 2003_b) indicates that RWP suffers from speed decay, as the simulation evolves, the average node speed decreases with time before reaching a steady state, in particular if the minimum speed is zero. The authors also outline how this affects ad-hoc routing protocols such as Dynamic Source Routing (DSR) (Johnson and Maltz, 1996) and Ad-hoc On-demand Distance Vector (AODV) (Perkins and Royer, 1999). Such speed decay can have a dramatic influence on measured performance and overhead. Consequently, time averaged metrics cannot be presented during this period of decay as the underlying process in not stationary (Bettstetter and Krauser, 2001; Bettstetter *et al.,* 2003; Chu and Nikolaidis, 2002). Several methods have been suggested to overcome the problem (Camp *et al.,* 2000). Amongst these, Yoon *et al.* (2003_a) recommended to use a positive minimum speed, warm-up every simulation by running it until steady state is reached and then deleting the initial data, before post processing. This improved method is able to quickly converge to a constant speed. This analysis, although based on simplifying assumptions, was used to accurately estimate the expected instantaneous average node speed given the minimum and maximum speeds. Through mathematical analysis, Yoon *et al.* (2003_a) demonstrated that for a uniformly distributed speed between 1 and 19 m/s, the settling time will be achieved after 142s. Therefore, for our study the warming up period was set to 150s, which was considered more than the speed decay. For a uniformly distributed speed between 1 and 19 m/s, $\overline{V}$ is the steady state average speed calculated according to Equation 4.1 (Yoon *et al.* 2003_a).

$$\overline{V} = \frac{E[D]}{E[T]} = \frac{V_{max} - V_{min}}{\log_e(\frac{V_{max}}{V_{min}})} \quad \textbf{4.1}$$

Where *E[D]* is the expected travel distance, *E[T]* is the expected travel time and *V* is the node speed.

Consequently, the RWP model used for this study was set with the following values: maximum speed $V_{max} = 19$ m/s, as most studies used a maximum speed of 20 m/s, and minimum speed $V_{min} = 1$ m/s, resulting in an average speed of 6.11 m/s according to Equation 4.1. The pause time was used as the mobility parameters to determine how it affected the dynamic topology. Figure 4.2 was obtained by averaging the number of link changes for each simulation run of this study and it shows that the number of link changes is inversely proportional to the pause time. A link change basically means that a link changes its states from either valid or invalid (i.e. broken). It shows that as the pause time increases the network will be expected to be more stable hence, control packet generated by the routing protocols should be kept to a minimum. Movement files were generated for 7 different pause times (0, 30, 60, 120, 300, 600 and 900 s). A pause time of 0s corresponds to continuous movement and a pause of 900s (length of the simulation) corresponds to stationary nodes. Because the performance of the protocol is very sensitive to the movement pattern, the scenarios were repeated 9 times for each pause time, resulting in a total of 63 different movement files. AODV and DSR protocols have been analysed with the same 63 movement patterns files.

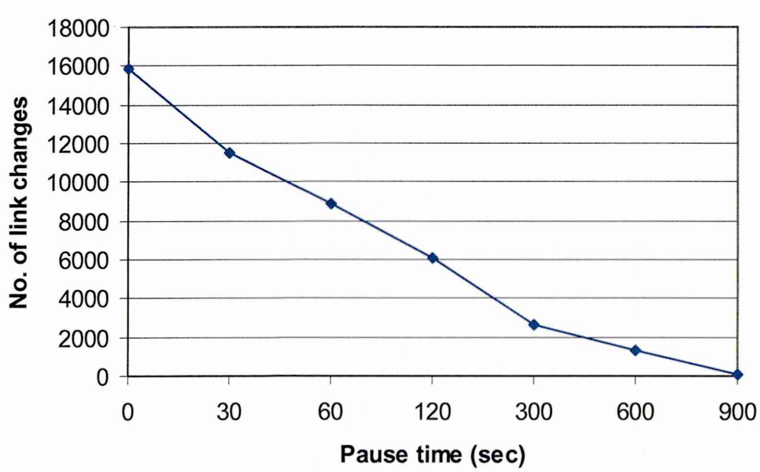


Figure 4.2: Relationship between number of link change and pause time

4.2.2 Physical Layer Model (PHY)

The physical layer model of each mobile node's network interface was chosen to approximate the early generation commercial wireless adapters. Hence for this study, the channel data rate was set to 2Mbps for data packets transmission and the basic rate was set to 1Mbps for control packets transmission. The radio transmission range was 250m for the given model.

4.2.3 IEEE 802.11 MAC Layer Implementation

The wireless model in NS-2 was implemented with the IEEE 802.11 MAC Protocol Distributed Coordination Function (DCF) mechanism (Van Nee, 1999; Crow *et al.*, 1997; IEEE, 1997). DCF is the basic access mechanism used by mobile devices to share the wireless channel and all simulations were carried out based on this model. It provided scalability, simplicity and availability (Ziao and Pan, 2005). The IEEE 802.11 standard was modelled to approximate the Lucent WaveLAN^{TM} DSSS (Tuch, 1993) at a frequency of 914 MHz and a *DSSS* radio interface card.

4.2.4 Interface Queue (IFQ)

This specifies the number of packets that can be stored in the interface queue at each node. If this queue fills up, packets will be dropped. Each node has a FIFO (first-in first-out) priority queue for packets awaiting transmission. The network interface can hold up to 50 packets and is managed in a DropTail fashion. During packet bursts problems occur, such as queue overflows, packets that cannot be admitted will be dropped as they are entering. This explains why all queuing discipline based on FIFO are known as DropTail queues (Parris, 2001). The queue mechanism gives priority to routing packets by inserting them at the head of the queue. Each on-demand routing protocol (i.e. DSR, AODV) can maintain an additional buffer of 64 packets, which contains the data packets waiting to be sent. A schematic of a mobile node is illustrated in Figure 4.3, as it is presented in NS-2 (NS-2, 2006), the variables represents the C++ classes names.

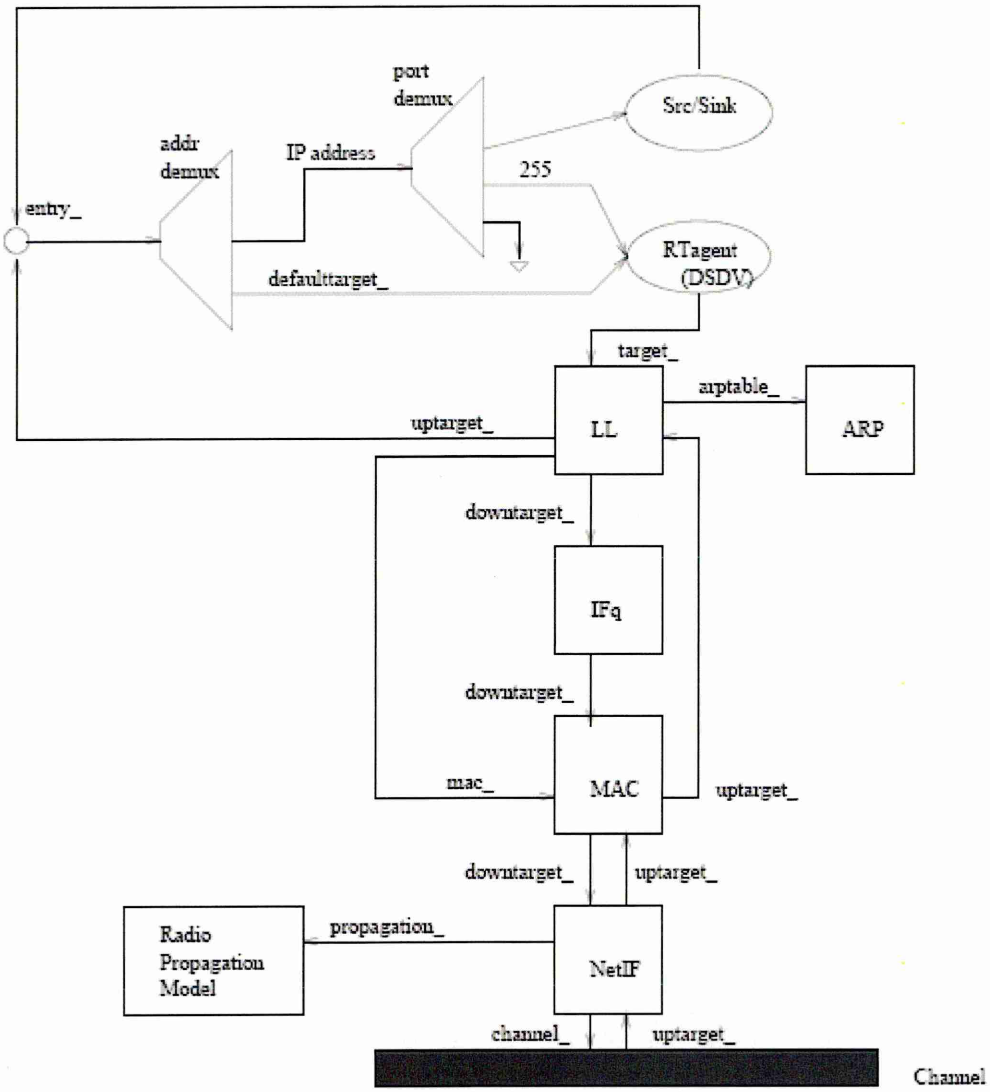


Figure 4.3: Schematic of a mobile node in NS-2 (Fall and Varadhan, 2006)

4.2.5 Network Layer Routing Protocols Default Setup

The simulations are executed according to the routing protocols specific parameters given in Tables 4.1 and 4.2. These values conform with previous related studies (Celebi, 1998 and Boursier *et al.,* 1998) and NS-2 simulator default settings.

Table 4.1: Constants used in AODV simulations

Parameters	Values
Hello interval	1 s
Active route timeout	30 s
Reverse route life	5 s
Route reply lifetime	1 s
Allowed hello loss	3
Request retries	3
Time between retransmitted requests	3 s
Time To Leave (TTL)	8 s
Maximum rate for sending replies for a route	1 / s

Ad-hoc On-demand Distance Vector (AODV) setup defaults are shown in Table 4.1. AODV uses IP level *hello* messaging to detect link breakages. If a *hello* message is not received within 1 second, referred to as hello interval, the link is assumed to be broken. An active route times out after 50 seconds (active route timeout) and a new route discovery is initiated. The reverse route life, taken by route reply packets, is set to 5 seconds (reverse route life). The route reply message should be received within 1 second after the request. If any node does not answer with a *hello* message three times (allowed hello loss), it is assumed that the link is broken. Route discovery is only initiated three times (request retries). Request retransmissions are done with three seconds intervals. Packets are held for 8 seconds while they await a route to be discovered and a node can send one route reply every second.

Table 4.2: Constants used in DSR simulations

Parameters	Values
Time between retransmitted requests	500 ms
Size of source route header carrying an address	$4n + 4$ bytes
Timeout for non-propagating search	30 ms
Maximum rate for sending replies for a route	1/s

Table 4.2 presents the defaults values set up for DSR routing protocol. To find a new route, DSR protocol waits for 500 ms before broadcasting route request packets. This time period is defined as the time between retransmitted requests. In DSR, routing packets are added to data packets during transmissions. The source routing packets' header length is a function of the estimated route length n, denoted by the number of hops. It is defined as $4n+4$ in bytes (Celebi, 1998). After 30 seconds, if a route finding process is still in progress it is interrupted. To avoid control message overhead, no more than one reply can be sent to the requesting hosts per second.

4.2.6 Traffic type

In section 2.5.3, a description of the characteristics and general requirements of voice, video and data applications is given. Thus, for this study, two types of multimedia traffic were considered: IP telephony and video on-demand. Both were generated using the Constant Bit Rate traffic generator included in NS-2. Tobagi *et al.* (2001) stated that voice connections generate a stream of small packets at relatively low bit rate ranging from 5Kbps to 64Kbps, depending on the encoding scheme (e.g. G.711, WMA, etc.). For this study, the audio application data rate was set to 64Kbps with a packet size equal to 128 bytes. Video traffic spans a wide range of data rates, from tens of Kbps to tens of Mbps. The data rate greatly varies according to the content, the video compression scheme and the video encoding scheme (H.261, MPEG, etc.). For example, the application of H.261 includes video on-demand; the data rates are multiples of 64Kbps up to 2 Mbps and the coding is designed to achieve a fairly uniform bit rate across frames (Tobagi *et al.,* 2001). Moreover, H.261 UDP packets, range anywhere in size from 48 bytes to 1024 bytes (Colin, 2002). Consequently, for this study the video traffic data rate was set to 192 Kbps with packet size equal to 1024 bytes.

The transport layer in the OSI (Open Systems Interconnection) reference model provides a connectionless transport protocol known as User Datagram Protocol (UDP) (Halsall, 1992). UDP provides a best-effort (connectionless) service for the transfer of individually-addressed message units known as datagram. This mode of operation minimises the overhead associated with each message transfer since no network connection is established prior to sending a message (datagram). UDP does not perform any error control, no acknowledgement is required thus reducing overhead due to retransmission. Applications which are time-sensitive would benefit from a transport protocol like UDP (Zheng and Boyce, 2001), as opposed to applications which are much more sensitive to loss of data such as data transfer applications. Applications such as data transfer would rely on the transport protocol to attempt to retransmit the data in cases of congestion. The Transmission Control Protocol (TCP) provides a user application process with a reliable service.

The data packet size, sending rate and the number of connection together make up the total load of the network. These were:

- For voice traffic the sending rate was fixed to 64Kbps. The number of connections varied between 1, 3, 5, 7 and 10 resulting in a total maximum constant bit rate (CBR) transfer rate of 64Kbps to 0.625 Mbps.
- For video traffic the sending rate was fixed to 192Kbps. The number of connections also varied between 1, 3, 5, 7 and 10, resulting in a total maximum constant bit rate (CBR) transfer rate of 192Kbps to 1.875 Mbps.

All communication patterns were peer to peer and transmissions started at times uniformly distributed between 150 and 200 seconds to allow the system to settle into a steady state after the speed decay problem highlighted detail in section 3.2.1.

Five different scenarios with a different number of connection (1, 3, 7, 5, and 10), taken in conjunction with the 63 movement profiles provided a total of 315 different scenarios files with which the performance of networks was compared. Each scenario was stored in a separate file for further processing.

4.2.7 Assumptions

Once the network simulation environment including the topologies and the other simulation parameters were determined, some assumptions were required. The channel was considered as error free and collision was the only cause of transmission failure over the channel. The capacity of the channel was set to 2 Mbps. This helped to minimise the simulation time into a manageable duration and limited the size of the output simulation file. The propagation times were assumed to be negligible with respect to the packet transmission time. Each station transmitted one type of traffic to its corresponding destination. All simulations were performed in *WLAN* environments with a different number of connections.

4.3 Experiment design

4.3.1 Simulation variables

Table 4.3 shows a summary of the simulation parameters used in this study. Description of chosen set up variables was previously argued in the above sections.

Table 4.3: Simulation parameters

Parameter	Value
MAC protocol	IEEE 802.11b
Propagation model	Two Ray Ground
Antenna	Omni-directional Antenna
Channel bandwidth	2.0 Mbps
Basic rate (Preamble)	1.0 Mbps
Transmission range (m)	250
Frequency band	914 MHz
Interface queue length	50 packets
Simulation time (s)	900
Number of nodes	50
Pause time (s)	0, 30, 60, 120, 300, 600, 900
Mobility speed (m/s)	Max 19 (Avg. 6.11)
Traffic type	Constant Bit Rate
Packet rate (packets/s)	6, 24
Packet size (byte)	128, 1024
Maximum number of flows	1, 3, 5, 7, 10
Scenario size	1500x500m
No. of iteration for each pause time	9

4.4 QoS Parameters and Routing Performance Metrics

Packet delivery fraction, delay, jitter and loss were considered the main QoS parameters used for this study. These parameters are described in section 2.5.4 (c.f. Chapter 2). According to the application type, delay, jitter and packet loss were considered the main QoS parameters for time-sensitive applications such as audio and video. However, normalised routing load and hop count were used as the evaluation metrics for routing protocols performance. Recommendations for delay, jitter and loss from ITU G.1010 were considered in this study as show in Table 2.3. RFC (Request For Comments) 2501 (Corson and Macker, 1999) outlines some metrics for evaluation of routing protocols. Amongst other many studies (Celebi, 2001; Das 2000_a and 2000_b; Jiang and Garcia-Luna-Aceves, 2001; Johnson and Maltz, 1996) have used a normalised routing load and a hop count, hence the choice of these metrics for this study which are further explained in the following sections.

4.4.1 Normalised routing overhead ratio

The normalised routing overhead ratio was calculated by dividing the total number of routing packets sent by data packets sent over the simulation interval. For packets sent over multiple hops, transmission of the packets over a hop counted as one transmission. This quantity points towards the efficiency of the protocol in limiting the control overhead during data transmission. Normalised routing overhead ratio is a very essential measurement for evaluating routing protocols, as it determines how a protocol behaves in congested or low-bandwidth environments. Protocols generating and forwarding large amounts of routing overhead amplify the probability of packet collisions. And data packets may remain in the network interface queues for longer period of time, hence increasing delay (Perkins *et al.,* 2001; Corson and Macker, 1999). Only routing packets were used and the performance were evaluated and quantified. Therefore, we did not consider other control packets such as IEEE 802.11 MAC packets or ARP (Address Resolution Protocol) packets. The normalised routing load ratio metric was computed according to Equation 4.2.

$$Normalised\ routing\ load\ ratio = \frac{r_p}{\sum_{i=i}^{S}(S_i + f_i)}, \qquad \textbf{4.2}$$

Where,

r_p = total number of routing packets generated by source and forwarded at intermediate nodes

S = total number of generated data packets

f = total number of data packet forwarded at intermediate nodes

i = data packet ID

4.4.2 Average hop count

This metric represents the average number of hops travelled by data packets that reached destinations. It was computed using Equation 4.3

$$Average\ hop\ count = \sum_{i=1}^{S}\left[\frac{S_i + f_i}{S_i}\right] \qquad \textbf{4.3}$$

Where,

S = total number of generated data packets

f = total number of data packet forwarded at intermediate nodes

i = data packet ID

A low hop count can indicate effectiveness of route selection. This statement is true when different routing protocols have the same packet delivery fraction. However, if routing protocols have different packet delivery fraction (especially in networks with high mobility rates and link changes), hop count is closely related to the packet delivery fraction (Lee, 2000). The higher the delivery rate, the higher the hop count. Since only data packets that survive all the way to their destinations are reflected, a low hop count means that of the data packets delivered are destined for nearby nodes, and packets sent to remote hosts are likely dropped. Thus, the hop count measure provides information and survivability of the protocols.

Note, however, that all these metrics are not completely independent. For example, lower packet delivery fraction means that the delay metric is evaluated with less number of samples. Typically, the longer the path length, the higher the probabilities of a packet drop. Thus, with lower delivery ratio, samples are usually based in favour of smaller path lengths and thus less delay. Also, low routing load impacts both delivery ratio and delay as it causes less network congestion and multiple access interference.

4.5 Related Work

The QoS performance of mobile wireless ad-hoc networks with two popular on-demand routing protocols, AODV (Perkins, Royer and Das, 2000 and Perkins and Royer, 1999) and DSR, (Johnson and Maltz, 1996 and Johnson *et al.*, 2001), were studied. AODV and DSR were chosen as comparison points, as these two protocols are more popular (Broch *et al.*, 1996; Johansson *et al.*, 1999; Maltz *et al.*, 1999_b and Das, Perkins and Royer, 2000). Recent IETF drafts also exist that clearly describe their specifications (Perkins, Royer and Das, 2000 and Johnson *et al.*, 2001). Also, simulation codes for these protocols are available in the public domain by the authors of these protocols making the comparisons easy and fair. Both DSR and AODV are source initiated and flood-based protocols, where routes are established as a response to a flooded query initiated by the source. The major difference between them is that DSR uses source routing and aggressively caches source routes, possibly multiple routes, to reuse later; AODV, on the other hand, is more conservative, and uses a sequence number-based scheme to keep only the most recently learnt route.

Lu *et al.* (2004) investigated packet loss in MANET using NS-2, mobile hosts were randomly placed using random waypoint model, traffic was modelled using UDP and TCP and their work outlined the impact of traffic load and congestion control scheme on packet loss. AODV and DSDV, which is a table driven protocol, were used. They outlined the impact of mobility on AODV packet loss, which represented more than half of the transmitted data as load increased. Pucha *et al.* (2004) investigated the impact of traffic patterns on AODV and DSR based protocols. They increased the traffic load by keeping the number of connections per source and the number of traffic sources fixed but increased the packet rate per connection. Their study showed that with increased packet rate per connection, both routing protocols have incurred an increase in routing load which implies a reduction in packet delivery. Their results also showed that low interval time between data packet on the same connections affects protocols with static and adaptive timeout in addition to reducing caching efficiency. This affects the time a packet is buffered before it is sent. If the interval time is too low then the active route can be timed out even before the packet is sent.

In Campos *et al.* (2005), an evaluation of three routing protocols is reported namely, AODV, DSR and TORA for a videoconferencing application. This used CBR at a rate of 28.8kbps. Their study showed that AODV and DSR always have high packet delivery fraction in all mobility levels (i.e. low, moderate and high speed), however only AODV can achieve the required QoS for the application with a packet delivery higher than 95%. However, in their study they did not consider jitter measurement which is another important metric for videoconferencing along with end-to-end delay and loss. Taking into account the fact that for a video application end-to-end delay needs to be less than 150 ms (ITU, 1998; Baldi and Ofek, 2000), their results indicated that AODV and DSR seem to be suitable for such a class of application in low load, even in high mobility. Despite such an outcome, it was stressed that DSR is not adequate in high traffic load and high user mobility.

4.6 Summary

This chapter described the procedure followed to investigate the QoS performance of the application using AODV and DSR on-demand routing protocols. It outlined the methods used to validate the performance of the proposed approaches and to compare their performance with the conventional on-demand mechanisms. Most of this study is based on simulations carried out using *NS-2*; and so its features were discussed in section 4.2. The data collected by the simulations are quantitatively analysed to validate the performance of the proposed approaches and compare their performance with the conventional mechanisms. The measurement process was supported by Java scripting language and MATLAB for plotting results since the latter contained the toolbox for implementing fuzzy logic system. And finally, a short survey is included outlining similar studies and their limitations.

Impact of Mobility on Routing Performance

5.1 Introduction

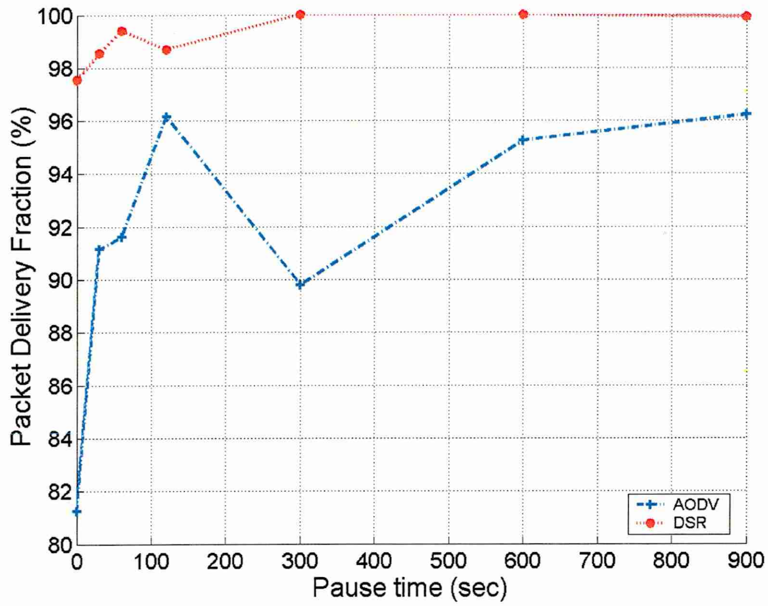
In this chapter, the effect of increasing the mobility factor and pause time on the network performance is explored. To carry out this set of experiments, the pause time was varied from 0 to 900s, as discussed in Chapter 4. This leads to a set of results that vary from a network of constantly mobile nodes to a network of stationary nodes. This experiment was based on a single connection sending CBR over UDP modelling audio IP telephony at 64Kbps and video on-demand at 192Kbps. Each simulation time was set at 900 seconds and results were represented as an average of all packets. A detailed QoS routing analysis has been carried out using AODV and DSR. The QoS metrics are graphically represented under those various conditions and are included in Section 5.2.

5.2 Results and Discussion

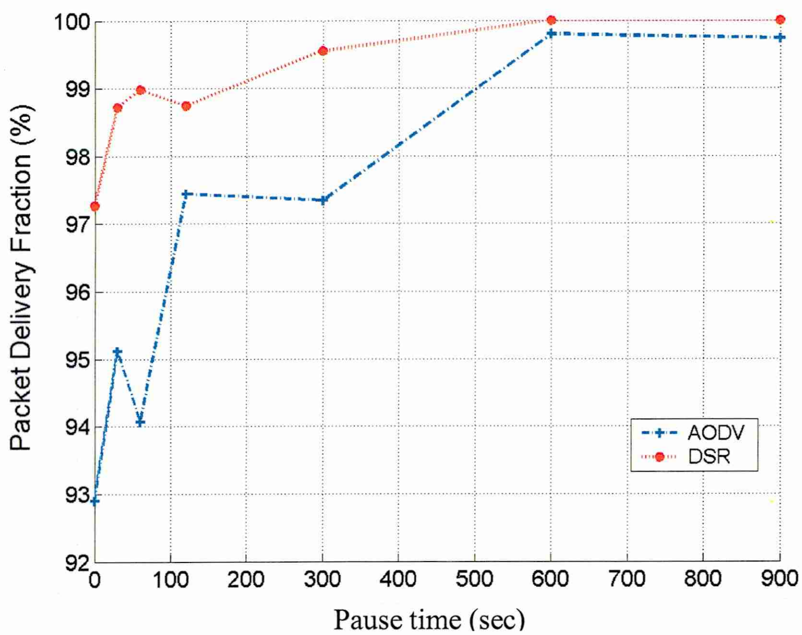
In this section, the effects of increasing the mobility of the mobile agents, i.e. nodes, on the network performance are discussed.

To simulate mobility, the number of connection pairs in the network was varied using the network load given in Table 4.3. The number of communicating hosts was increased from 1 to 10 connections, creating more active routes and more hosts involved in the routing process, therefore congestion-related and mobility-related packet losses are expected. The results were extracted for each flow (i.e. connection pairs) and presented as an average for the whole network. This experiment was based on 1-10 source(s) continuously sending data (CBR over UDP) at a rate of 64Kbps for audio applications and 192Kbps for video applications to 1-10 destination(s). The nodes were moving at an average speed of 6.11m/s, the pause time was increased from 0s (high mobility) to 900s (no mobility) for each set-up and the simulation time was set to 900s. Other simulation settings are as listed in Table 4.3 (see Chapter 4).

5.2.1 Packet delivery fraction



(a)



(b)

Figure 5.1: Percentage packet delivery fraction as a function of node pause time for (a) audio traffic and (b) video traffic

Figures 5.1a and 5.1b show the percentages of successfully delivering packets at different pause times. The graphs show the results for both AODV and DSR routing protocols for audio and video applications, respectively.

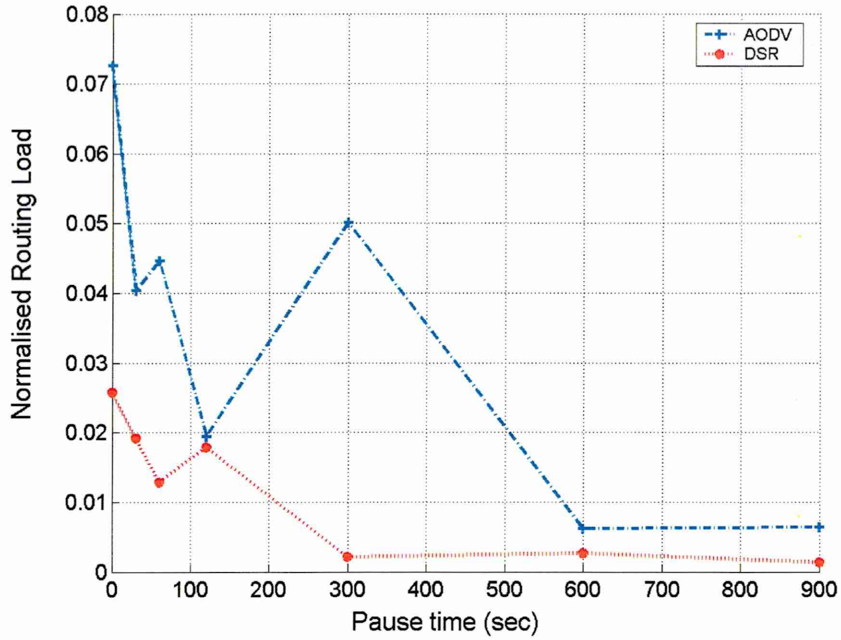
Figure 5.1a shows that DSR achieved a better quality of service than AODV for the transmission of audio application with an overall packet delivery fraction (pdf) above 98% (i.e. 2% packet loss). DSR uses its route cache to route the traffic in case of congestion thus reducing the packet waiting time before retransmission. At 900s pause time, DSR delivered all generated packets thus achieving 100% pdf. This is expected since a static network does not pose major challenge to the route maintenance mechanism.

On the other hand, at zero pause time (i.e. highest mobility) AODV exhibited a pdf of 88.4% (i.e. 11.5% packet loss) which is above the minimum QoS requirement of 3% presented in Table 2.3 for audio applications. This is due to the loss of periodic hello messages, which allow the node to know about its neighbours. Since the hello messages are sent periodically, in the case of high link breaks, the protocol gets the message too late and cannot update its knowledge of the current topology. The AODV has a default value, defined as hello interval, which sets a time at which the 'hello' message should be received and it is set to one per second according to Table 4.1. There is also a default maximum number of hello message loss (c.f. Table 4.1) to detect link failures. Since the latter is low and there are high drops in the network, protocol fails to recognise link's continuing reliability. If the 'hello' interval is decreased, link breakages would be detected earlier, but the control overhead would increase much more. As pause time was increased, the AODV pdf increased up to 95.7%. This gave a better QoS service, however it was still below the acceptable value of 3% packet loss (Table 2.3). At pause time 300s, the AODV pdf decreased to 87.4% which could be the result of high link connectivity changes. Packets were buffered in the queue and after the maximum retransmission attempt was reached, the packet were dropped, hence the reduction of packet delivery. For a pause time of 900s (i.e. stationary nodes), the AODV pdf was not 100%, the reason being that packets were sent before the routing tables had time to be fully setup. This resulted in dropped packets, as data were routed through broken routes and buffered packets were dropped because of congestion and route timeouts.

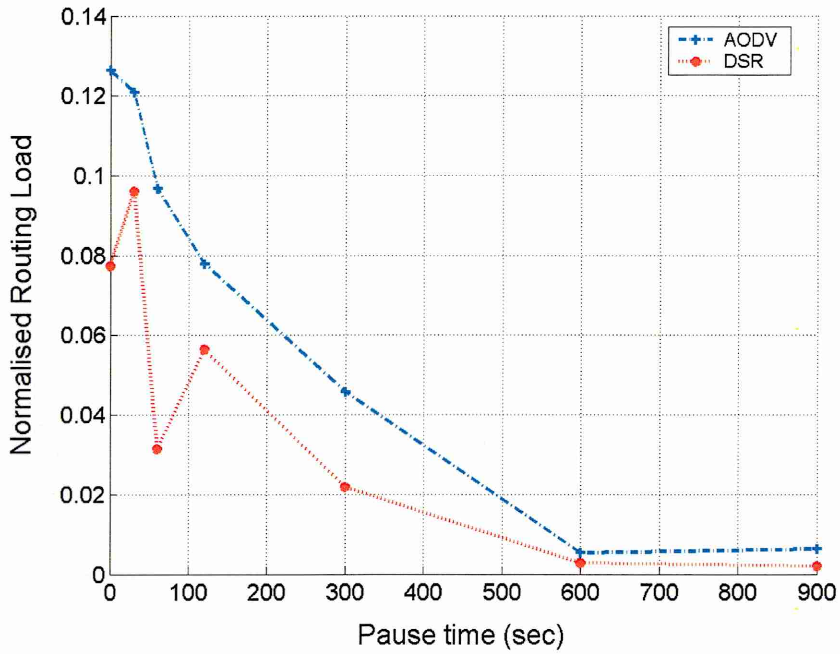
Figure 5.1b shows the results where the video-conferencing application was transmitted, DSR outperformed AODV by maintaining a high packet delivery percentage even during high mobility. At zero pause time, it managed to deliver 97.3% of the packets. As the pause time increased pdf rose up to 100 % providing a loss free path to the application for pause times higher than 300s. In this case DSR complied with the application requirement where acceptable loss should be less than 5% (i.e. pdf >95%), as seen on Table 4.4. At high mobility, the AODV achieved 93.5% packet delivery which again must be a lack of successful acknowledgement reply to detect link failures and route error messaging which informs source about the route invalidation. When the link breaks occurred frequently, these special control messages got lost and thus eliminated the advantage they might have had for the establishment of a new route. For pause times equal to 30s and 60s, values were equal to 95% and 94.3% respectively. Above these values, AODV maintained an acceptable QoS in terms of delivery with a pdf above 97% (i.e. 3% packet loss) but did not manage to provide a loss free path even for a static network (i.e. pause time of 900s).

5.2.2 Normalised routing load

Because a routing protocol needs to send control information to achieve the task of finding routes, it is important to investigate how much control information is sent for each protocol. Figures 5.2a and 5.2b show the normalised routing load (nrl) at different pause times, for the lowest load associated with audio and video applications, respectively. For the audio traffic in Figure 5.2a, the AODV routing protocol used more bandwidth to transmit packets as it has a greater control overhead, approximately 2.7 and 5.3 times more than DSR for pause time of 0 and 300s respectively. During topology changes, link breaks started to increase along with control messages, consequently the AODV has no information of alternative route, therefore it must re-initiate a route discovery resulting in bandwidth consuming retransmissions. DSR's routing load maintained a routing load of less than 0.03, which highlights the benefit of having an alternative route in the cache. When a route is unavailable, the source node has an alternative path ready to be used. At higher pause times, all members of the network learnt about the overall topology much better, therefore decreasing the attempts for route discovery. For video traffic in Figure 5.2b, the routing load for DSR still showed better performance than AODV's due to its caching mechanism. As the pause time was increased, AODV's routing load showed a linear decrease from 0.126 down to 0.046 until pause time reached 600s, while DSR's routing load exhibited unstable results with a peak value of 0.096.



(a)



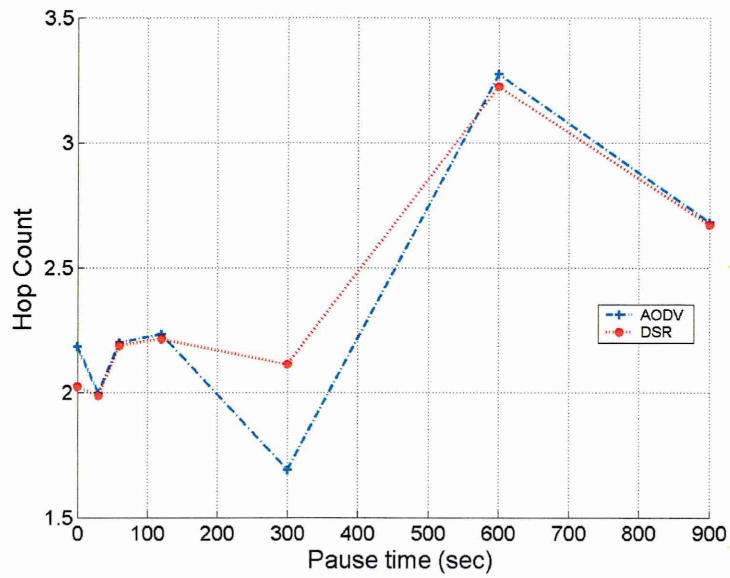
(b)

Figure 5.2: Normalised routing load as a function of node pause time for; (a) audio traffic and (b) video traffic

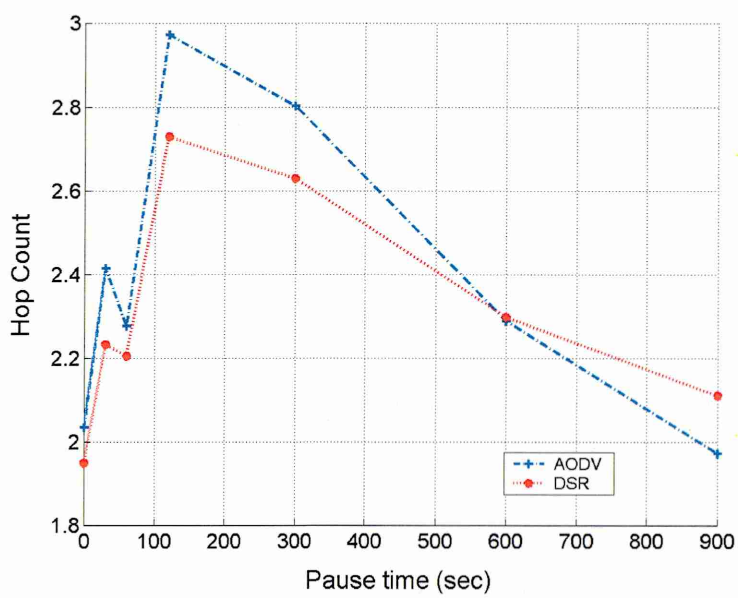
Referring to DSR pdf in Figure 5.1b, it still maintains the best QoS in terms of delivery. For pause time above 600s, the routing load was kept to a minimum under 0.01 for both protocols, when the network was close to being static, link changes occurred less frequently, thus reducing the need to forward routing messages.

5.2.3 Hop count

Figures 5.3a and 5.3b depict the average number of hops that successfully delivered packets that went through, as the pause time was increased. The hop count gives information on how the protocol performed. If the count is high, it indicates three situations: Firstly, the route used to transmit the data has multiple intermediate nodes. Secondly, the network is experiencing high congestion resulting in multiple packet retransmissions. Thirdly, as forwarded packets are also considered in the hop count Equation 4.3, it will affect the results. In all the situations, protocols QoS service will degrade therefore this parameter is worth considering when evaluating the network QoS.



(a)



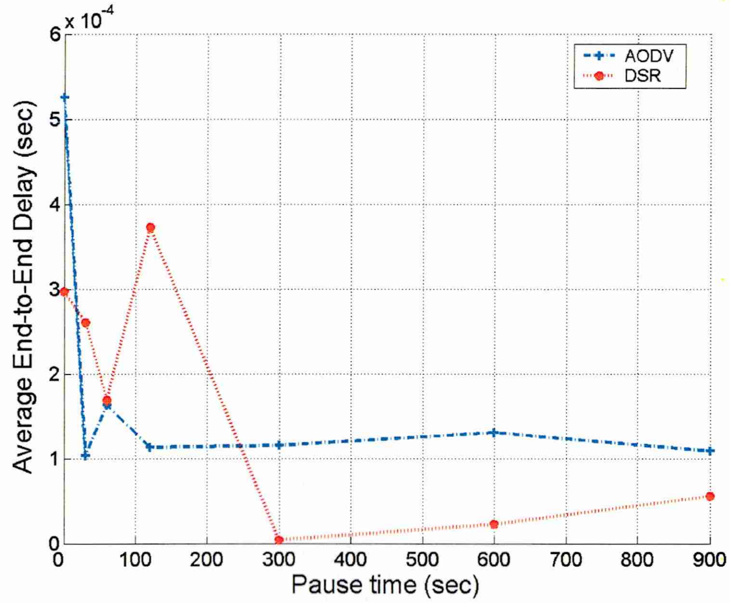
(b)

Figure 5.3: Hop count as a function of node pause time for; (a) audio traffic and (b) video traffic

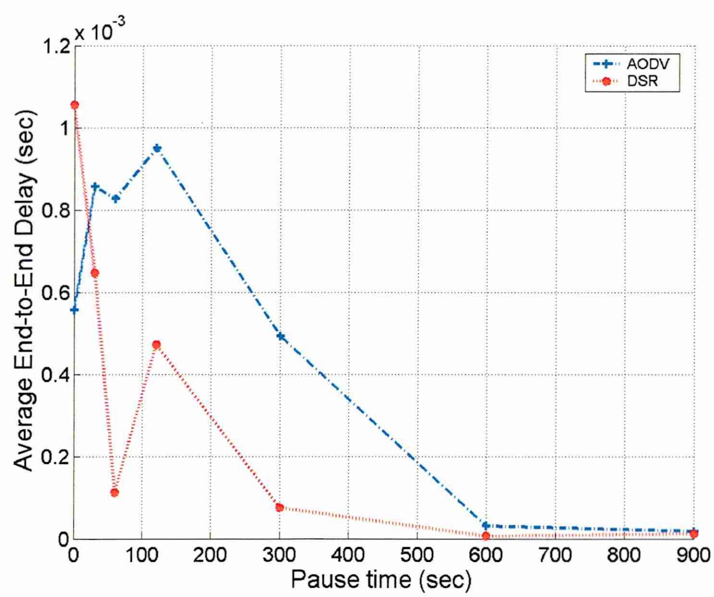
Figure 5.3a shows that AODV and DSR exhibits similar behaviour. At high mobility both protocols sent data through paths with approximately 2 hops (i.e. through 1 intermediate node). At 120s, AODV routing protocol exhibited a higher hop count than DSR due to the increase in forwarded packets. As the links break, packets were held in the queue and AODV attempted to find new shortest path to forward packets which can have an effect in hop count. Both protocols forwarded data along a longer route at pause time equal to 600s, hop count equal to 3.3 for AODV and 3.2 for DSR. When the nodes were static, the hop count came to 2.8 for AODV and 2.7 for DSR, this was a congestion related effect. When packets were held in buffers, the node would attempt to forward packets several times, which is a factor considered when defining hop count according to Equation 4.2. When the video on-demand traffic was transmitted the hop count was between 2 and 3 for both protocols (c.f. Figure 5.3b). The AODV constantly exhibited a higher hop count than DSR. At zero and 30s pause times, the hop count was less than 2.5 and at 60 and 120s it was between 2.5 and 3 respectively. This variation was mobility related since link breaks were high. Referring to Table 4.1, if the data packets were not sent within Time To Leave (TTL) (i.e. 8 second), packets were dropped, which reduced the number of packets to be forwarded and affected the hop count (c.f. Equation 4.3). As the nodes became less mobile the hop count decreased towards 2 hops for both protocols. However, there was a bigger difference at 600s as the route length for AODV was 2.66 whereas DSR was kept to a minimum at 1.96. The DSR route caching technique allowed for a fast route re-establishment. This limited the number of retransmitted packet during link breaks or congestion, thus reducing the value of hop.

5.2.4 End-to-end delay

For real-time audio and video applications, the end-to-end delay is an important QoS metric and following ITU G.114, the task group defines a preferable end-to-end delay of 150ms and a maximum of 400ms would be acceptable for both audio IP telephony and video-on-demand applications (c.f. Table 4.4). There are several factors that might lead to increased delay. The route discovery causes the routing protocol to make incoming packets wait in the routing agent queue during route discovery. The ability of the routing protocol to find the shortest number of hops, i.e. shortest path, can be viewed by looking at end-to-end delay measurements. Since the end-to-end delay increases as the packets travel through more hops and so causing longer routes. During high mobility, the route discovery time increased due to frequent occurrence of invalid routes. As a result the average end-to-end delay was the highest as seen in Figures 5.4a and 5.4b. When the AODV and DSR routing protocols performed poorly, this situation is reversed, such that the average end-to-end delay decreased because the routing protocol dropped waiting packets in the queue that have the highest delay. Additionally, when a route was found, the packets with less-delay left the queue, which appeared as if the routing protocol was performing better during high mobility. Notice that the packet drops and the losses in the network are not counted in the calculation of the average end-to-end delay expressed in Equation 2.2.



(a)



(b)

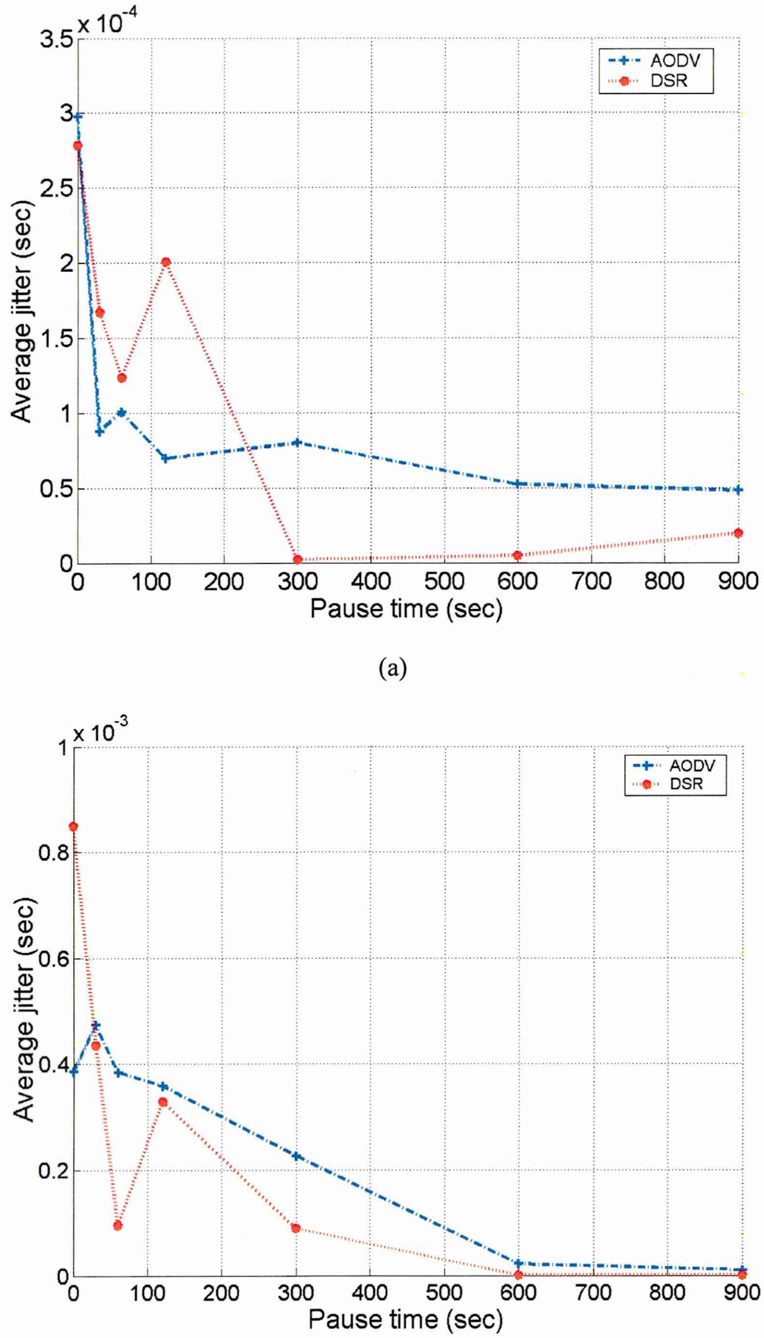
Figure 5.4: Average end-to-end delay as a function of node pause time for; (a) audio traffic and (b) video traffic

Figure 5.4a illustrates the average end-to-end delay for all the received packets that was achieved by both protocols whilst sending audio traffic. Since the maximum delay observed was 0.16ms and 0.3ms for AODV and DSR respectively, the QoS requirement of the audio application was met by both protocols. At low pause times the AODV provided a better performance. DSR exhibited a peak delay of 0.3ms at 0s pause time and then dropped down to 0.16ms delay at 120s while AODV values fluctuated around 1 ms. As the pause time was increased from 300s to 900s, the situation was reversed and DSR performed better than AODV with the average end-to-end delay equal or below 0.55ms. This behaviour can be explained as follows: the AODV performed well and kept up with routing packets with a smaller drop rate. When the pause time was increased, the AODV started dropping packets with the highest delay from its queue, which led to a decrease in the average end-to-end delay at the destination node. This was due to the default time to hold packets awaiting routes and the fixed buffer length for this protocol.

In Figure 5.4b, the QoS requirement was still met with an average end-end delay below 1.2ms for both protocols. Video traffic has a higher data rate, hence a higher end-to-end delay compared with audio application delay performance. At zero pause time, DSR had the highest delay, i.e. 1.06ms, as both protocols values fluctuated it was not obvious to see which protocol performed best. At 600s pause time, AODV was showing a higher end-to-end delay, as route finding took longer than DSR, packets were being held longer in buffers but still went through the networks, thus increasing delay. If a packet is held in the queue for more than the Time To Leave (8s), then it will be dropped, so only packets with minimum delays are forwarded. This can explain why the AODV delay was lower at 0s and 120s pause times. On the other hand, it was more obvious to see that during high pause times, when network mobility was reduced, end-to-end delay decreased to a minimum of 0.018ms and 0.013ms at 900s for AODV and DSR, respectively. In this case, links were prone to less breaks which allowed for better packet delivery.

5.2.5 Jitter

Closely related to end-to-end delay is the delay variation. It is also referred to as jitter and is another important QoS metric. ITU G.114 (c.f. Table 4.4) recommends a preferable jitter under 1ms for audio applications and 50ms for video applications. Figures 5.5a and 5.5b illustrate the jitter observed at the destination. In general, DSR experienced a higher jitter during low pause times, i.e. below 120s, for both application types. With one exception when video on-demand is transmitted with a pause time equal to 60 seconds Figure 5.5b shows that the DSR jitter (0.094ms) is lower than AODV's (0.39ms). Due to mobility, the links along the path were broken, DSR selected another available path in the cache, hence the end-to-end packet delay varied, affecting the average jitter. In this case links were more prone to breakage and to bigger losses, the AODV drops packets with the highest delays at the network layer. Thus, packets with lowest delays were forwarded, reducing average delay which also reduced the value of jitter. As shown on Figure 5.5a, the turning point came at a pause time equal to 300s, AODV jitter was higher than DSR. The difference was more notable for audio applications in Figure 5.5a where the AODV jitter was maintained around 0.5ms and DSR was kept to a minimum under 0.019ms. In a low mobility situation, packet drops were caused by congestion on the selected path. The benefit of having an alternative route available in cache helped since DSR performed better. AODV needed to re-initiate route discovery thus creating more delay before forwarding packets which also affected jitter. Overall, both protocols managed to provide a good QoS for the transmitted applications, with respect to their requirements.



(a)

(b)

Figure 5.5: Average jitter as a function of node pause time for; (a) audio traffic and (b) video traffic

5.3 Summary

A summary is presented in Table 5.1. In this chapter detailed simulations were carried-out to investigate the impact of mobility in MANET and how well popular on-demand routing protocols deal with different type of multimedia applications. Results are presented for a network under various mobility levels. They revealed that in terms of QoS performance, DSR source routing exhibited the best performance with higher delivery percentage than AODV. As stated in the previous chapters, DSR uses source routing and aggressively caches source routes, possibly multiple routes, to reuse later; AODV on the other hand, is only aware of a single route and uses a sequence number-based mechanism to keep the most recently learnt route (Royer and Das, 2000).

Table 5.1: Summary results Chapter 5

Routing Protocols	Results
AODV	• Low delay and jitter
DSR	• Best packet delivery fraction • Lower routing load • Low delay and jitter

Consequently, AODV showed the highest routing load thus consuming more bandwidth during high mobility. Mobility variations led to some packet loss highlighting the limitations of the routing protocols in dealing with route breaks as investigated by Lu *et al.* (2004) (c.f. section 4.7).

Impact of Increasing The Number of Connections on Routing Performance

6.1 Introduction

In this chapter, the effects of increasing the number of connections on the routing performance are discussed. To achieve this, the number of communication pairs in the network was varied for 5 different scenarios. This created more active routes and more hosts involved in the routing process, therefore congestion-related and mobility-related packet losses are expected. The results were extracted for each flow (i.e. connection pairs) and presented as an average for the whole network. These experiments were based on 1-10 source(s) continuously sending data (CBR over UDP) at a rate of 64Kbps for audio applications and 192Kbps for video applications to 1-10 destination(s). The nodes were moving at an average speed of 6.11m/s, the pause time was increased from 0s (high mobility) to 900s (no mobility) for each set-up and the simulation time was set to 900s. Other simulation settings are as listed in Table 4.3 (see Chapter 4).

6.2 Results and Discussion

6.2.1 Packet delivery faction

The results obtained for the packet delivery fraction is depicted in Figures 6.1 and 6.2, for both audio and video applications respectively. In general, as the number of connections increased, the percentage of successfully received packets gradually decreased. At low loads (i.e. 1 and 3 connections), Figure 6.1 shows that DSR outperformed AODV, achieving 97.5%, 99% and 100% packet delivery with 1 connection and 90.87%, 95.4% and 98% with 3 connections, for high, medium and no mobility respectively.

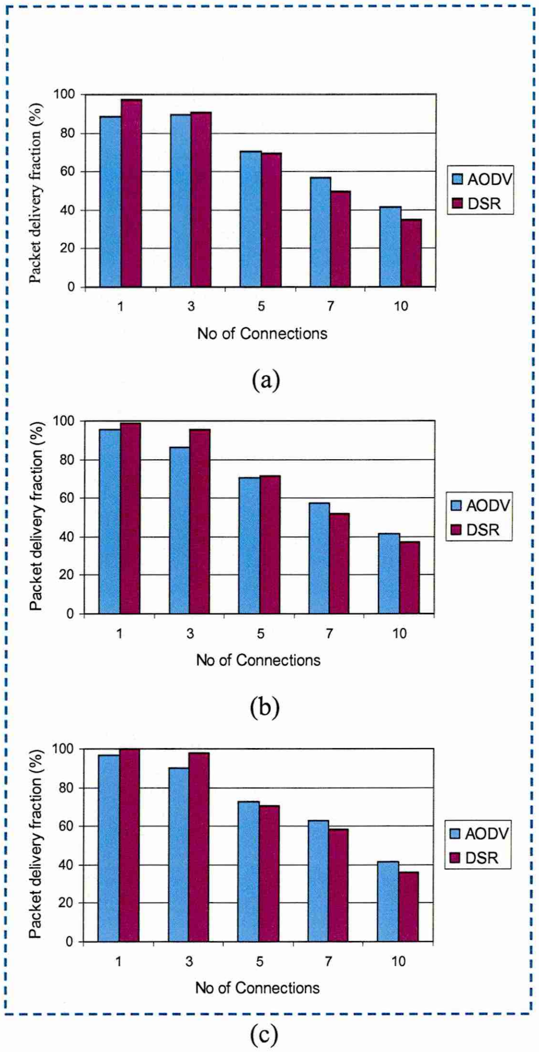


Figure 6.1: Packet delivery fraction versus Number of Connections for audio applications (a) pause time=0s (high mobility), (b) pause time=120s (medium mobility) and (c) pause time=900s (no mobility).

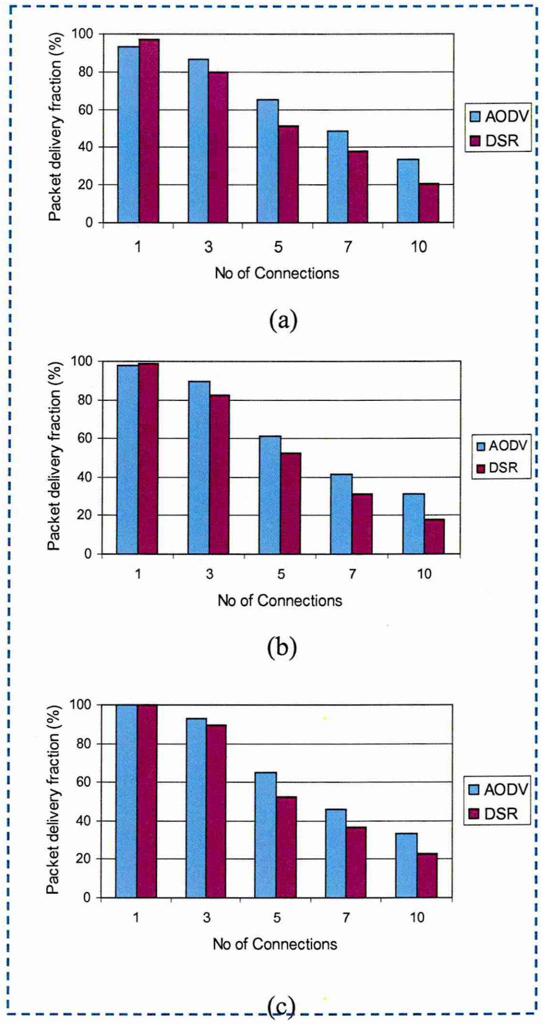


Figure 6.2: Packet delivery fraction versus Number of Connections for video applications (a) pause time=0s (high mobility), (b) pause time=120s (medium mobility) and (c) pause time=900s (no mobility).

QoS being met when pdf is above 97%, DSR managed to comply with the audio application requirements when the nodes were static at low load and when there was a single connection at high mobility. As the workload was increased the situation reversed and AODV outperformed DSR with higher delivery. AODV uses route expiry, dropping some packets when a route expires and a new route must be found. This however, gives better quality routes to AODV in general. AODV has a short convergence time for propagating changes in link states. Information about link failures will be propagated following the pointers to reach nodes using the failed link. This implies that the messaging overhead to announce link failures will be less than DSR, since data packets do not contain the complete list of all the nodes on a route in AODV. This reduces the size of the message packet. The use of broadcast during route discovery contributes to high messaging overhead, which causes a major overhead, due to periodic hello messages to maintain pointers at every node on a path. When the number of connections was greater than 3, both AODV and DSR dropped a large fraction of data packets. At the highest mobility and with 10 connections, only 35-45% of packets sent were received. The results for AODV and DSR were fairly similar under the 3 connections scenario for high mobility. This was mainly due to the congestion which occurred at the MAC layer. Because CSMA/CA was used in the simulations, packets may be dropped due to congestion for two reasons: i) the wireless channel was so busy that the backoff time exceeded the retransmission limit defined in Table 3.2. Or ii) the channel was associated with a queue that buffered all the packets waiting to be sent and when the queue was full, all further received packets were dropped.

Transmission of video on-demand traffic had slightly different results. As seen in Figure 6.2, DSR exhibited better results than AODV when there was only 1 connection. AODV met the application requirements (pdf > 95% - table 4.4) when the nodes were static, whereas DSR managed to achieve QoS for under no and medium mobility. With the number of connection increasing to 10 connections, only 15-25% of the total packets generated were received by the destination. As the nodes were randomly distributed and mobile, it was likely that several nodes were trying to access the channel in the same sensing range, thus creating MAC packet loss. As mobility decreased, the packet delivery fraction decreased much faster at high load as shown on Figure 6.2c. Similar to the scenarios with audio traffic, when the load increased, AODV performed better than DSR with higher delivery rate. However, for the video traffic, this behaviour started as soon as the number of connections was set to 3 (576 Kbps), due to the higher data rate. In Figure 6.2, AODV's pdf was 1.5-1.7 times higher than DSR's pdf at the highest load for all mobility scenarios, however when audio was sent AODV was only 1.1-1.2 times larger than DSR. Video traffic had a higher data rate and larger packet size, additionally DSR used source routing, so each data packet contained route information (including a list of all intermediate routers to each destination) making the packets larger than AODV. This resulted in a load increase on the network and more packet drops as opposed to AODV which allowed for more data packet to be forwarded through the network. For both applications, the percentage of packet delivery decreased much faster when the number of connections involved more than 3 sources.

6.2.2 Normalised routing load

The bar charts in Figures 6.3 and 6.4 show that as the number of connections increased, more routing control packets were generated due to mobility and congestion. AODV needed to repeatedly generate RREQs, more routing packets than DSR during link breaks. In one hand it wasted bandwidth which limited achievable throughput, and on the other hand, it increased collisions with other communication pairs.

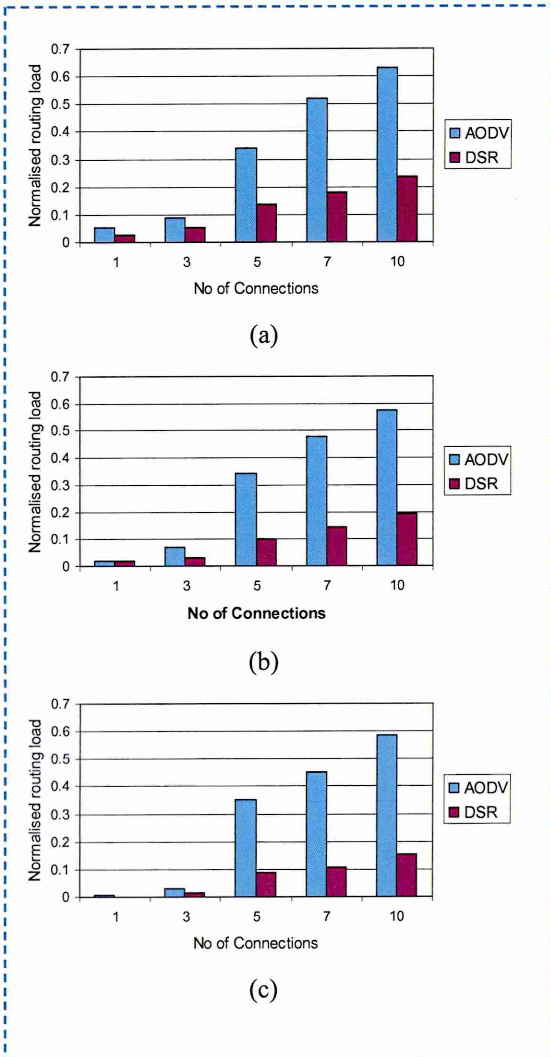


Figure 6.3: Normalised routing load versus Number of Connections for audio applications (a) pause time=0s (high mobility), (b) pause time=120s (medium mobility) and (c) pause time=900s (no mobility)

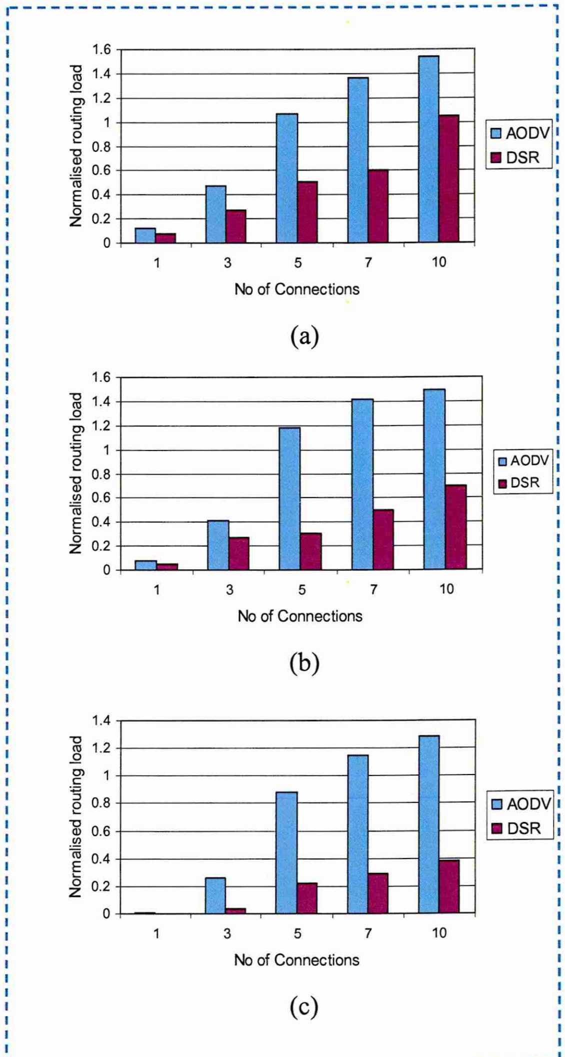


Figure6.4: Normalised routing load versus Number of Connections for video applications (a) pause time=0s (high mobility), (b) pause time=120s (medium mobility) and (c) pause time=900s (no mobility)

The use of caching made the performance of DSR always better than AODV. In general, AODV was often generating 3 times more control messages than DSR as seen in Figure 6.3, for 7 and 10 connections when sending audio traffic. Referring to Figure 6.3a, AODV routing packets represented up to 63% of the packets sent through the network under 10 connections. This was more than half of the traffic going through the nodes and emphasised on the disadvantage of using a flood-based routing mechanism as opposed to a table driven protocol (c.f. section 2.4). However, the mobility factor did not have a great impact on the routing load, all three graphs exhibited similar results with a maximum routing load of 0.58, for 10 connections in both medium and no mobility. From this, it was concluded that the generation of routing control packets was incurred mainly by link breaks related to congestion rather than mobility. Congestion occurs whenever the demand exceeds the maximum capacity of a communication link, especially when multiple hosts try to access a shared medium simultaneously.

Packets are dropped at the source if a route to the destination is not available, or the buffer that stores pending packets is full. It may also be dropped at an intermediate host if the link to the next hop is broken. AODV is based upon distance vector, and uses destination sequence numbers to determine the freshness of routes. It requires the hosts to maintain the information about on the active route. An active route is used to forward at least one packet within the past active timeout period. When a host needs to reach a destination and does not have active route information , it broadcasts a RREQ, which is flooded in the network. A RREP is unicast back to the originator of the RREQ to establish the route. Every route expires after a predetermined period of time. Sending a packet via a route will reset the associated expiry time. Default values of AODV are presented in Table 4.1.

In Figure 6.4, video on-demand traffic was sent through the network and similar observations were made, DSR outperformed AODV by far, with the routing load sometimes reduced by half of what AODV would generate. Video traffic had larger packet size thus creating more congestion than audio traffic, both protocols flood twice as much control packets for video traffic independently of mobility. Normalised routing load is calculated using routing packets, sent and forwarded data packets (c.f. Equation 4.2), hence, if the routing packet size is larger than the data packets sent the equation would become larger than 1, which is the case for video traffic on Figure 6.4. Traffic patterns had on average 1.5 connections per node. Maintaining multiple connections at each node resulted in route competition, which occurs when a route to a particular destination A is evicted from the cache upon receiving a flurry of routes to a different destination B. If a route to A is required in the future for data delivery, a rediscovery of the same destination route becomes necessary which can cause unnecessary route discoveries.

By looking closely at DSR, one can notice that as the mobility is reduced, DSR generates fewer routing packets. For 10 connections, Figure 6.4 shows the normalised routing load equals to 1.05, and 0.7 and 0.38 for high, medium and no mobility respectively. This outlines the effect of mobility on DSR, as the nodes became more static, DSR route caching technique was much more effective. An available route would be found much quicker by just looking up in routing table thus reducing routing packets overhead and transmission delay. AODV generated twice more routing packets than data packets with a maximum routing load up to 1.53, 1.5 and 1.3 for high, medium and no mobility respectively. As the mobility was reduced, the routing load slightly decreased but it was still above 1 for AODV, indicating that mobility has not a great impact on AODV unlike DSR. Whether link breaks are related to mobility or congestion, the source still needs to be notified and start flooding route requests in the network to establish routes. Most routing protocols benefit to some degree as the number of connection grows large. This is because a single route repair can potentially benefit many connections.

6.2.3 Average hop count

Figures 6.5 and 6.6 illustrate the hop count for a number of mobility factor when sending audio and video applications respectively. DSR performed better than AODV in all the values of the pause time and more so as the number of connections was increased. Such behaviour can be justified by the way routes are established in DSR. By adopting the concept of multiple routes, during its route discovering process, DSR identifies multiple routes to the target node, it also discovers routes to intermediate nodes. As a consequence, DSR is almost always able to find a valid route or can quickly update invalid ones. Source routing learns all routes in the source route and therefore does not need to send as many route requests. The reason for the increase in the number of control packets is the MAC-layer support. The increase in the number of control packets rate means that the MAC-layer will detects link failures much faster. This means that the triggered Route REPlys (RREPs) packets are sent much earlier also causing the source node to send out a new request much earlier. The DSR source routing approach affects the performance of the protocol in higher mobility levels. For example, DSR uses routes stored in its cache and does not adopt any mechanism to discard old routes. Also, in the case of multiple routes to a given node, DSR does not differentiate old and new routes and so, can continue using an old route that does not represent the best option. The results illustrated in Figures 6.3 and 6.4 verify that, for all applications, the average routing load increased with the rise of user mobility and connection pairs. Such behaviour can be justified in a simple and intuitive way. The bigger the mobility level, the higher the possibility of connectivity failure, and the higher the number of activations of route maintenance mechanisms, and so, the bigger the number of routing messages.

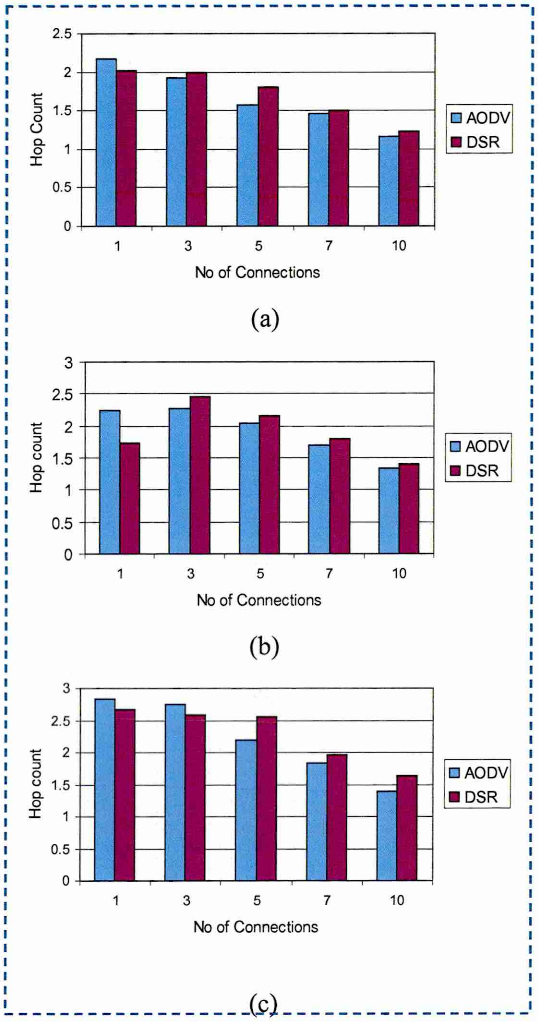


Figure 6.5: Hop count versus Number of Connections for audio applications (a) pause time=0s (high mobility), (b) pause time=120s (medium mobility) and (c) pause time=900s (no mobility).

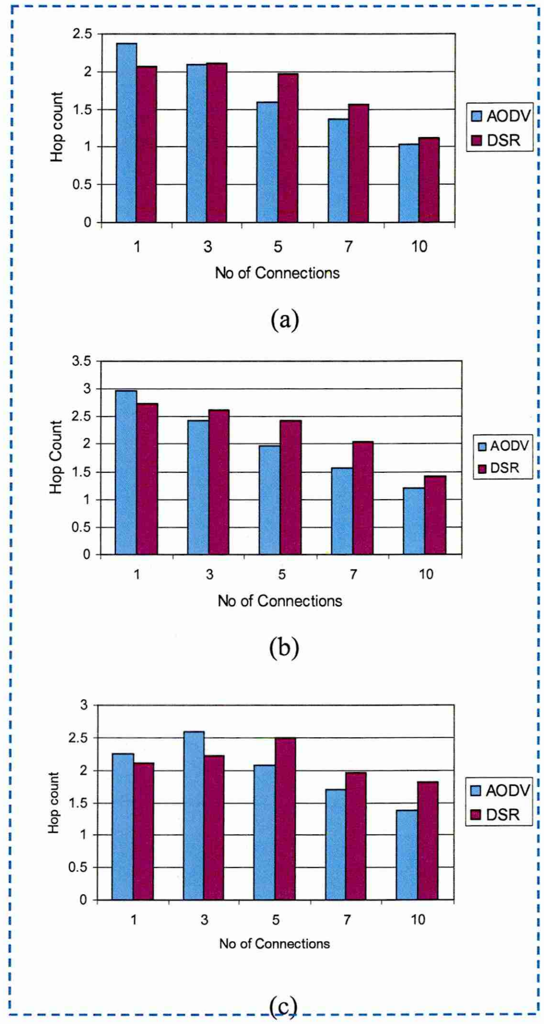


Figure 6.6: Hop count versus Number of Connections for video applications (a) pause time=0s (high mobility), (b) pause time=120s (medium mobility) and (c) pause time=900s (no mobility).

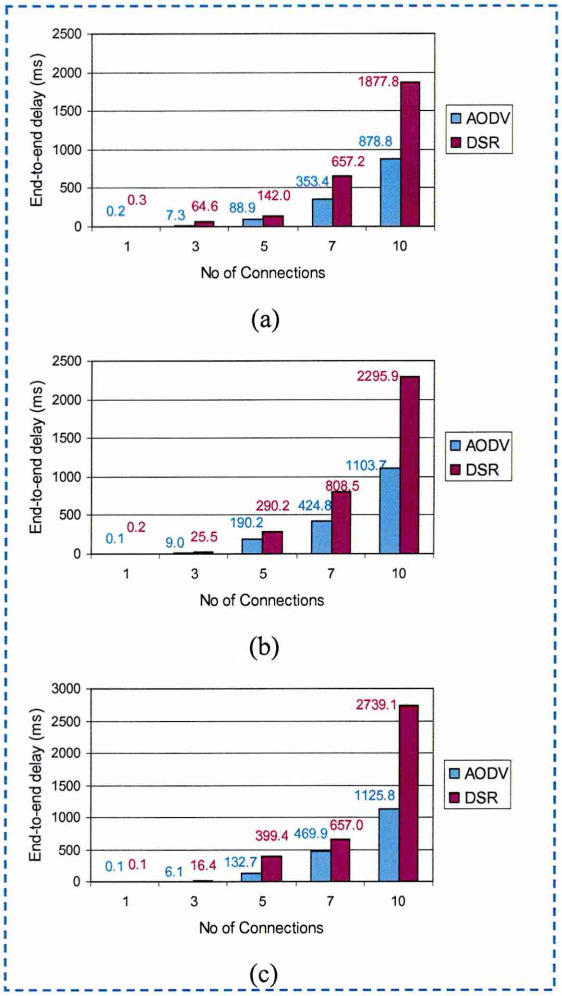
The hop count represents the average number of intermediate nodes that successfully delivered packets went through. This is closely related to how many times packets have been forwarded. Hence it can vary over time since the nodes are mobile and distributed in a random manner. AODV dropped a considerable number of packets during the routing discovery phase, as route acquisition takes time proportional to the distance between the source and the destination. However, at low load and with all the different mobility factors AODV maintained the highest hop count, when links break MAC attempted to retransmit at intermediate nodes in the route thus number of forwarding packets increased resulting in higher hop count according to Equation 4.1. In general, Figures 6.5 and 6.6 show that as the number of connections increased, DSR gradually had larger hop count than AODV, because more packets were being dropped (Figures 6.1 and 6.2) which lead to less packets being forwarded, hence decreasing the hop count.

In general, packet loss is mainly caused by packet collisions. Routing packets which are generally broadcasted are more likely to collide with other data packets in the medium. If a path is used bi-directionally, the collision rate will increase much more. The collision probability increases if the hop count from the source to the destination is greater. Packet drop occurs at different network layers. A routing agent may drop the packet if it knows that it will not be able to route the received packet, because there is no route. If there is network congestion because of high load, the interface queue may become filled up. In this case the packets received and the packets which cannot be delivered for a long time, may be dropped. Packet collisions mean that AODV is certain to transmit significantly less hello packets, consequently topology changes take longer to be broadcasted throughout the network. This causes packets to wait in IFQ, thus resulting in more delay. When AODV cannot find a new route for those awaiting packets, it drops the one that waited the longest. Therefore, only packets with the smallest delay leave the buffers. At low pause times, the traffic load becomes significant due to the flooding of the hello messages and AODV drops more packets with high waiting time in IFQ. Broken links are not detected fast enough. AODV can only detect a link breakage only after being able to send the hello messages and having lost all maximum allowed count. During this time, it will send packets without knowing links has already been broken, which leads to a high packet loss.

Mobility-related packet loss may occur at both the network layer and the MAC layer. When a packet arrives at the network layer, the routing protocol forwards the packet if a valid route to the destination is known. Otherwise, the packet is buffered until a route is available. A packet is dropped in two cases: i) the buffer is full when the packet needs to be buffered and ii) the time that the packet has been buffered exceeds the limit (8s for AODV and 30s for DSR). MAC layer packet loss occurs when the next hop of a packet is out of range at the moment the packet is sent by the MAC protocol. The reason is that the routing information is obsolete. This occurs frequently in a high mobility network rather than in a low mobility network.

6.2.4 End-to-end delay

Figures 6.7 and 6.8 represent the end-to-end delay measurement for different loads. A few parameters caused the delay to increase: packet drops occurred when the queue was overflowing thus all the buffered packets waiting to be sent created even more delay. End-to-end delay included all possible delays caused by buffering during route discovery latency at the RTR (RouTeR) level, queuing at the IFQ (interface queue), retransmission delay at the MAC layer and propagation and transfers time. While the network was congested, all this parameters were affected thus degrading the performance of the network. As the number of connections increased, link failures created more delay reaching a peak value of 2.3 sec and 1.1 sec for AODV and DSR respectively whiles sending audio (Figure 6.7). When video on-demand was sent, end-to-end delay increased to a maximum value of 1.54 sec for AODV and 7.2 sec for DSR. In general, DSR performed worse than AODV, because it did not make quick link reversals, instead referred to its topology knowledge, which was updated periodically. The delay was also affected by the high data rate (Figures 6.7 and 6.8). The buffers became full much quicker so packets had to stay in the buffers for a much longer period of time before they were sent. This can clearly be seen at the highest number of connections. When audio traffic was sent, end-to-end delay increased gradually with higher workload while the mobility factor did not incur a great difference on the measurements. AODV provided a better QoS for the transmitted applications, as seen on Figure 6.7a for high mobility and workload set to 7 connections. The delay for AODV was 353 ms which complied with audio applications delay requirement (as shown on Table 2.3) whereas for DSR, the delay rose to 657 ms. As the application data rate was higher for video, Figure 6.8a illustrates the large increase in delay due to packet being generated faster than the MAC could send them, thus longer buffering time was required. AODV was still the protocol that performed best with the lowest delay, however QoS requirement was only met for workload below 7 connections, since at this particular connection factor, the delay is 409 ms which is slightly above the acceptable value (i.e. 400 ms).



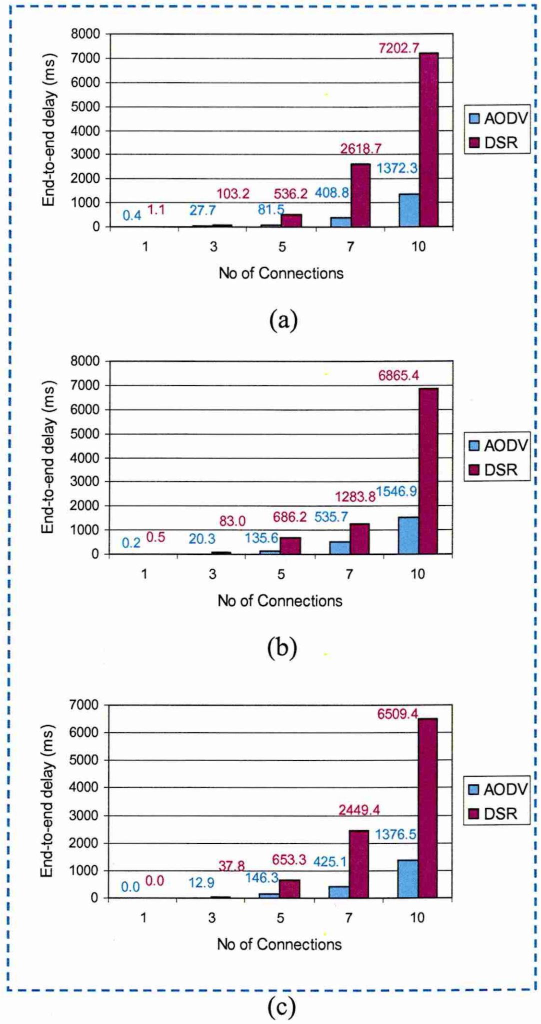


Figure 6.7: End-to-end delay versus Number of Connections for audio applications (a) pause time=0s (high mobility), (b) pause time=120s (medium mobility) and (c) pause time=900s (no mobility).

Figure 6.8: End-to-end delay versus Number of Connections for video applications (a) pause time=0s (high mobility), (b) pause time=120s (medium mobility) and (c) pause time=900s (no mobility).

As the number of connections reached 10, the delay reached a peak value of 7.2 sec when using DSR. This was about 5 times higher than AODV delay in high mobility.

Low mobility meant that already found routes were valid for a much longer time period. This meant the found routes can be used for more packets. Even packets that stayed in the buffer for a long time had a chance to get through. When mobility increased, more routes became invalid and new requests were necessary. While the requests flooded the network in search for a new route, buffers were filled and packets were dropped. These packets were the ones that stayed in the buffers for the longest time and therefore only packet with short delays remained, hence the low average delay shown in Figures 6.7a and 6.8a. As the mobility decreased delay was highest as observed in Figure 6.7c and 6.8c. In Figures 6.7a and 6.8a, when the number of connections was set to 7 and 10, mobility was high and the topology changed frequently, only 20-40% of the packets got through the network for both applications. Topology changes meant that the protocol needed more time to converge before the packets could be sent. The buffers would therefore be congested almost all the time so the packets that actually got through had approximately the same delay. As identified in previous studies (Lu *et al.* 2003; Johnson et al. 2003; Royer and Toh 1999), DSR is more suitable for small networks and moderate mobility levels. In addition, although AODV also adopts a drop-tail queue management approach, it poses a limit of 30s for queuing time. Any packet that stayed in the queue for more than 30s is discarded. As a consequence, the end-to-end delay of each successfully delivered packet was bounded and decreased the average delay.

6.2.5 Jitter

Jitter is another metric for QoS describing how much the packets vary in latency and is determined by calculating the average variation of end-to-end delay. The possibility that two or more stations starting to access the channel simultaneously can be very high. Consequently, all data packets that are dropped due to collisions require retransmission by the MAC protocol; this in turn increases the time between any two consecutive packets that are successfully received at the destinations. In Figures 6.9 and 6.10, AODV exhibited the best performance in general. When audio application was sent, DSR had a very large jitter with a maximum of 2.6 sec when the nodes were static (c.f. Figure 6.9) and 7.7s when video traffic was sent under medium mobility (c.f. Figure 6.10). As seen in Figure 6.9, when the number of connections was set to 1 and 3, the jitter was kept to a minimum (0-6ms). In terms of applications QoS, AODV managed to achieve the requirement for number of connections up to 5 at high mobility with jitter below 100ms, whereas DSR complied with the requirements for number of connections up to 5 with maximum jitter equal to 49.74ms. Jitter gradually increased as the network became more congested for both protocols, yet DSR showed the biggest increase. AODV was stable and reliable since during link breaks it would not initiate RREQ from source. But instead the node upstream the broken link, would attempt to repair the link locally given that the destination was no further than a maximum default distance (i.e. hops) known as MAX_REPAIR_TTL. This decreased the delay and further reduces the occurrence of congestion because of excess data packets buffered in queues. When video applications were sent jitter greatly increased as shown on Figure 6.10. AODV was still achieving the best results as opposed to DSR. Jitter was higher in this case, since video data rate is faster, more packets were generated and buffered, which had a negative impact on DSR. More so, when the number of connections was set to 10, DSR jitter was 6 times higher as compared to AODV. In this scenario, QoS was met by AODV when the number of connections was set between 1 and 3, however DSR was only able to meet the applications' requirements for the same load and with a pause time equal to 900s.

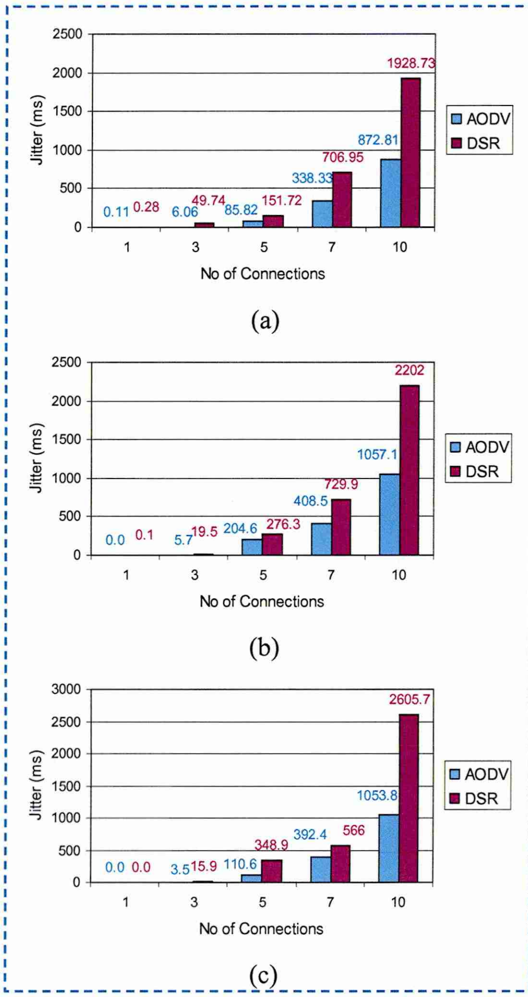


Figure 6.9: Jitter versus Number of Connections for audio applications (a) pause time=0s (high mobility), (b) pause time=120s (medium mobility) and (c) pause time=900s (no mobility).

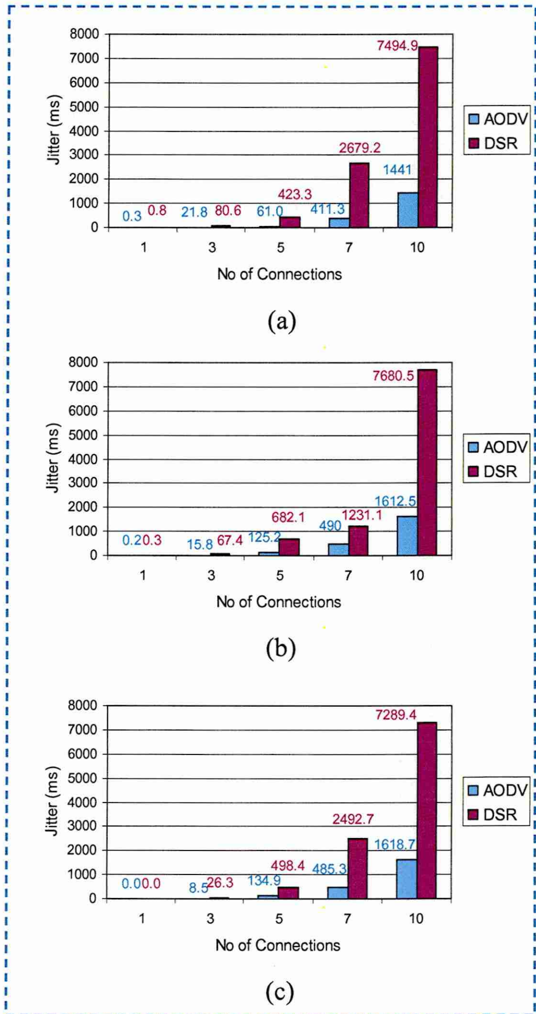


Figure 6.10: Jitter versus Number of Connections for video applications (a) pause time=0s (high mobility), (b) pause time=120s (medium mobility) and (c) pause time=900s (no mobility).

6.3 Summary

In this chapter experiments were carried to investigate the impact of increasing the number of connections hence imposing demanding conditions to the network. Results showed that as the number of connections is increased the network performance quickly degrades. Initially, source routing (i.e. DSR) performs best however as soon as the number of connections is higher than 3, distance vector mechanism (i.e. AODV) achieves a higher delivery fraction which is consistent with results in (Das *et al.* 2000). AODV and DSR are periodic protocols so they suffer in latencies in their delay in adapting to changing topology. Low pause times also introduce increased delay, because of packets waiting in interface queues while routing protocols tried to find a valid route to the destination. Protocol control message collisions increased the waiting duration of such packets. After the maximum waiting time specified by the protocol setup, packets are dropped. This gives the impression of decreased delay, while the successful delivery rate is also reduced at low pause times. A summary is provided Table 6.1.

Table 6.1: Summary results Chapter 6

Routing Protocols	Results
AODV	• Outperforms DSR when number of connection increased • High routing load
DSR	• Poor packet delivery fraction with many sources and high mobility • High jitter • High end-to-end delay

The evaluation carried-out so far has allowed for a better understanding of the WLAN routing protocols mechanisms and their advantages and disadvantages in various conditions. All the metrics were relatively dependant on one other. Additionally, this can also be viewed as a sound baseline for network performance improvement and QoS solutions provisioning for MANETs. According the extensive literature review carried out in this research study, It is necessary to propose an analysis method to assess the network and the application performance. An intelligent system such as fuzzy logic can prove to be valuable in combining this metric to obtain a general view of the QoS provided to the application. This is the objective of the next chapter.

Network Quality of Service Performance Analysis using Fuzzy Logic Approach

7.1 Introduction

Due to the inherent nature of mobile wireless networks gathering precise network information can be a difficult task for a network administrator. Therefore, the aim of this chapter is to simulate appropriate wireless networks configurations, develop a Fuzzy Inference System (*FIS*) to intelligently assess QoS achieved during the transmission of audio and video applications using on-demand routing protocols. This value represents the QoS level provided to the applications based upon the network conditions compared to the QoS level required for those applications.

This chapter is organised as follows: Section 7.2 outlines the related work relevant to this study. Section 7.3 describes the benefits of using a fuzzy logic approach for this work. The fuzzy logic assessment approach is presented in Section 7.4. Experimental and analysis procedures are provided in Section 7.6. The results and discussions are included in Section 7.7. This chapter is then concluded by a summary given in Section 7.8.

7.2 Related Work

Emerging new technologies, such as handheld personal digital assistant (PDA) or laptop computers, allow users to run diverse multimedia applications whilst on the move. This has led to the need to understand the behaviour of audio and video traffic as it affects the end-user's perception of quality. Due to the mobility in wireless ad-hoc networks, topology updates become increasingly frequent. Collected information becomes outdated as soon as a new topology emerges. Additionally, QoS parameters must represent the network properties, but the challenge is to find a way to evaluate these network states information in the most effective way. By representing the QoS parameters as a fuzzy goal one avoids the need to deal with multiple data parameters and this should allow for appropriate decision making, in from the viewpoint of network performance improvement.

A number of studies have used fuzzy logic for network analysis problems, for example (Saraireh *et al.,* 2004; We and Chen, 1999; Oliveira and Braum, 2004; Pitsillides and Sekercioglu, 1999; Fernandez, et al., 2003). To the best of the author's of this thesis knowledge, the only work focusing on QoS evaluation using fuzzy logic prior to this current work has been by Saraireh *et al.* (2004). In their study, a fuzzy logic approach was used to evaluate the QoS for image transmission over an emulated network and the frame rate was considered as the required reference for assessing the quality of the image received.

7.3 The Benefits of Applying Fuzzy Logic to Wireless Ad Hoc Networks

Fuzzy logic systems are flexible mathematical functions which can approximate other functions or, in this present case, just data (measurements) with a desired accuracy (Babuska, year). This property is called general function approximation (Kosko, 2004; Wang, 1994; Zeng and Singh, 1995). Compared to other well-known approximation techniques such as artificial neural networks, fuzzy systems provide a more transparent representation of the system under study, which is mainly due to the possible linguistic interpretation in the form of rules (c.f. Chapter 3). The logical structure of the rules facilitates the understanding and analysis of the model in a semi-qualitative manner, close to the way humans reason when making real world decisions.

The use of fuzzy logic to describe the state of a connection mimics how a person makes decision but only much faster. It is used here, to add more flexibility and capability for operating with the imprecise information due to node mobility in the wireless ad-hoc network. Fuzzy logic incorporates a simple approach to evaluate the QoS level rather than attempting to model the traffic characteristics mathematically. Its characteristics make it suitable for evaluating the QoS where the uncertainties and requirements of combination of more than one parameter are present. The fuzzy logic model is relying on one's network experience rather than detailed understanding of traffic characteristics. Fuzzy logic uses non-numerical terms is imprecise and yet very descriptive of how well data are received. Therefore, fuzzy logic is essentially a computational method that can be used to associate different metrics so as to produce a crisp value that can describe the network state whilst being fast and accurate (Mendel, 1995).

7.4 Assessment Approach Description

In the following sections, the use of intelligent methods to evaluate the network QoS is described. A performance assessment method for compiling the actual network QoS experienced by the network end-users has been proposed based on a fuzzy logic system. The results obtained, using this approach, are illustrated and discussed. This approach has been designed based on the information and background provided in Chapter 3.

7.4.1 Traffic and Scenarios Characteristics

In this chapter, to demonstrate the application of the fuzzy logic approach, the input data used were obtained following the experimental procedure detailed in Chapter 3 and the results presented in Chapters 5 and 6. The proposed approach was evaluated through simulating single and multiple audio and video connections in a mobile wireless ad-hoc networks composed of 50 nodes. Both applications were modelled using the Constant Bit Rate (CBR) data in conjunction with the User Datagram Protocol (UDP). The audio traffic was characterised by a packet size of 128 bytes and a data rate set to 64Kbps, whereas, video traffic was modelled with a packet size and a data rate set to 1024 bytes and 192Kbps, respectively. For single connection scenarios, pause time was varied to create mobility and results were gathered in two separate simulation sets, one after audio traffic was transmitted and another when video was transmitted. The impact of traffic load was also assessed by simulating a wireless ad-hoc network with multiple connections (i.e. 3, 7, 5, and 10). The results are presented using two different on-demand routing protocols, AODV and DSR. Simulations were repeated 9 times, each time using different mobility scenario for the same set of variables, this introduced randomness in the network starting conditions. For each value of pause time, 9 output files were obtained and the packet events were averaged to obtain a true representation of the different QoS metrics measured. The simulation time was set to 900 seconds to obtain accurate and consistent results in a steady state condition.

7.4.2 Fuzzy Inference System for End-User QoS Assessment

A fuzzy logic approach for assessing the QoS provided to audio and video traffic has been developed. The structure of the developed approach is illustrated in Figure 7.1. The Fuzzy Inference System (FIS) is composed of four main blocks, these are: fuzzy inputs, fuzzy rules, fuzzy inferencing and fuzzy output. Since the transmitted applications are time-sensitive, the most pertinent parameters that were used as fuzzy inputs were delay, jitter and packet loss. In this study the Gaussian membership function was used for the input variables because of its smoothness, computing simplicity and concise notation (Cheng and Marsi, 2001). The assigned membership functions are defined in such a way that they overlap to account for uncertainty on the boundaries.

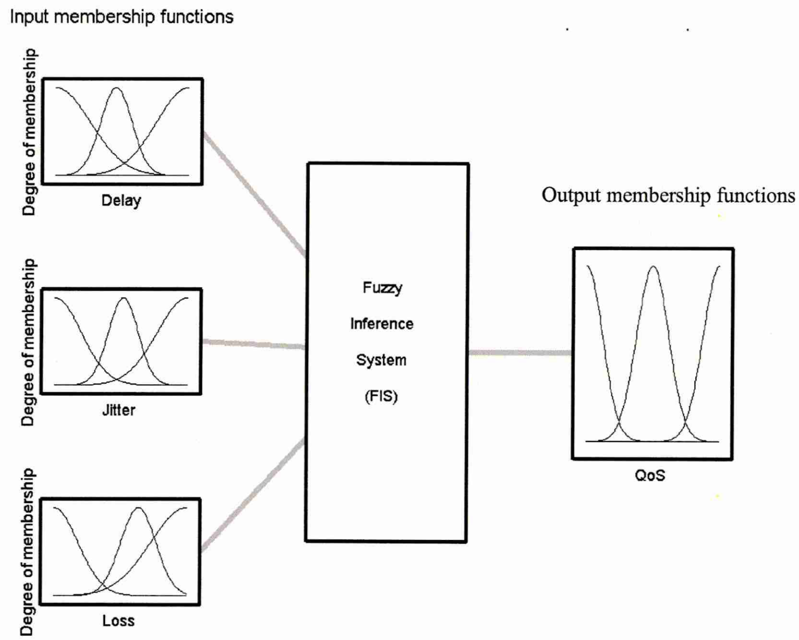


Figure 7.1: FIS Structure for QoS assessment

7.4.2.1 Fuzzy System Inputs

According to the QoS requirements discussed in Chapter 2 (cf. Table 2.3) for each parameter every single fuzzy input was represented by three fuzzy sets to produce the required membership functions. Figure 7.2 illustrates the membership functions set for the inputs parameters: delay, jitter and packet loss and the output parameter: QoS used for video on-demand applications. The corresponding fuzzy linguistic variables were classified as: low, medium or high for jitter, delay and packet loss and poor, good and excellent for QoS. The Gaussian membership function μ_X associated with each element x given in Equation 7.1, where c_i and σ_i are the mean value and the standard deviation (width) of the ith fuzzy set X^i, respectively.

$$\mu_X(x_i) = \exp\left(-\frac{(c_i - x)^2}{2\sigma_i^2}\right) \qquad 7.1$$

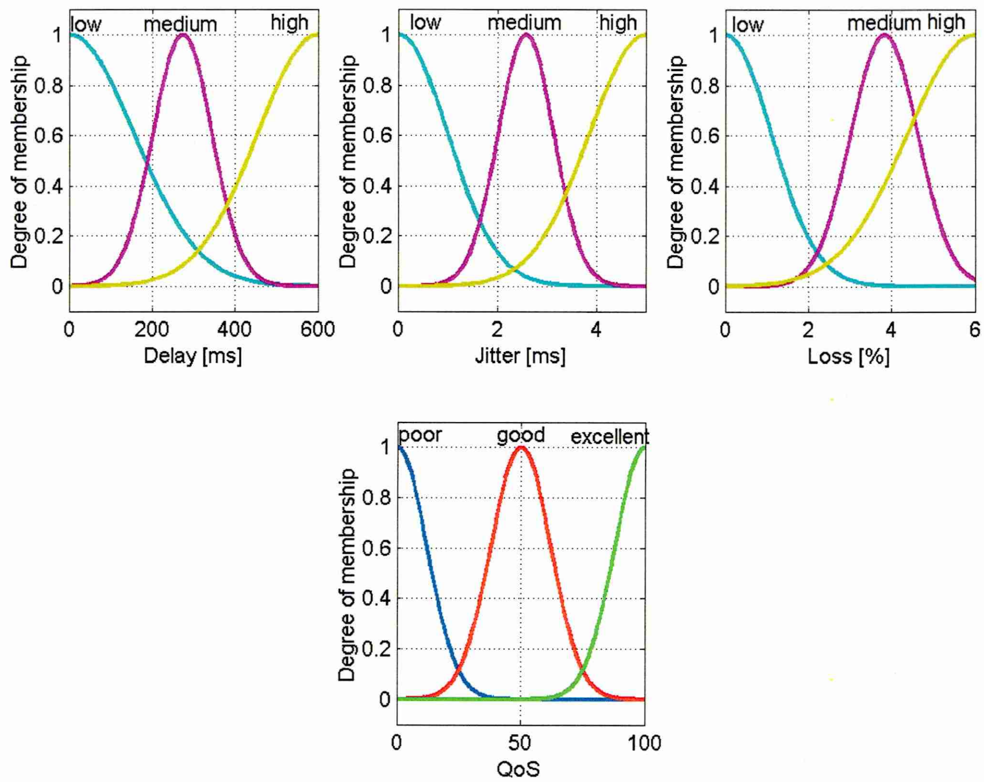


Figure 7.2: Fuzzy logic membership functions video

The thresholds values of the fuzzy sets are defined in Tables 7.1 and 7.2, for audio IP telephony and video on-demand respectively. These values were selected based on the QoS requirements (c.f. Chapter 4, Table 4.4) of each input parameter and in order to provide a reliable output of the received application QoS.

Table 7.1: Fuzzy Gaussian membership functions parameters of audio inputs and output

Label	Input variables						Label	Output variable	
	Delay (ms)		Jitter (ms)		Packet Loss (%)			QoS (%)	
	Mean	SD	Mean	SD	Mean	SD		Mean	SD
Poor	156	0	0	1	0	1	**Poor**	0	12
Medium	273	71	2.58	0.56	3.28	0.83	**Good**	50	12
High	600	146	5	1.14	6	1.62	**Excellent**	100	12

Table 7.2: Fuzzy Gaussian membership functions parameters of audio inputs and output

Label	Input variables						Label	Output variable	
	Delay (ms)		Jitter (ms)		Packet Loss (%)			QoS (%)	
	Mean	SD	Mean	SD	Mean	SD		Mean	SD
Poor	0	156	0	6	0	0.55	**Poor**	0	12
Medium	273	71	15.51	3.36	1.64	0.42	**Good**	50	12
High	600	146	30	7.17	3	0.81	**Excellent**	100	12

7.4.2.2 Fuzzy rules

The following stage applied a set of fuzzy rules over fuzzy sets using inference engine. The fuzzy rules are presented in Figure 7.3. The number of fuzzy rules is proportional to the number of input variables and the number of sets associated with each input. However, only the rules making the greatest impact on the output are stated in Figure 7.3. The devised QoS assessment approach had nine rules produced from a combination of three input variables with each input represented by three fuzzy sets. This rule base provided the various QoS level for different ranges of delay, jitter and packet loss inputs.

"IF Delay is Low AND Jitter is Low AND Loss is Low THEN QoS is Excellent"
"IF Delay is Medium AND Jitter is Low AND Loss is Low THEN QoS is Good"
"IF Delay is Low AND Jitter is Low AND Loss is Medium THEN QoS is Good"
"IF Delay is Medium AND Jitter is Low AND Loss is Medium THEN QoS is Good"
"IF Delay is Low AND Jitter is Medium AND Loss is Low THEN QoS is Good"
"IF Delay is Medium AND Jitter is Medium AND Loss is Low THEN QoS is Good"
"IF Delay is Low AND Jitter is Medium AND Loss is Medium THEN QoS is Good"
"IF Delay is Medium AND Jitter is Medium AND Loss is Medium THEN QoS is Good"
"IF Delay is High OR Jitter is High OR Loss is High THEN QoS is Poor"

Figure 7.3: Rule base of fuzzy inference system

7.4.2.3 Fuzzy system output

The type of FIS technique was Mamdani (also referred to as “max-min”), which is the one most widely used (Lee, 1990_a). The fuzzy input (crisp input) values were mapped into membership functions (fuzzification process) and assessed according to the rules considered. The fuzzy QoS output was defined as a Gaussian membership functions composed of three symmetrical fuzzy sets ranging from 0 - 100. These fuzzy set were classified with the following linguistic variables: Poor, Good and Excellent as illustrated in Figure 7.2. Fired rules are numerically aggregated by fuzzy set union and then collapsed (defuzzified) to yield a single crisp output. The Centroid (Ross, 2004) method expressed in Equation 3.2 was used as the defuzzification method to produce the crisp (non-fuzzy) QoS output. In the Centroid method, the real value of the output variable is computed by finding the variable value of the centre of gravity of the membership function for the fuzzy output.

7.5 Analysis Procedure

The critical assessment of the network performance was carried out following the stages illustrated in Figure 7.4. This assessment was based on results obtained Chapters 5 and 6. QoS metrics such as packet delivery fraction, delay, jitter and packet loss were extracted for each traffic type. For scenarios with multiple connections, each flow is evaluated separately and its associated parameters were obtained. An average was computed and fed into the FIS for intelligent overall QoS assessment. The output QoS was then used to interpret the way the network had dealt with the transmitted applications with respect to the traffic types and routing protocols used.

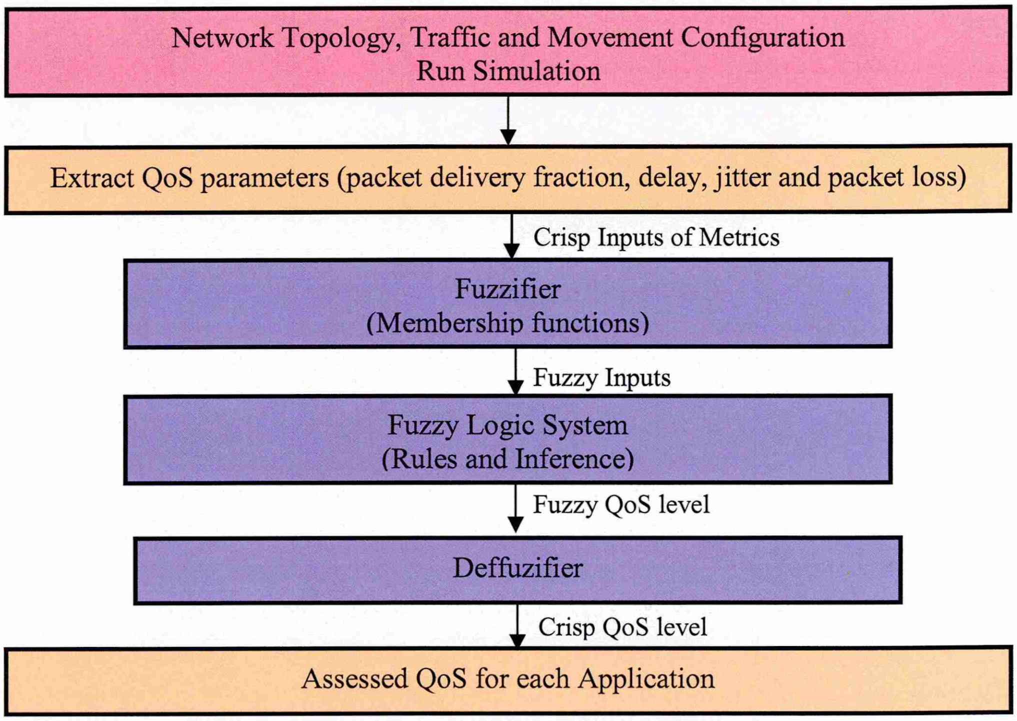


Figure 7.4: Fuzzy logic based analysis procedure

7.6 Results and Discussion

A typical set of results obtained using the FIS QoS evaluation method is shown in Table 7.3 for audio IP telephony and Table 7.4 for video on-demand. It can be observed that the mechanism has successfully processed the QoS requirements of the video and audio application. As expected, for high values of delay, jitter and packet loss resulted in a poor QoS. However, medium and low values of these parameters resulted in good and excellent QoS level, respectively.

Table 7.3: A sample of QoS input parameters with their expected QoS outputs (IP telephony based fuzzy system)

Delay (ms)	Jitter(ms)	Loss (%)	Overall QoS (%)	QoS Level
20.7	0.3	0.2	90.3	Excellent
33.5	0.92	0.46	87.1	Excellent
139	0.2	0.37	72.9	Excellent
164	0.94	1.77	60.5	Good
244	1.85	2.38	51	Good
365	0.2	0.35	42.9	Good
359	0.44	4.71	34.4	Poor
400	3.84	3.27	22	Poor
522	4.38	0.81	10.7	Poor

Table 7.4: A sample of QoS input parameters with their expected QoS outputs (video on-demand application based fuzzy system)

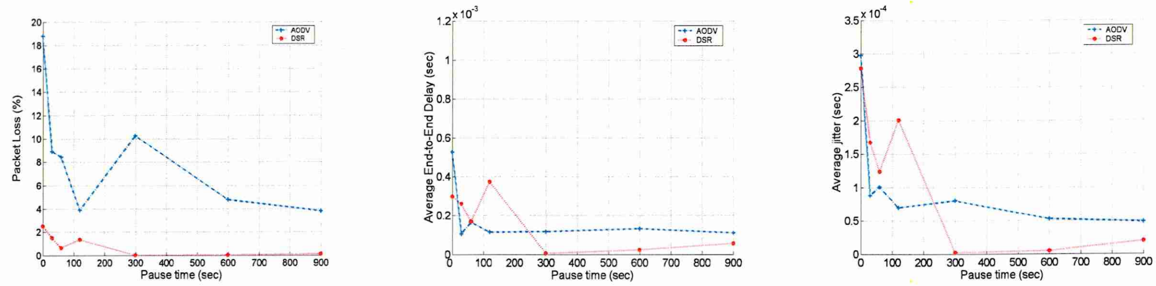
Delay (ms)	Jitter(ms)	Loss (%)	Overall QoS (%)	QoS Level
30.3	2.26	0.04	83	Excellent
60	15.6	0.36	71.6	Excellent
228	4.52	0.16	67.8	Excellent
36.7	4.12	0.86	51.2	Good
87.8	2.53	0.97	48.9	Good
343	14.2	1.83	39.7	Good
110	45.9	0.93	34.5	Poor
292	37.1	2.51	23.6	Poor
560	8.64	2.53	11.4	Poor

The delay requirements of the various applications are different. Video on-demand, for example, requires low delay (less than 400 ms) (ITU, 2001), a percentage packet loss under 3% (Boyce and Gaglianello, 1998) and an acceptable jitter under 50ms. Therefore, to obtain the best QoS, these parameters should be kept to the minimum (Dalgic and Tobagi, 1996). In order to provide an excellent QoS for audio IP telephony traffic, delay, jitter and packet loss have to meet stringent requirements. End-to-end delay cannot be greater than 400 ms (ITU, 2001), jitter has to be limited to less than 5 ms to ensure smooth playback at the receiving end. These values have been used to draw a conclusion on the QoS performance of the network.

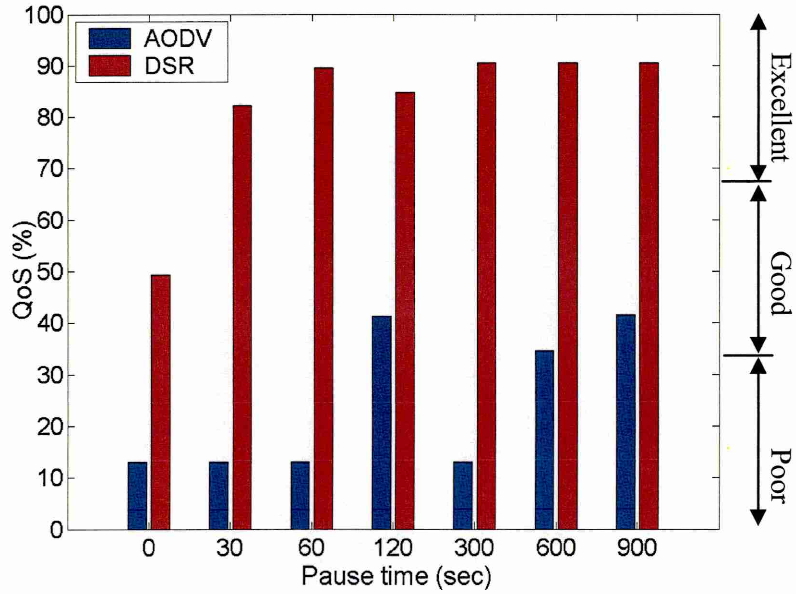
7.6.1 FIS QoS assessment for a single connection

For a single connection, the QoS perceived by the end-user is mainly affected by the mobility factor which in this study is represented by the pause time. QoS for audio IP telephony and video on-demand applications is analysed, using two on-demand routing protocols. It also outlines how both protocols, AODV and DSR, are incapable of providing QoS during various demanding conditions and these will be identified here.

Figures 7.5 and 7.6 illustrate the FIS output when audio IP telephony and video on-demand applications, respectively, were transmitted.



(a)

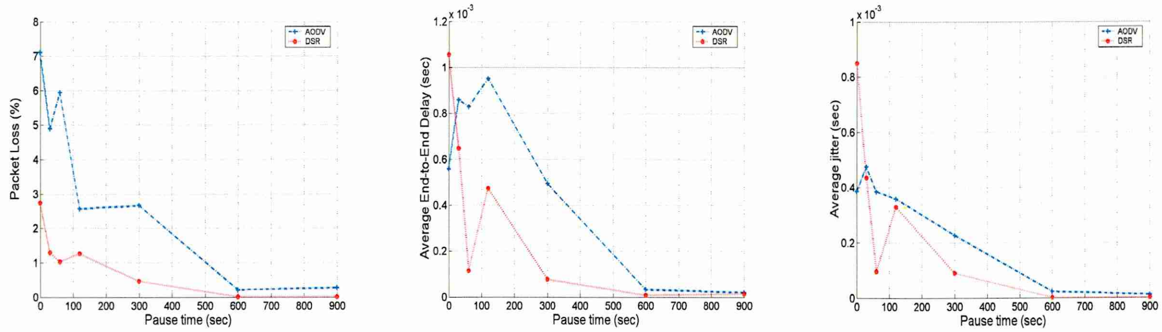


(b)

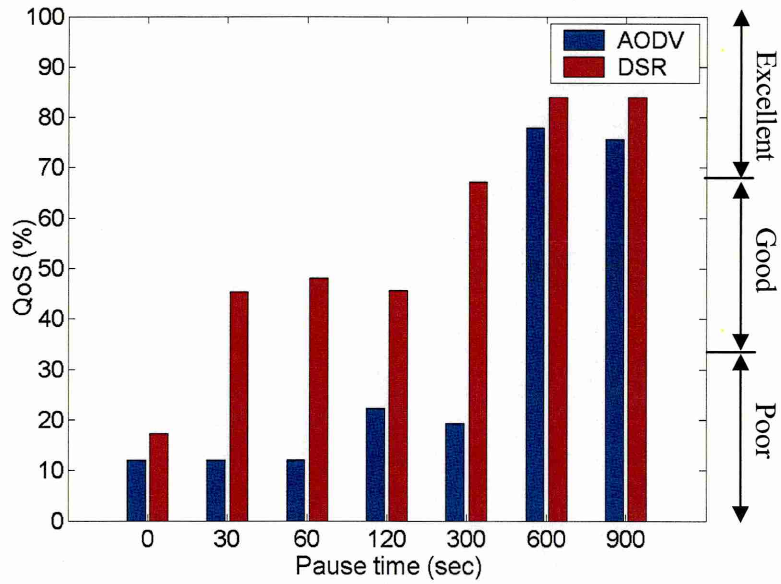
Figure 7.5: (a) QoS parameters input (c.f. chapter 5) and (b) End-user QoS output during audio IP telephony transmission

In general, it can be seen that DSR exhibited the best overall performance. At high mobility (i.e. zero second pause time), DSR achieved medium QoS with mean value equal to 49.3% when audio is transmitted and a poor QoS with a mean value equal to 17.3%, for audio and video respectively. As the pause time increased to 900s, the QoS achieved an excellent level, with a mean value of 90.5% and 83.9% for audio and video, respectively for the same protocol. This implied that DSR provide the best route during traffic flow. This scenario was based on only 1 source and 1 connection were set up, thus reducing the probability of congestion due to collisions, making the route discovery for DSR less challenging. On the other hand, when the pause time was equal to zero second AODV exhibited a poor QoS with a mean value equal to 13.1 % and 12.1%, for audio and video respectively. As the network became less mobile the QoS increased to a medium level equal to 41.2% for audio transmission and an excellent QoS level reaching a mean value of 75.6% for video on-demand applications at pause time equal to 900s. This was unexpected, as videoconferencing is more bandwidth demanding due to its characteristics, however since this value is a combination of several parameters, this difference can be due to the fact that audio applications have a more stringent requirement on jitter.

In detail, when requesting a new route, DSR first searches the route cache containing information it has learned over the past routing discovery stage. In DSR, route information is not only added to the route reply packet but also in the data packet which ultimately increase the data packet size (Yuan and Li, 2006) which in turn affects the QoS. Throughout this particular experiment, DSR protocol exhibited stability and reliability even though the literature (Biao *et al.,* 2006) has drawn the conclusion that when the pause time is reduced DSR has higher delay than AODV. During the route discovery phase, AODV and DSR temporarily buffered the data packets while they searched for the route to the destination. When the source receives a route reply packet, it empties all the data packets with the buffer whose destination correspond to the route just obtained. In this study the buffers employed for AODV and DSR had a timeout of 30 seconds and a maximum capacity of 64 packets. If the route search took longer, packets entering the queue were dropped.



(a)



(b)

Figure 7.6: (a) QoS parameters input (c.f. chapter 5) and (b) End-user QoS output during video on-demand transmission

7.6.2 FIS QoS assessment for multiple connections

Tables 7.5 and 7.6 present the results obtained for multiple connections IP telephony and video on-demand connections respectively. It can be observed that medium (i.e. 33% - 66%) and excellent (above 67%) QoS was achieved only for scenarios with 1 connection during both audio and video transmission. Packet loss in MANET was high, consecutive packets were heavily dropped due to congestion and link breaks. Packet loss has to be at very low level (less than 3%) in order to have acceptable QoS. The QoS parameters for each connection and fed into the QoS assessment FIS to evaluate the overall QoS. The result output showed a poor level of QoS (i.e. less than 33%) as the number of connections was increased. The degradation of the QoS for video and audio connections was due to high values of delay and jitter which also resulted in high variation in the assessed QoS.

Both AODV and DSR continue to use the original shortest path when traffic is sent more frequently than the route timeout. They restart their route timers each time a data packet is successfully sent, therefore they loose the ability to take advantage of shorter routes that become available on the network. Paths that use long hops, reduce the capability to meet hop count requirements and degenerate overall network performance by wasting bandwidth (Novatnak *et al.*, 2005).

As shown in Table 7.5, DSR exhibits the best performance at a pause time of 120s and 900s. The QoS is equal to 24.7% as opposed to 13.1% when the number of connections was equal to 3. However, this value is still showing that DSR cannot provide a good QoS to the IP telephony application. This is explained by the way DSR routing mechanism functions. As discussed in detail in previous chapters as the pause time increased, nodes became less mobile thus reducing the occurrence of link breaks. DSR manages to maintain stable and reliable routes to transmit the traffic by mainly avoiding flooding since it has information of alternative routes on its cache memory. At low pause times, nodes were moving constantly and route breaks were at their maximum hence increasing delay and packet loss during route re-discovery. Poor values on these parameters led to a low overall QoS especially for AODV.

Table 7.5: FIS percentage QoS assessment for audio connections

	Pause time (sec)					
No. of Connections	0		120		900	
	AODV	DSR	AODV	DSR	AODV	DSR
1	13.1	49.3	41.2	84.8	41.2	84.8
3	13.1	13.1	13.1	24.7	13.1	24.7
5	13.1	13.1	13.1	13.1	13.1	13.1
7	13.1	13.1	13.1	13.1	13.1	13.1
10	13.1	13.1	13.1	9.3	13.1	9.23

Table 7.6: FIS percentage QoS assessment for video connections

	Pause time (sec)					
No. of Connections	0		120		900	
	AODV	DSR	AODV	DSR	AODV	DSR
1	12.1	17.1	22.2	45.7	75.6	83.9
3	12.1	12.1	12.1	12.1	12.1	12.1
5	12.1	12.1	12.1	12.1	12.1	12.1
7	12.1	12.1	12.1	12.1	12.1	12.1
10	12.1	12.1	12.1	12.1	12.1	12.1

7.7 Summary

This chapter presented a method to assess the overall QoS performance provided to IP telephony and video on-demand traffic by popular on-demand routing protocols. The method employed was based on fuzzy logic and allowed the combination of critical metrics to obtain a concise view of the level of service under various network conditions. The measured QoS has been classified into Excellent, Good, and Poor categories. In addition, for each application, based on the proposed system, the overall QoS have also been obtained under mobility scenarios and the number of connections scenarios. As the network conditions became challenging to the routing protocol, network performance quickly degraded. Both routing techniques were unable to forward the multimedia traffic and with the appropriate level of QoS. The measured QoS using the proposed evaluation systems was a superior indication of the network conditions and resources availability.

Conclusion and Future Work

This chapter concludes the thesis and identifies some future research directions.

An ad-hoc wireless network is a distributed system in which wireless nodes are dynamically self-organised into an arbitrary network. Each node in the network acts as a router, forwarding packets to other nodes. Due to mobility and a lack of infrastructure support, network topologies are constantly changing and the communication capabilities of the network are limited by the bandwidth and the battery power of the network nodes. Due to these factors, it is more challenging to provide QoS in ad-hoc networks.

In this thesis, an intelligent method is proposed that is capable of assessing the QoS level provided to audio or video multimedia applications and make use of the results in the purpose of enabling QoS routing. In chapter 2, an extensive literature review is carried out, describing the characteristics of wireless network technologies and more precisely, ad-hoc network technologies. An emphasis on routing protocols, their mechanisms and their limitations in providing QoS is provided in this chapter. The challenges and difficulties of QoS routing in ad-hoc network research were also outlined. Several definitions of quality of service were presented and a discussion of some parameters that are used to evaluate QoS from a technical perspective is carried out.

After the introduction of QoS based routing, this study tackled the research challenges of QoS routing in ad-hoc networks and focused on QoS support in ad-hoc networks based on simple and robust protocol - AODV and DSR. To perform a sound evaluation of the current routing protocols and their ability to provide QoS, simulation experiments were carried out using NS-2 network simulator. The performance evaluation was based on the analysis of several applications' QoS parameters (delay, jitter and packet delivery fraction). Chapter 4 outlined the various experiments set-up and the conditions that results were obtained. Results discussed in Chapters 5 and 6 confirmed that current conventional routing protocols and transmission over multi-hop networks significantly affected the performance of the perceived application QoS. This confirmed that single metric protocols, considering only the hop count, do not consider network changes and thus do not adapt well in strenuous conditions. The presence of this problem led to a significant performance degradation.

In MANETs, the nodes can be constantly on the move and the routing protocol needs to converge quickly to deal with link break. Additionally, if a valid shortest path is chosen, data will be transmitted over the selected route eventually creating network congestion even though another possibly longer route is available. These led to increased link failures as packet were buffered for longer period before being forwarded to the next shortest path. End-to-end delay, jitter ad packet loss quickly increased, degrading the overall QoS. Shortest path algorithms proved that they could not provide the required applications' QoS. Using multiple criteria selection metric to select a route has been introduced in previous research studies leading to an improvement in network performance. Previous studies have used methods such as sequential filtering, however studies that use a single metric that can mirror the QoS of the network are scarce.

To deal with the NP-problem linked to multi-constraint routing, fuzzy logic was introduced. The developed FIS assessment system proved beneficial in order to evaluate the current QoS provided to audio and video application. The QoS assessment was based on combining QoS parameters (delay, jitter and packet delivery fraction) using fuzzy inference system. The assess QoS gave a very valuable and overall indication of the network condition. It was also used to perform an analysis of the overall network QoS. This study demonstrated that AODV and DSR were incapable of providing the minimum QoS requirements for multimedia application transmission. The results also indicated that the QoS quickly degraded and exhibited a poor level as soon as the number of connections was greater than 1 or during medium and fast mobility.

In summary, AODV and DSR had limitations when transmitting various applications due to their mechanism properties. Furthermore, network evaluation confirmed that the application of fuzzy logic technique for assessing the perceived QoS was effective. Through the use of this approach, this study gave a good baseline that can contribute to knowledge concerning QoS routing in MANETs.

Future Work:

In this thesis, a detailed discussion on QoS routing in MANETs was presented. However, different networks have different mobilities, communication capacities, etc. Considering this work, our solution can be generally implemented to provide QoS routing alongside current routing protocol using a single selection metric. As wireless systems become more popular, they are deployed faster and demands on QoS is an important factor in service cost. The proposed solution can also be implemented as part of a call admission control by setting an acceptance threshold governed by the level of acceptable QoS. The assessed QoS level can also be used during resource reservation, according to the level of QoS measured bandwidth can be reserved for the next connection demanding more resources. Various network parameters may be adjusted, if the network is more aware of the QoS it is currently providing, and it is not only restricted to the Network layer but parameters at the Data or Transport layer can be adjusted to improve network performance. At the Data layer, MAC protocol parameters can be adjusted to regulate how information is fragmented and the Transport layer, TCP parameters can be adjusted to regulate the transmission data rate and reduce congestion and loss. In this research, static fuzzy membership functions were been used. Further research can be carried out to utilise some learning techniques such as Neural Network or Genetic algorithms to have these parameters reflecting the variables they are controlling.

It is important consider to these further research directions to enhance the possibilities of wireless ad-hoc networks.

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