

DEPARTMENT OF ELECTRICAL AND ELECTRONIC ENGINEERING

INTELLIGENT SYSTEMS AND NETWORKS

Resource Allocation in Ad Hoc Networks

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Abstract

Unlike the centralized network, the ad hoc network does not have any central administrations and energy is constrained, e.g. battery, so the resource allocation plays a very important role in efficiently managing the limited energy in ad hoc networks. This thesis focuses on the resource allocation in ad hoc networks and aims to develop novel techniques that will improve the network performance from different network layers, such as the physical layer, Medium Access Control (MAC) layer and network layer.

This thesis examines the energy utilization in High Speed Downlink Packet Access (HSDPA) systems at the physical layer. Two resource allocation techniques, known as channel adaptive HSDPA and two-group HSDPA, are developed to improve the performance of an ad hoc radio system through reducing the residual energy, which in turn, should improve the data rate in HSDPA systems. The channel adaptive HSDPA removes the constraint on the number of channels used for transmissions. The two-group allocation minimizes the residual energy in HSDPA systems and therefore enhances the physical data rates in transmissions due to adaptive modulations. These proposed approaches provide better data rate than rates achieved with the current HSDPA type of algorithm.

By considering both physical transmission power and data rates for defining the cost function of the routing scheme, an energy-aware routing scheme is proposed in order to find the routing path with the least energy consumption. By focusing on the routing paths with low energy consumption, computational complexity is significantly reduced. The data rate enhancement achieved by two-group resource allocation further reduces the required amount of energy per bit for each path. With a novel load balancing technique, the information bits can be allocated to each path in such that a way the overall amount of energy consumed is minimized.

After loading bits to multiple routing paths, an end-to-end delay minimization solution along a routing path is developed through studying MAC distributed coordination function (DCF) service time. Furthermore, the overhead effect and the related throughput reduction are studied. In order to enhance the network throughput at the MAC layer, two MAC DCF-based adaptive payload allocation approaches are developed through introducing Lagrange optimization and studying equal data transmission period.

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- M. K. Gurcan, H. Ab Ghanai, J. Zhou and A. Chungtragarn, "Bit Energy Consumption Minimization for Multi-path Routing in Ad-hoc Networks", *the Computer Journal*, Oxford University Press, January 2011
- J. Zhou, M. K. Gurcan, and A. Chungtragarn, "Energy-aware Routing with Two-group Allocation in Ad Hoc Networks", *25th International Symposium on Computer and Information Sciences*, The Royal Society, London, United Kingdom, 22-24 September 2010
- J. Zhou and M. K. Gurcan, "An Improved Multicode CDMA Transmission Method for Ad Hoc Networks", *IEEE Wireless Communications & Networking Conference 2009 (WCNC2009)*, Budapest, Hungary, 5-8 April 2009

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List of Symbols

P_t	transmission power
P_r	received power
G_t	transmitter antenna gain
G_r	receiver antenna gain
λ	radio wavelength
d	distance between the transmitter and the receiver
L_s	system loss
h_t	height of transmitter
h_r	height of receiver
d_0	close-in reference distance
δ	path loss exponent
C	capacity
Γ	system gap
b	bit rate per symbol
C_T	total capacity
C_k	capacity achieved at kth channel
k	index of channel
K	the maximum number of channel
P_T	total transmission power
h_k	channel gain in the kth radio channel
p_k	transmission power at the kth transmitter
ω_k	receiver decorrelator coefficient in the kth channel
s_k	spreading codes used at the kth transmitter
ω_k^T	inverse of receiver decorrelator coefficient in the kth channel
σ	noise variance
T_s	symbol period
T_c	chip period

$E(b)$	energy required to b bits per symbol
e_{drece}	decremental energy
e_{inece}	incremental energy
I_N	$N \times N$ diagonal matrix
c_k	MMSE coefficient at the kth receiver
$\bar{S}$	vector of updated spreading codes with the MMSE coefficient
$\mathbf{R}$	correlation matrix of spreading codes
$\bar{s}$	vector of a significant sequence
β	granularity
b_k	bit rate per symbol achieved in the kth channel
c_k^I	real part of complex spreading code
c_k^Q	imaginary part of complex spreading code
b_k^I	real part of bit after complex modulation
b_k^I	imaginary part of bit after complex modulation
n_k^I	real part of noise
n_k^Q	imaginary part of noise
$Re(\cdot)$	real part of complex number
C_i^N	Hadamard matrices
dt_k	length of frame
r_{coded}	coded rate
A_k	amplitude at the kth transmitter
∂	interference factor
$\hat{b}_k$	estimated bit rate at the kth receiver
$\hat{dt}_k$	estimated length of frame at the kth receiver
f	carrier frequency
k	Boltzmann constant
T_n	noise temperature
T_e	error temperature
T_0	background temperature

p_{data}	probability of successful data packet transmission
p_{ack}	probability of successful ACK packet transmission
L	data packet size
r_{data}	data rate
θ	tunable parameter
J	routing path J
$r_{\max}$	maximum data rate
$I(k)$	total sum Transmission Contention Degree of link k
r_{data}^k	data rate received at link k
$ T_i $	number of nodes in the subtree with root i
S	throughput
$T_{i,j}^m$	medium time from node i to node j
η	increasing contention
T_i^m	the medium time of the i_{th} hop link
H	maximum number of hops in a routing path
R	total data rate
M	total amount of bits required to be transmitted for routing paths
Z	total number of routing paths
$R_{MAC,z,MAX}$	maximum MAC rate along routing path z
T	observe time
$\bar{T}$	average service time
P_{MAC}	MAC power
N_{SG}	spreading gain
m_z	bits allocated at routing path z
E_z	energy consumed at routing path z
$E_{b,z}$	energy consumed per bit for routing path z
A	adjacency matrix
$\bar{T}_B$	backoff time
T_s	successful transmission time
N	number of nodes or users in a network

t_{RTS}	time slots for RTS packet
t_{CTS}	time slots for CTS packet
t_{ACK}	time slots for ACK packet
t_{SIFS}	time interval between packet transmissions
t_{DIFS}	DCF Inter Frame Space interval
δ	transmission delay
t_c	collision time slot
$W_{\min}$	minimum window size
$W_{\max}$	maximum window size
q	successful transmission probability
τ	transmission probability
α	average time cost per event
p_i	probability of channel is idle state
p_s	probability of channel is in successful transmission state
p_c	probability of channel is in collision state
p	collision probability
p_{tr}	the probability that channel is busy
R_i	the nodes within node i's carrier sensing range
I	identity matrix
$\underline{q}$	vector of successful transmission probability
$\underline{\pi}$	vector of handshake successful probability
Φ	interference matrix
p_f	probability of frame successful reception
π	handshake successful probability
φ	Lagrange multiplier
M_T	total amount of bits required to be transmitted in a network
T_{data}	data packet transmission period
T_{EtE}^J	end-to-end delay for routing path J
$L_{J,0}$	optimal payload size
N_0	noise power spectral density

p	modulation scheme level
P	the maximum modulation scheme level achieved in systems
e_R	residual energy
B_{CA}	total bit rate achieved in channel adaptive HSDPA systems
n	number of channel used for transmission in HSDPA systems
SNR_T	total SNR measured at the input of receiver
SNR_{CA}	SNR measured at the input of each subreceiver
e_R^{CA}	residual energy in channel adaptive HSDPA systems
B_{TG}	total bit rate achieved in two-group HSDPA systems
m	a group of channel with high bit rate in two-group HSDPA systems
e_R^{TG}	residual energy in two-group HSDPA systems

Abbreviations and Acronyms

ACK	Acknowledgment
AIFS	ACs Interframe Space
AODV	Ad hoc On-Demand Distance Vector
AWGN	Additive White Gaussian Noise
BER	Bit Error Rate
BPSK	Binary Phase Shift Keying
CDMA	Code Division Multiple Access
CTS	Clear-to-Send
CW	Contention Window
DCF	Distributed Coordination Function
EDCF	Enhanced Distributed Coordination Function
HCF	Hybrid Coordination Function
HSDPA	High Speed Downlink Packet Access
IWF	Iterative Water Filling
MAC	Medium Access Control
MAI	Multiple Access Interference
MIMO	Multiple-Input Multiple-Output
MMSE	Minimum Mean Square Error
NAV	Network Allocation Vector
OFDM	Orthogonal Frequency Division Multiplexing
PAM	Pulse Amplitude Modulation
PSK	Phase Shift Keying
QAM	Quadrature Amplitude Modulation
QPSK	Quadrature Phase Shift Keying
QoS	Quality-of-Service
RTS	Request-to-Send
SIFS	Short Inter-Frame Space
SNIR	Signal-to-Noise-and-Interference Ratio
TXOP	Transmission Opportunity

ISO	International Organization for Standardization
OSI	Open Systems Interconnection
BW	bandwidth
FDMA	Frequency-Division Multiple Access
TDMA	Time-Division Multiple Access
FEC	Forward Error Correction
3G	Third Generation
GSM	Global System for Mobile Communications
IEEE	Institute of Electrical and Electronics Engineers
IT	Information Technology
Mb	Megabit
bps	bit per second
MHz	Megahertz
UMTS	Universal Mobile Telecommunication Systems
WLAN	Wireless Local Area Network
3GPP	3rd Generation Partnership Project
BLAST	Bell Labs Layered Space Time
BS	Base Station
CSI	Channel State Information
CSMA	Carrier Sensing Multiple Access
CA	Collision Avoidance
DBPSK	Differential Binary Phase Shift Keying
DQPSK	Differential Quadrature Phase Shift Keying
DS/CDMA	Direct Sequence Code Division Multiple Access
EDGE	Enhanced Data Rates for GSM Evolution
IC	Interference Cancellation
OVSF	Orthogonal Variable Spreading Factor
P/S	Parallel-to-Serial
PHY	Physical Layer

WCDMA	Wideband Code Division Multiple Access
TD-SCDMA	Time Division-Synchronous Code Division Multiple Access
AMC	Adaptive Modulation and Coding
DIFS	Distributed Interframe Space
DLL	Data Link Layer
DSSS	Direct Sequence Spread Spectrum
FER	Frame Error Ratio
FDD	Frequency Division Duplex
TDD	Time Division Duplex
LTE	Long Term Evolution
DVB	Digital Video Broadcast
DAB	Digital Audio Broadcast
SC-FDMA	Single Carrier-Frequency Division Multiple Access
OFDMA	Orthogonal Frequency Division Multiple Access
WBE	Welch Bound Equality
AODVM	Ad hoc On-Demand Distance Vector Multipath
AOMDV	Ad hoc On-Demand Multipath Distance Vector
EASR	Energy-aware Single-path Routing
EADMR	Energy-aware Disjoint Multi-path Routing

Chapter 1

Introduction

1.1 Ad Hoc Networks

An ad hoc network is a local area network (LAN) where the individual node forwards the packet from one node to another. The network does not relay data packet from/to a centralized station to control and coordinate the communication to each node. The ad hoc network is self-organized by a group of devices, such as bluetooth device, laptop or mobile phone (Figure 1.1).

In the 1970s, the first generation of ad hoc networks was developed, and it was known as Packet Radio Networks (PRNET). This was a form of packet switching based network to transmit data packets over wireless communication links. The development of ad hoc networks was further improved in the 1980s and as such, ad hoc networks were implemented as a part of the Survivable Adaptive Radio Networks (SUARN), which was applied in the mobile battlefield without the requirement of

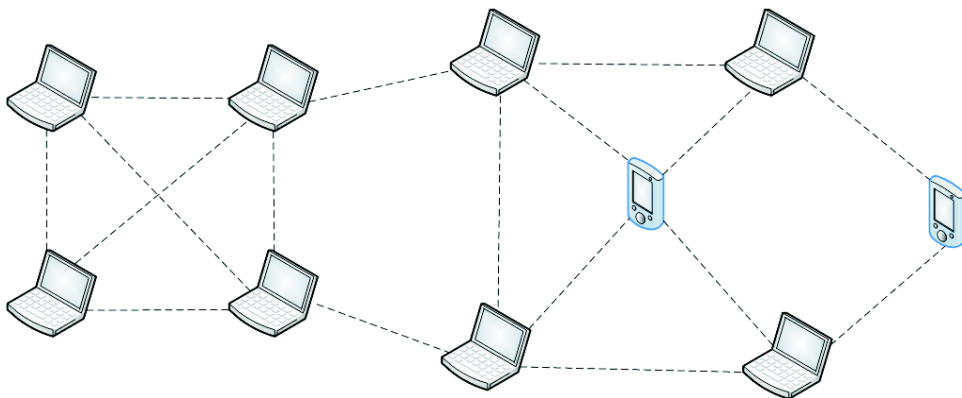


Figure 1.1: A Wireless Ad Hoc Network

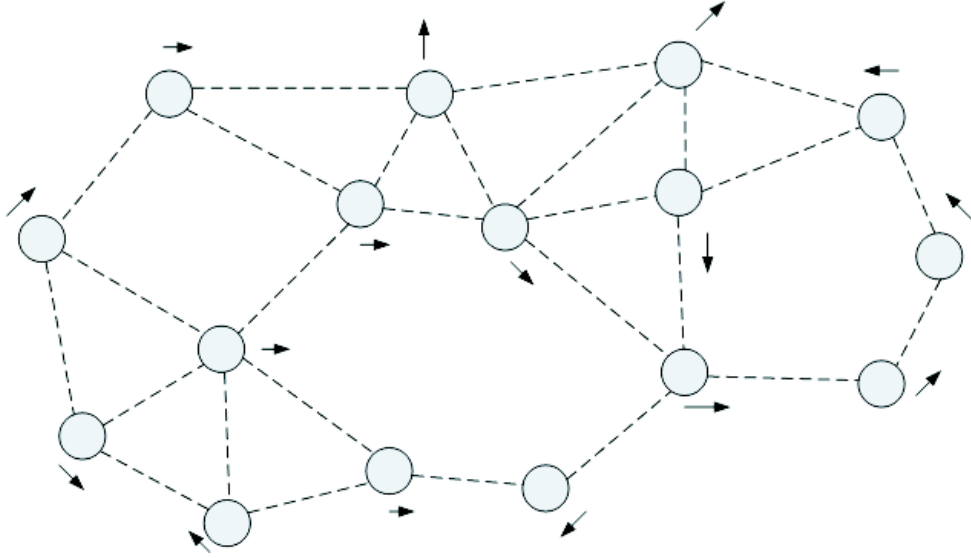


Figure 1.2: A Mobile Ad Hoc Network

infrastructure. During the 1990s, the ad hoc network concept was developed for commercial application. In 1994, Bluetooth was created by Ericsson as an open wireless technology standard used for short range communications from fixed or mobile devices. Since the launch of the first version of IEEE 802.11 standard [1] in 1997, WiFi has been developed over a decade with many different Wireless LAN (WLAN) commercial products.

Based on the application, wireless ad hoc networks can be further classified into three main networks: mobile ad hoc networks (MANETs), wireless mesh networks (WMN) and wireless sensor networks (WSN). Unlike static networks, nodes in MANETs move independently in any direction (Figure 1.2), therefore the communication link and related network topology change frequently. According to this, due to the mobility, the study on wireless link quality, fading, and dynamic topological change become the challenge for MANETs. With regard to the application of MANETs, the study on efficiently distributed algorithms, link scheduling and routing is always a hot topic.

A wireless mesh network is organized in a mesh topology, and it normally includes gateways, mesh clients and mesh routers. In a mesh topology, individual nodes are connected to a group of nodes and communicate with nodes in other groups through the gateway, which plays an administration role in ad hoc networks (as illustrated in figure 1.3). A mesh cloud is known as the coverage of a group of nodes in a wireless mesh network.

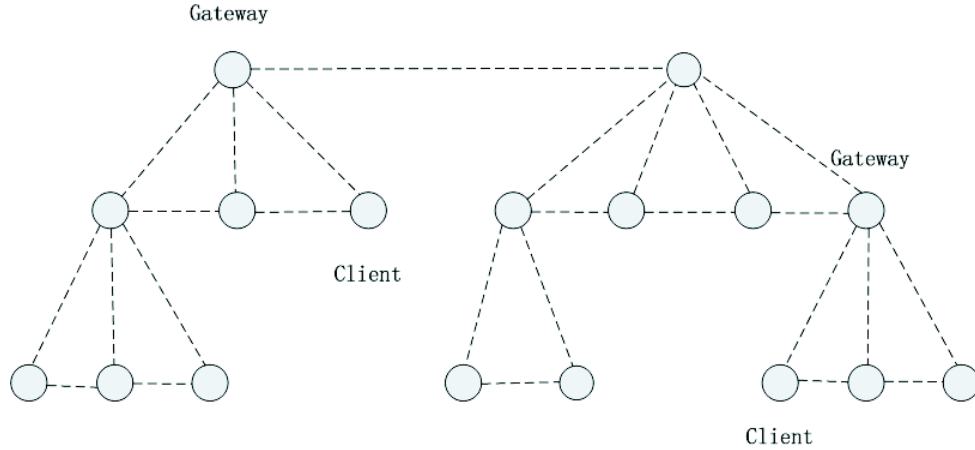


Figure 1.3: A Wireless Mesh Network

In a wireless ad hoc sensor network, sensors are distributed across a geographical area to collect information and also to monitor physical conditions, such as temperature, pressure or sound. Wireless sensor networks need to offer certain signal processing abilities and communication capabilities. As the physical size of a sensor is small, the battery normally becomes the power supply in wireless sensor networks. In [2], an accurate experimental measurements of the power consumption is done for both CPU state change and communications (e.g. data transfer and receive) in wireless sensor networks. The result identifies that the power consumption of CPU state change is from 0.145 Watts to 0.221 Watts, and the power consumption of communication is from 0.228 Watts to 0.261 Watts in a wireless sensor networks under given configuration. On this basis, the energy is constrained in wireless sensor networks, so it is of no surprise that there is much lively debate and discussion around how to utilize the limited energy efficiently and extend the network lifetime for wireless sensor networks.

1.2 Open System Interconnection (OSI) Seven-layer Network Model

In 1978, an Open System Interconnection (OSI) model was developed by the International Organization for Standardization (ISO) to divide a computer and communication network into small sections called layers. Seven layers are defined in the OSI network model (Figure 1.4), which are normally divided into two areas: the

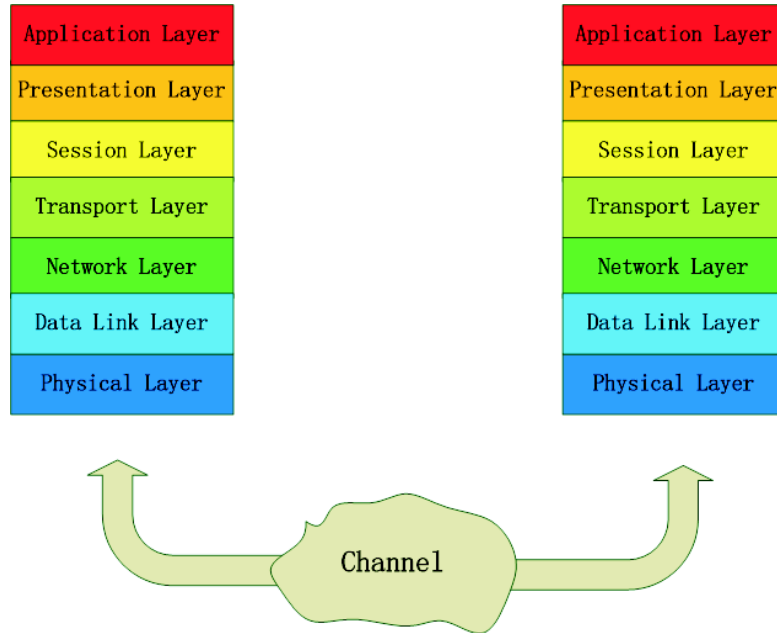


Figure 1.4: OSI Seven-layer Model

lower layer and the upper layer. The lower layer includes a physical layer, data link layer, network layer and transport layer. The upper layer consists of a session layer, presentation layer and application layer.

1.2.1 Lower Layers

The Physical Layer

The physical layer is the lowest and the first layer in the OSI model, and it consists of the hardware and fundamental specifications of computing networks, such as data rates, transmission distance, and antennas. The information bits are grouped into symbols and converted into physical signals over a physical link between nodes in a network. A physical signalling sub-layer provides an interface between the data link layer and the medium channel. Many functions and services are provided by the physical layer, such as the establishment and termination of a communication link between nodes, and the signal processing of raw digital bits.

Regarding the communication systems, the major parameters provided at the physical layer are the data rate and the bit error ratio (BER). Like the roots of a tree, the data rate achieved at the physical layer provides the upper bound of the throughput for a computing network. Based on the application, the physical layer is categorized as Local Area Network (LAN) and Metropolitan Area Network (WAM)

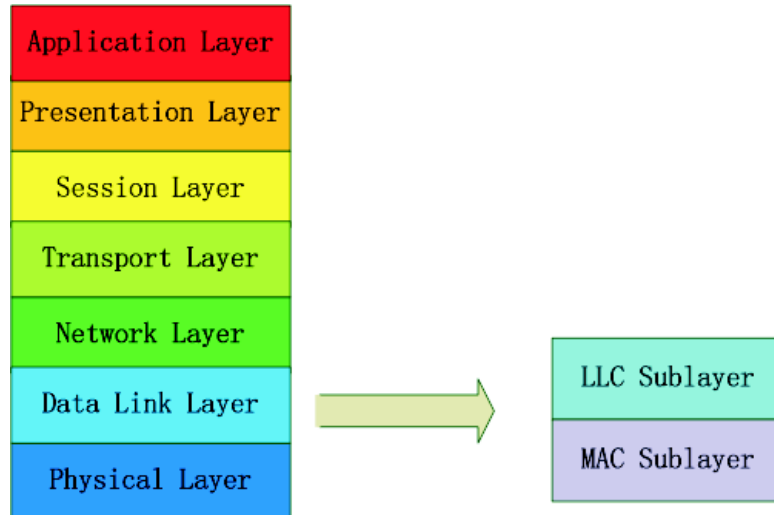


Figure 1.5: LLC Sublayer and MAC sublayer

specifications.

The Data Link Layer

The data link layer provides the functional to transfer data across network and detects (or corrects) errors from the physical layer. The data link layer includes two sub layers: the Media Access Control (MAC) and the Logical Link Control (LLC) layer (figure 1.5). The MAC sublayer controls the permission of access medium channel for a node (or user) to transmit packets in a network, and this makes nodes share the same physical medium.

The MAC protocols are responsible for detecting or avoiding the collision of data packets, hence they reduce the probability of retransmissions. Two main protocols are defined at the MAC layer: the distributed MAC protocol and the central MAC protocol. Some other services are also provided at the MAC sub-layer in computing networks, such as MAC addressing, packet switching, scheduling, store-and-forward switch, and Virtual LANs (VLANs).

The LLC is the upper sublayer of the Data Link layer, and it provides services, such as protocol multiplexing, flow control, acknowledgment, and error control (Automatic Repeat Request (ARQ)). The LLC sub-layer specifies mechanisms and protocols, which are used for controlling the data that is to be exchanged between nodes as well as addressing the nodes over the same network medium.

The Network Layer

Switching and routing technologies are provided at the network layer to create logical paths to forward data from one node to another. Some functions of network

layer include congestion control, addressing, internetworking, and packet sequencing. In order to ease the management of the network and to better control the flow of packets, network layers are divided into smaller parts known as subnets. Routers are specifically configured and route traffic between different networks through network portion of the IP addressing. The communication is established between routers by using routing protocols (i.e. ad hoc On-Demand Distance Vector Routing (AODV) [3]) to discover the best route from the source to the destination based on a variety of criteria, such as the shortest path.

The Transport Layer

The transport layer transfers data between end users and provides a data transfer service to the upper layers. It is also responsible for recovering end-to-end error. Two typical protocols are used at the transport layer: Transmission Control Protocol (TCP) and User Datagram Protocol (UDP).

TCP is a connection-oriented protocol which requires handshaking to establish end-to-end communication links and it is responsible for exchanging data directly between two nodes in a network. The reliable data transfer service is provided by TCP through managing message acknowledgment, retransmission and timeout.

Unlike TCP, UDP is a simple message-based connectionless protocol, and it transmits information in one direction without verifying the state of receiver. Due to the fact that UDP does not provide reliable communication, it therefore is used for simple messaging transmissions. However, as retransmissions are not possible in large networks, UDP is used for multicasting and broadcasting to achieve higher throughput and lower latency, and for this reason it is often used for real-time communication where low packet loss can be accepted.

1.2.2 Upper Layers

The Session Layer

The session layer is responsible for establishing, terminating and managing a connection (or dialogue) between end-user application processes, such as communication session requests and responses between computers. The main services provided at the session layer are full-duplex, half-duplex, authentication, permissions and session restoration (otherwise known as check pointing and recovery).

The Presentation Layer

The presentation layer is responsible for providing services to the application layer for further data processing. Encryption and compression are two main ser-

vices provided at the presentation layer to deliver and format information to the application layer. The key functionality of the presentation layer is the serialization which serializes complex data structures into flat byte-strings.

There are two sublayers in the presentation layer: Common Application Service Element (CASE) and Specific Application Service Element (SASE). The CASE sub-layer provides common application services and acts as an interface between the session and the application layers. For example, it obtains the services from the session layer and forwards the services to the application layer. Unlike the CASE sub-layer, the SASE sub-layer provides support for specific application services.

The Application Layer

The application layer is one of the upper layers and is closest to the end user. It can be considered as one of the high-level services for the application program or an interactive user. The application layer provides conversion between users and application processes through interacting with the software applications. The function of the application layer includes: determining and identifying the availability of communication partners for the data transmission; deciding the network resource availability for the requested communication applications; managing the communication synchronization between applications. Some examples of the protocol existing at the application layer include the File Transfer Protocol (FTP) and the Simple Mail Transfer Protocol (SMTP).

1.3 Research Motivation

With the demand of multi-media communication, signal processing technology aims to provide high data rates in order to satisfy Quality of Service (QoS) requirements. This results in multi-rate networks and related problems. This section addresses several problems regarding multi-rate wireless ad hoc networks from different layers in the OSI model.

1.3.1 Energy Utilization and Data Rate at the Physical Layer

The data rate achieved at the physical layer provides the upper bounds of throughput in ad hoc networks. Accordingly, it is necessary to increase the data rate and hence improve the network performance. The wireless medium channel results in the signal power reduction (or path loss) due to attenuation or multipath during the transmission. In addition, the noise generated at the input of receiver also interrupts

the received signal, thereby causing errors on the recovered signal. In 1948, the theoretical system capacity with error free was defined by Shannon [4] as follows:

$$C = \frac{1}{2} \log_2(1 + SNR) \quad (1.1)$$

$$C = \frac{1}{2} \log_2\left(1 + \frac{P_T h}{P_n}\right) \quad (1.2)$$

Where C is the theoretical system capacity with error free, P_T is the transmission power allocated at the transmitter, h is the path loss (which has negative value in dB) due to propagation, and P_n is the noise power. Based on the equation (1.2), both path loss, h , and noise, P_n , result in system capacity reduction and hence low data rate for transmissions.

However, the theoretical system capacity cannot be implemented in practical wireless communication systems as it is continuous. The data rate used for practical transmissions is discrete and is achieved by using modulation schemes and coding schemes. As the data rate is not continuous, if the energy is not appropriately allocated, it will cause inefficient energy utilization and result in residual energy in wireless communication systems. On this basis, the resource allocation technique is developed and aims to efficiently utilize the energy and improve the data rate for transmissions. Based on the channel condition, different data rates are achieved at transmission links and this results in multi-rate ad hoc networks.

1.3.2 Effective Throughput

In an ad hoc network, the maximum throughput is achieved when only one node activates and there is no other node to contend the channel. However, even if the maximum throughput is achievable due to no contention in a network, the effective throughput is still limited by non-data information (or overhead). In order to achieve the functions of each layer in the OSI model for transmissions, headers are added into frame and extra control packets are also applied. Given the upper bound of throughput in ad hoc networks, both headers and control packets result in inefficient transmissions and therefore cause the throughput reduction. For example, at the MAC layer, the length of MAC header and control packets is up to 154bytes in IEEE 802.11 standard. These bits are used to implement the function at the MAC layer but result in throughput reduction. In an ad hoc network, the throughput is simply defined as follows:

$$S = \frac{L}{\overline{T}} \quad (1.3)$$

Where L is the payload (or data packet) size during one successful transmission, and $\bar{T}$ is the average service time for one successful transmission. Once the payload size is fixed for transmission, only service time, $\bar{T}$, will affect the network throughput. The service time includes: the propagation delay, the time interval between packets, and the transmission time slots for data packets, headers, and control packets. According to this, the proportion of the payload transmission time slot in relation to other time slots in service time determines how effective the network throughput is. As such, it becomes clear that how to use the resource allocation to overcome the effect of overhead on the effective throughput in multi-rate ad hoc networks becomes an interesting topic.

1.3.3 Energy Consumption of Routing Transmissions

Unlike the centralized network, the ad hoc network does not have any central administration. The communication in ad hoc networks is to forward information from one node to another based on a specific routing, so the relay node uses significant additional energy. However, the energy is always constrained in an ad hoc network. When the energy runs out at a node, this node will “die” and the related links will be disconnected, and hence the network topology will change.

Currently, the most popular routing scheme is Ad Hoc On-demand Distance Vector (AODV) [3], which provides the shortest path discovery solution. However, this routing scheme is mainly based on equal data rate networks, and not multi-rate networks. In a multi-rate ad hoc network, the shortest routing path may not provide good performance from an energy utilization perspective. Although the shortest path provides the minimum number of hops, the channel condition could be extremely poor at some links and could result in low network throughput or more energy consumption. On this basis, by considering different aspects at layers, such as the channel condition and related achievable data rates, it is necessary to really consider the energy-aware routing for multi-rate ad hoc networks in order to manage the limited energy and extend the network lifetime.

1.3.4 End-to-End Delay along a Routing Path

Given a routing path for transmissions, the end-to-end delay equate to the total time consumed between the departure of a data packet at the source node and the successful reception of a data packet at the destination node. However, the

traditional routing scheme only takes account of the propagation delay as the time consumed for end-to-end delay calculation. At the MAC layer, the service time also results in delay for transmissions due to contention between nodes. Furthermore, in a full-load ad hoc network, the transmission of data packets is continuous, which means that next data packet does not need to wait the successful reception of a previous data packet at the destination, and it is able to be transmitted if the channel is available. Regarding a full-load routing path, given amount of information bits at the source node, the management of data packet transmissions becomes a challenging problem in term of controlling the end-to-end delay along a routing path.

1.4 Research Contribution

- The energy utilization problem is analyzed and addressed in existing High-Speed Downlink Packet Access (HSDPA) [5] systems, and the related data rate allocation and residual energy are studied and examined. (Chapter 3). Based on the result, it is observed that at certain bit rates that are up to a 6 dB increase in the total SNR do not provide any increase for the bit rate due to the residual energy in original HSDPA systems.
- A channel adaptive approach is developed through easing the constraints on the number of channels used for transmission in HSDPA systems. It improves the data rate for transmissions through reducing the residual energy in HSDPA systems, especially in low SNR range (i.e. 14dB–26dB) (Chapter 3).
- A two-group allocation approach is developed in HSDPA systems through allocating the bits into two groups of channels with two different modulation schemes. By significantly decreasing the upper bound of residual energy in HSDPA system from incremental energy in K parallel channels to only one channel, the data rate is enhanced for transmissions in both low and high SNR ranges (i.e. 14dB–39dB) (Chapter 3).
- An energy-aware routing scheme is developed in order to find the routing paths which provide low energy consumption. Given the transmission power and data rate at the physical layer, the energy consumed per bit is defined as the cost function for selecting the low energy routings. A trellis-hop diagram is developed to provide an easy way to discover all possible routing paths for a given node pair in a network topology. In order to reduce the computational

complexity of a trellis-hop diagram approach when the network size tends to be large, a modified Viterbi approach is proposed. On this basis, an energy-aware single-path routing (EASR) scheme and an energy-aware disjoint multipath routing (EADMR) scheme are developed (Chapter 4).

- In order to further reduce the energy consumption, a load balancing approach is developed through efficiently allocating the bit to discovered multipath routing paths (Chapter 4). However, the performance of load balancing approach is dependent on the cost difference between multiple routing paths. If the cost difference between multiple routing paths is large, the proposed load balancing approach will show significant reduction on energy consumption.
- The throughput enhancement achieved by the two-group allocation in HSDPA systems is examined in ad hoc networks by studying the MAC Distributed Coordination Function (DCF) [1] service time. (Chapter 5). The result identifies that two-group allocation improves the overall network throughput, especially when the total payload required to be transmitted is large (i.e. more than 10Mbits in a 50-node network).
- After studying the overhead effect at the MAC layer in ad hoc networks, given the total amount of information bits required to be transmitted in a given network, two MAC DCF based adaptive payload allocation approaches are developed to improve the overall network throughput, especially when the total payload required to be transmitted is small (i.e. less than 10Mbits in a 50-node network). An optimization solution is proposed by introducing Lagrange multiplier, and this approach significantly enhances the overall network throughput. Another equal payload transmission period approach not only achieves the same performance of Lagrange optimization approach, but also provides a simple solution to overcome the overhead effect at the MAC layer, and hence improve the network throughput (Chapter 5).
- After allocating the information bits to multiple routing paths based on the load balancing approach developed before, the end-to-end delay of a routing path is examined by considering the MAC DCF service time. Through managing the payload size for transmissions, a mathematical solution is developed to find the optimal payload size and corresponding minimum end-to-end delay. (Chapter 5)

1.5 Thesis Outline

The rest of this thesis is organized as follows.

As this thesis focuses on the resource allocation studies at the physical layer, the MAC layer and the network layer, chapter 2 gives the related background and the system model. The IEEE 802.11 is introduced and the development of different released versions is presented. Regarding the physical layer, the wireless communication propagation and multiple channel access schemes are presented in details. Next, three main protocols used at the MAC layer in IEEE 802.11: Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA), distributed MAC protocol and central MAC protocol are introduced. Section 2.4 discusses the routing classification in Ad Hoc Networks. Next, the development of cross-layer design is explored. Finally, the system model used in this thesis is given.

Chapter 3 studies a physical-layer technology called multi-code Code Division Multiple Access (CDMA) [6] [7] and related resource allocation techniques. Firstly, the capacity in multi-code CDMA is presented and discussed. The main objective here is to improve the capacity of multi-code CDMA using two main resource allocation techniques: power allocation and optimal spreading codes design. These techniques are studied and the related work is reviewed. Secondly, the HSDPA, which is an application of multi-code CDMA, is presented and modelled. The energy utilization problem is then analyzed and addressed in HSDPA systems, and related residual energy and data rate achieved in HSDPA systems are studied and examined. In order to solve this inefficient energy utilization problem in HSDPA systems, two resource allocation techniques are developed: a channel adaptive approach eases the constraints on the number of channels used for transmissions in HSDPA systems, and a two-group allocation approach [9] distributes the bit into two groups of channels with different modulation schemes.

In chapter 4, the energy-aware routing scheme is studied at the network layer in ad hoc networks. Firstly, the related work focused on energy-aware routing, MAC-aware routing, multipath routing and load balancing are reviewed. Secondly, the energy utilization problem for routing scheme in multi-rate networks is addressed and formulated through considering the physical data rate as the core part of the cost function. A trellis-hop diagram approach is developed to discover all possible routing paths for a given node pair in a network. In order to reduce the computational complexity of trellis diagram approach, a modified Viterbi solution [10] [11] is proposed. On this basis, an energy-aware single-path routing (EASR) scheme

and an energy-aware disjoint multipath routing (EADMR) scheme are developed to achieve low energy consumption in ad hoc networks. Furthermore, a load balancing approach is developed to further reduce the energy consumption for multipath routing schemes.

Chapter 5 focuses on the IEEE 802.11 MAC DCF protocol. Firstly, the related work on the MAC DCF study is reviewed. Secondly, the MAC DCF service time and related network throughput are studied through analyzing a DCF model developed in [12] for multi-hop ad hoc networks. In addition, the throughput enhancement achieved by two-group allocation in HSDPA systems at the physical layer, is examined in ad hoc networks. With regard to the issue at the MAC layer, the overhead effect on throughput reduction is addressed. Two adaptive payload allocation approaches, Lagrange optimization and equal payload transmission period, are developed to overcome the overhead effect in multi-rate networks and therefore improve the overall network throughput. After allocating the information bits to each routing path based on the load balancing approach developed in chapter 4, the end-to-end delay along a full-load routing path is studied and modelled. An end-to-end delay minimization solution is developed through finding the optimal payload size and managing the payload transmission.

Finally, the summary of this thesis is given, and some open questions regarding future work are addressed in chapter 6.

Chapter 2

Background and System Model

2.1 IEEE 802.11

IEEE 802.11 is the first Wireless Local Area network (WLAN) standard developed by IEEE LAN/MAN Standards Committee in 1997 [1]. It defines the media access control (MAC) layer and the physical (PHY) layer for a LAN with wireless connectivity [1], and works at 2.4GHz which is a public frequency spectrum for the Industrial, Scientific and Medical (ISM) band. Due to the fact that 2.4GHz band is free, however, 802.11 devices suffer interference from other devices (e.g. microwave ovens and Bluetooth products) operating in the same band. At the physical layer, IEEE 802.11 applies the Direct Sequence Spread Spectrum (DSSS) and the Frequency Hopping Spread Spectrum (FHSS) and provides both 1Mbit/s and 2Mbit/s data rate by using Differential Binary Phase Shift Keying (DBPSK) and Differential Quadrature Phase Shift Keying (DQPSK) modulation schemes. At the MAC layer, both the Distributed Coordination Function (DCF) and the Point Coordination Function (PCF) protocols provide distributed and central channel access services, respectively. Since the first IEEE 802.11 standard was applied, more advanced standards have joined the IEEE 802.11 family, such as the 802.11a, 802.11b, 802.11g, 802.11e and 802.11n. In the following sections, the development of IEEE 802.11 is presented.

- **IEEE 802.11a**

The 802.11a was released in 1999 and uses the same MAC layer protocol and frame format as the 802.11 standard. The difference between the original standard and 802.11a is that the 802.11a uses the 48-subcarrier Orthogonal Frequency Division Multiplexing (OFDM) at the physical layer to provide a maximum data rate of

<i>IEEE 802.11 Family</i>					
Versions	802.11	802.11a	802.11b	802.11g	802.11n
release	1997	1999	1999	2003	2009
data rate per stream (Mbits)	1,2	6,9,12,18 24,36,48,54	5.5,11	6,9,12,18 24,36,48,54	15,30,45,60 90,120,135,150
modulation	BPSK QPSK	BPSK QPSK 16QAM 64QAM	CKK	BPSK QPSK 16QAM 64QAM	BPSK QPSK 16QAM 64QAM
channel access	DSSS FHSS	OFDM	DSSS	OFDM	OFDM
bandwidth (MHz)	20	20	20	20	40
frequency (GHz)	2.4	5	2.4	2.4	2.4

Table 2.1: IEEE 802.11 Family

54Mbit/s. In addition, 802.11a operates in the 5GHz band which is a high carrier frequency.

However, 802.11a signals are more absorbable by walls and other solid objects during signal transmissions due to higher carrier frequency (or smaller wavelength) compared with that of the 802.11 signals. This results in shorter effective transmission range of 802.11a than that of 802.11b/g. Nevertheless, it is worth considering the advantage of the OFDM when the signals are propagated in a high multipath environment, this enables the 802.11a to suffer less interference. Based on the strength of received signals, each subcarrier is operated by different modulation schemes (e.g. BPSK, QPSK, 16QAM and 64QAM) to achieve the available data rates (9Mbit/s, 18Mbit/s, 36Mbit/s and 54Mbit/s).

• IEEE 802.11b

Unlike the 802.11a, the 802.11b was released in 1999 as an extension of the original 802.11 standard which uses DSSS. In order to provide high data rate transmission, the 802.11b uses Complementary Code Keying (CCK) as its modulation techniques to achieve a maximum data rate up to 11Mbit/s. CCK has unique mathematical properties of shorter chipping sequence to obtain higher data rate. The CCK modulation scheme transmits data in symbols of 4 bits per carrier at a 5.5Mbit/s rate and 8 bits per carrier at a 11Mbit/s rate. If the signal quality becomes weak, 802.11b will scale back at the 5.5Mbit/s and original 802.11 data rate (2Mbit/s and 1Mbit/s) through adaptive rate selection. Regarding the transmission range, 802.11b has a typical indoor range up to 30 meters at 11Mbit/s and 90 meters

at 1Mbit/s.

- **IEEE 802.11g**

The 802.11g was released in 2003 as the third 802.11 standard. It uses the OFDM like the 802.11a, but works in the 2.4GHz band just like the 802.11b. The 802.11g provides a maximum data rate up to 54Mbit/s, which is same as the 802.11a. However, due to the fact that the carrier wavelength of the 802.11g is longer than the carrier wavelength of the 802.11a, its signal will not suffer large absorption from the solid objects in the transmission path, and hence it provides a longer transmission range.

- **IEEE 802.11n**

The 802.11n is considered as the next generation of the 802.11 standard, which should provide significant improvement at both the physical and MAC layers. In particular, one area is worth highlighting with regard to the 802.11n is the application of Multiple Input and Multiple Output (MIMO) technology.

MIMO is a technology which applies the multiple antenna arrays at both transmitter and receiver. One application of the MIMO system is to increase the data rate through spatial multiplexing, e.g. the V-BLAST system. In order to enhance the quality of signals at the receiver and therefore extend the transmission coverage, the spatial diversity is exploited by using a space-time block coding technique known as Alamouti coding. In 802.11n, the MIMO system is allowed up to 4 by 4 antenna array. The configuration of 802.11n device normally supports 2 by 2, 2 by 3, or 3 by 2 antenna array. All of these achieve the same maximum throughput but different diversity property, because the capacity of MIMO system is limited by the minimum number of antenna at either transmitter or receiver. In addition, another configuration 3 by 3 MIMO is becoming popular as high maximum throughput performance due to extra data stream.

Another feature incorporated into the 802.11n is the 40MHz bandwidth. This feature increases the usual 20MHz bandwidth used in previous 802.11 standards by double. The 802.11n also increases the number of OFDM subcarriers from 48 to 52 in each 20MHz bandwidth, to achieve a total of 104 OFDM subcarriers. The maximum achievable data rate is up to 600Mbit/s when 4 by 4 antennas MIMO systems and the 64QAM modulation scheme are applied.

However, the significant data rate enhancement at the physical layer does not improve the throughput in the same way due to the overhead, such as the PHY/MAC

header, contention time and the MAC control packet interval. In order to solve this problem, frame aggregation is introduced in the 802.11n. The frame aggregation aims to reduce the number of access attempts to the medium channel and overhead through aggregating multiple frames into a single transmission, and hence significantly increase the throughput. However, this mechanism is not proposed to build a huge frame, but a reasonable frame size, as huge frames may result in negative effect on fairness and efficiency in networks.

2.2 Wireless Communications

2.2.1 Wireless Communication Propagation

Unlike wired channels, wireless channels are more unpredictable due to complex channel states, e.g. rain loss and barrier attenuation. Basic propagation mechanisms contain reflection, attenuation, diffraction and scattering. Thanks to the propagation medium and obstacles, the attenuation degrades the power strength of received signals. In order to study and estimate the channel condition, three main wireless propagation models are developed for a large-scale radio channel in wireless communication systems: free-space propagation model, two-ray ground propagation model and log-distance path loss model [8]. In the following sections, these three basic propagation models will be introduced and presented.

2.2.1.1 Free-Space Propagation Model

When there is a clear and unobstructed line-of-sight (LOS) path between the transmitter and receiver, the free space propagation model is used to predict the strength of received signal as follows.

$$P_r = \frac{P_t G_t G_r \lambda^2}{(4\pi d)^2 L_s} \quad (2.1)$$

where P_t is the transmission power, G_t and G_r are the antenna gain at the transmitter and the receiver respectively, λ is the wavelength of radio, d is the distance between the transmitter and the receiver, and L_s is the system loss. In practice, the satellite communication system and the microwave line-of-sight link typically undergo free space propagation.

2.2.1.2 Two-Ray Ground Propagation Model

In radio channels, a single direct path between the transmitter and the receiver is rarely the only physical means for transmission propagation, so the free space propagation model is inaccurate in most cases. The two-ray ground reflection propagation model considers both a direct path and a ground reflected path. It has been indicated that it is accurate for predicting the strength of large-scale signal over long distances with a tall height basestation. The received power can be calculated by using following equation.

$$P_r = \frac{P_t G_t G_r h_t^2 h_r^2}{d^4 L_s} \quad (2.2)$$

where h_t and h_r are the height of transmitter's and receiver's antennas respectively.

2.2.1.3 Log-Distance Path Loss Model

Based on both theoretical and experiential measurements, it indicated that the average strength of received signal decreases logarithmically with distance, regardless of whether the environment is outdoor or indoor. Through large practical measurement work and statistics, the average large-scale path loss is expressed as a function of distance by introducing a path loss exponent as follows.

$$PL(dB) = \overline{PL}(d_0) + 10n \log_{10}\left(\frac{d}{d_0}\right) \quad (2.3)$$

where $PL(dB)$ is the path loss in dB, d_0 is the close-in reference distance determined from measurements close to the transmitter, and n is the path loss exponent which indicates the rate at which the path loss increases with distance. Based on the experiential measurement, different path loss exponent value are chosen from 2 to 4 for different environments, e.g. $n=2$ for free space.

2.2.2 Multiple Channel Access Schemes in Communications

Multiple channel access schemes are applied in telecommunication systems and computer networks. Based on the multiplexing method which is provided by the physical layer, multiple channel access schemes allow several users to connect to the same multi-point transmission medium and share its channel and capacity. In the following sections, three main multiple channel access schemes are presented.

2.2.2.1 Frequency Division Multiple Access (FDMA)

FDMA is a channel multiple access approach applied in the first generation of analog cellular communication systems (e.g. Advanced Mobile Phone Service (AMPS) installed in North America). FDMA allocates one or several frequency bands to the individual user at a time. In other words, each user transmits and receives at a unique frequency band which is different from frequency bands other users used.

However, users who share the same frequency band will produce interference which will disrupt the transmissions between each other, so the frequency band is reserved by the users and cannot be used by other users in a specific range. As the resource of frequency spectrum is limited, FDMA results in low efficiency of the frequency spectrum utilization, hence the FDMA system is considered as the typically narrowband communication systems. In order to reuse the frequency spectrum, channel allocation and demand assignment become very important in FDMA systems.

2.2.2.2 Time Division Multiple Access (TDMA)

Unlike FDMA, TDMA allows multiple users to share the same frequency band through allocating signals into different time slots, and this increases the amount of data that can be transmitted. TDMA along with FDMA and Frequency-Division Duplexing (FDD) is widely applied in the digital second generation cellular communication systems, such as the Global System for Mobile Communications (GSM) in Europe and IS-136 in the USA. A major advantage of TDMA is that each user only needs to transmit and receive for its own time slot. In order to reduce the effect of co-channel interference, multipath and fading, the frequency hopping is used in TDMA systems (e.g. GSM) to enable the information of users to move from one frequency channel to another at fairly short intervals.

2.2.2.3 Code Division Multiple Access (CDMA)

CDMA spreads the frequency bandwidth for the signals through a coding data stream with a high rate chip (otherwise known as significant sequence or spreading codes). This technology reduces the noise effect at the input of the receiver through spreading bandwidth to obtain a spreading gain from this process. Each user in CDMA systems is allocated a different sequence of spreading codes to modulate signals. The signal encoded with the specified spreading codes is detected at the

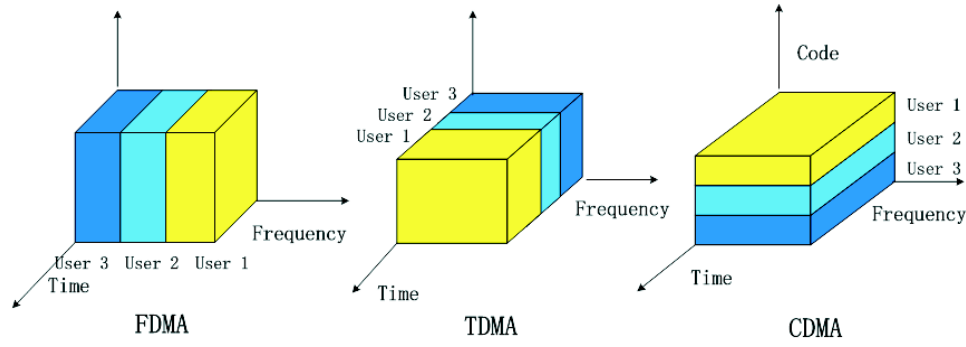


Figure 2.1: FDMA vs TDMA vs CDMA

receiver, while signals with different spreading codes are considered as the interference. On this basis, the capacity of CDMA system is mainly dependent on the number of users within a cellular coverage.

Due to the fact that users in CDMA systems share the same bandwidth, the spreading codes are used to distinguish the target user from others in order to reduce the interference at the receiver. If the spreading codes of different users are orthogonal, there will be no interference to interrupt the signals at the input of receiver. However, the number of orthogonal spreading codes is limited in practical CDMA systems, so the spreading code design and management become very important. To date, many different types of spreading codes have been designed, such as Walsh codes, m-sequence and Gold codes. Walsh codes have good cross-correlation to provide orthogonal property between multiple spreading codes and therefore result in less MAI. However, the auto-correlation of Walsh codes is very poor, hence Walsh codes are only available for synchronized transmissions. Unlike Walsh codes, the auto-correlation property of the m-sequence is excellent in terms of overcoming the multi-path effect, but the cross-correlation of the m-sequence is not as good as that of Walsh codes. Gold codes are designed through two m-sequences, where the corresponding cross-correlation is much better than that of the m-sequence. However, non-codes have both good cross-correlation and auto-correlation to overcome both the MAI and multi-path effects. In order to achieve the best trade-off performance, one possible solution is to use spreading codes with good cross-correlation as the inner code like Walsh codes and spreading codes with good auto-correlation as the outer code like the m-sequence, in a CDMA communication system, such as

the IS-95.

As the same frequency and time is shared by all users, there exists a near-far problem in CDMA systems. The near-far problem is a condition that it is difficult for a receiver to detect weak signals transmitted from far-away users due to strong signals transmitted from users who are closed to the receiver. In order to solve this problem, the CDMA system uses a simple power control solution, which achieves the same Signal to Noise Ratio (SNR) at the receiver for all transmitters through reducing the transmission power at closer transmitters.

The main difference between FDMA, TDMA and CDMA can be simply concluded: FDMA divides the user access by frequency; TDMA divides the user access by time; and users in CDMA systems share the frequency and time but are distinguished by different spreading codes (Figure 2.1). In the third and second generation cellular communication systems, CDMA and TDMA are always combined with FDMA in their systems.

Currently, CDMA has been widely applied in the third generation (3G) cellular communication systems and WiFi networks. Three different 3G standards are defined in the Universal Mobile Telecommunication System (UMTS): Wideband CDMA (WCDMA), CDMA2000 and Time Division Synchronous CDMA (TD-SCDMA).

WCDMA is a 3G standard deployed in Europe. WCDMA uses the Direct Sequence CDMA (DS-SS) channel access and FDD methods. WCDMA operates on a 5MHz bandwidth radio channel with a 3.84Mbps chip rate and achieves a 384Kbps data rate with wide area coverage and 2Mbps with local coverage. Based on Signal-to-Interference Ratio (SIR), the adaptive power control scheme is applied to solve the near-far problem in WCDMA systems.

CDMA2000 (also known as IMT-CDMA multi-carrier) is an evolution of IS-95 to 3G communication services. It has four phases: CDMA2000 1X, CDMA2000 1XV-DO, CDMA2000 1XV-DV and CDMA2000 3X. The first phase, CDMA2000 1X operates on a 1.25MHz bandwidth and supports a peak data rate of up to 144kbit/s. The second phase, CDMA2000 1XV-DO improves the data rate to 2.4Mbit/s through separating voice and data on separate channels. The third phase, CDMA2000 1XV-DV further enhances the data rate and provides a data rate ranging from 3Mbit/s to 5Mbit/s. Finally, the fourth phase, unlike the previous three phases, CDMA2000 3X uses three 1.25MHz spectrum channels and achieves a data rate from 2Mbit/s to 4Mbit/s.

TD-SCDMA is pursued in China as it has advantage of being able to handle the need of densely populated areas. Unlike WCDMA, TD-SCDMA uses Time-division Duplexing (TDD) to dynamically adjust the number of timeslots for both uplink and downlink to accommodate asymmetric traffic based on different data rate requirements. TD-SCDMA also takes the advantage of TDMA by reducing the number of users in each time interval. This in turn reduces the implementation complexity of multi-users and beamforming, but also reduces the coverage and weakens the mobility of systems. TD-SCDMA also applies the continuous timing adjustment to synchronize the uplink signals at the basestation, and this technique reduces the interference between users through more complex hardware design.

2.3 Medium Access Control Protocols

The MAC layer controls the permission of the access medium channel for a node (or user) to transmit packets in a network, and this therefore allows the nodes to share the same physical medium. The MAC protocols are responsible for detecting or avoiding the collision of data packets, hence this reduces the probability of re-transmissions. Based on the application, the MAC protocols can be classified into many categories. In this section, three main MAC protocols for wireless communications are presented: Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA), Distributed MAC Protocol and Central MAC Protocol.

2.3.1 Carrier Sense Multiple Access with Collision Avoidance

CSMA/CA is a wireless network multiple access approach at the MAC layer in computer networks. CSMA/CA is used in the 802.11-based WLAN and can be considered as the original version of the Distributed Coordination Function (DCF) protocol. Three features of CSMA/CA are exploited: carrier sensing scheme, Request_to_Send (RTS) and Clear_to_Send (CTS) exchange schemes, and collision avoidance.

Regarding the carrier sensing scheme, the transmitter gets feedback from the receiver which detects a carrier wave within coverage before transmission. If a carrier is detected (which means that the channel is busy), the transmitter will wait for a period and hold its own transmission. Once the channel within a carrier

sensing range is idle or free, the transmitter is permitted to use the channel for transmissions.

The RTS/CTS handshake scheme is an optional method to implement virtual carrier sensing in CSMA/CA, and it aims to reduce the collisions caused by the hidden-terminal problem in ad hoc networks. When a source node wishes to transmit a data packet, it firstly sends a RTS packet. If the destination node receives the RTS packet successfully, a CTS packet is replied to the source node to confirm that the destination node is ready to receive the data packet. On this basis, the source node needs to wait for a certain amount of time which should include both the RTS and CTS packet transmission intervals before trying to access the medium channel for data transmissions. This RTS/CTS handshake scheme is based on the assumption that all nodes have the same coverage in networks.

Another technique used in CSMA/CA is collision avoidance, which is used to reduce the probability of collisions that occur on the channel. If the channel is sensed as busy, the transmitter will be hold for a "random" interval before retrying transmission. Another version of CSMA is CSMA/CD (CSMA with collision detection) is also a modification of pure CSMA. CSMA/CD is normally used in the Ethernet to terminate transmission when a collision is detected as to also reduce the probability of another collision occurring.

2.3.2 Distributed MAC Protocols

In the IEEE 802.11, the Distributed Coordination Function (DCF) is defined as the fundamental technique at the MAC layer to allow for the sharing of the medium channel between nodes in a network. Based on the CSMA/CA, the DCF employs its binary exponential backoff algorithm for nodes to contend channel access permission, and uses the RTS/CTS mechanism to avoid collisions from occurring.

However, DCF does not provide Quality of Service (QoS) (which is the requirement of users from different aspects) for users to obtain high and low priority to access the channel early or not. Another limitation of DCF is that, when a node obtains permission to access the medium channel for transmissions, it will take over the channel as long as the duration of the transmission period lasts. If a low data rate is achieved at the transmission link, it will take a long time to send its packets. Other nodes have to endure a long waiting period to obtain the permission for transmissions, resulting in inefficient transmissions and low network throughput. The details of DCF will be presented and discussed in section 5.2.

In 2005, Enhanced DCF (EDCF) was developed in the IEEE 802.11e [13] as the enhancement of the initial DCF and was used to support service differentiation using a distributed control. The EDCF is based on different priorities where traffic is to be delivered with four access categories (ACs). Each AC achieves a different channel access, and the differentiation is achieved through varying the length of the contention window and Arbitrary Interframe space (AIFS). Smaller AIFS and contention window sizes are used for high priority users to have a better chance of accessing the channel earlier. The EDCF provides 8 different traffic classes (or 8 queues) in one station to support 8 user priorities. If there is more than one queue finishing the backoff process at the same time, an internal collision occurs. In order to avoid this, a scheduler inside the station grants the Transmission Opportunity (TXOP) to the AC with the highest priority. Then, the highest priority frame is chosen to transmit and other lower class AC frames increase their backoff counters with corresponding contention window sizes. By using this approach, the highest priority AC frame is able to access the channel earlier.

In order to enhance the performance and achieve better throughput, packet bursting and Contention Free Burst (CFB) are used for EDCF in the 802.11e. Once a node is allowed to access the channel, it can send more than one frame without contending again. If a collision occurs, the packet bursting will be terminated. EDCF bursting reduces the network overhead, yet it also increases throughput through multiple transmissions.

2.3.3 Central MAC Protocols

The Point Coordination Function (PCF) is an optional MAC protocol used in the IEEE 802.11 to provide QoS for users. PCF is only available when nodes are connected to the network through an Access Point (AP). APs send beacon frames at regular intervals, and the PCF defines two periods between beacon frames, known as the Contention Free Period (CFP) and the Contention Period (CP).

Within the CP, DCF is adopted to provide distributed MAC control. Within the CFP, the AP sends Contention-Free-Poll (CF-Poll) packets to each node and enables nodes to send a packet. At the beginning of each contention-free period, the AP senses the medium channel. If the medium channel is idle after a PIFS interval, the AP sends a beacon frame that includes the CF Parameter Set element. When nodes receive the beacon, they update their Network Allocation Vector (NAV). After sending the beacon frame, the point coordinator transmits frames after waiting for

at least one SIFS interval. However, although the PCF provides QoS management, the achievable QoS is limited as the Traffic Class (TC) is not defined in the PCF.

The Hybrid Coordination Function (HCF) combines the EDCF with generalized PCF features to simplify the 802.11e. Unlike the PCF, HCF uses the EDCF to all nodes during the CP, when the TC and the Traffic Stream (TS) are defined. Furthermore, the nodes provide information on the lengths of their queues for each TC, and this information is used to give priority to nodes.

2.4 Routing Classification in Ad Hoc Networks

Routing is used to decide which path to follow and how to forward packets between the source node and the destination node in an ad hoc network. Based on the application, the routing protocols can be classified into many categories, such as proactive (table-driven) routing, reactive (on-demand) routing, multipath routing, flow-oriented routing, adaptive (situation-aware) routing and multi-cast routing. In this section, three main routing schemes are introduced and discussed respectively.

2.4.1 Proactive Routing

Proactive routing protocols require that nodes should keep track of all possible routing paths to send the data packet immediately through a known route. Any changes in topology are broadcast throughout the network, hence all nodes obtain information on any topology changes. However, based on this mechanism, two main disadvantages exist in proactive routing in term of the respective amount of data required for maintenance and the delayed reaction as regards on restructuring and failures.

Several proactive routing protocols have now been developed. The Destination Sequenced Distance Vector (DSDV) routing [14] is a proactive hop-by-hop distance vector routing protocol. A routing table is maintained at each node and lists all available destinations, the number of hops and the sequence number assigned by the destination node. The sequence number is used to distinguish stale routes from new ones in order to avoid the formation of loops. The nodes periodically broadcast their routing tables to their neighboring nodes to update the information.

Global State Routing (GSR) [15] is similar to DSDV, but it achieves improvements by reducing the flooding of routing messages. Each node maintains a routing table which includes a neighbor list, a topology table, a next hop table and a dis-

tance table. However, the large size of update message in GSR consumes a large amount of network bandwidth. In order to solve this problem, Fisheye State Routing (FSR) [16] was developed as the updated version of GSR. Instead of containing all nodes' information in an updated message, FSR only frequently exchanges the information to closer nodes rather than farther nodes, and this reduces the updated message size. By using this approach, each node obtains accurate information about its neighboring node, but the accuracy of information decreases with the increasing of distance from the source node. However, even if nodes do not have accurate information regarding distant nodes, the packets will still be forwarded correctly as the information in routing table becomes more and more accurate when the packets get close to the destination node.

2.4.2 Reactive Routing

Reactive routing only attempts to establish routes when the source node is required to transmit packets and such that it is also known as on-demand routing. Once a route is discovered and established, it is maintained by using a routing maintenance procedure until this route is no longer need. To date, many reactive routing protocols have been developed, such as Dynamic Source Routing (DSR) [17], Cluster based Routing Protocol (CBRP) [18], and Ad Hoc On-demand Distance Vector (AODV) [3] amongst others. AODV is a typical and popular reactive routing protocol, and its routing discovery mechanism is based on finding the minimum number of hops. In this section, AODV is used as an example to show the process of a reactive routing protocol.

At the source node, the HELLO message is broadcast in order to find source node neighbors. Once the neighborhood of source node is conformed, route request packets (RREQ) are generated by the source node are broadcasted to neighbors. Each intermediate node that has received the RREQ message then checks its local routing table to see whether or not there is a valid route towards the destination node. All nodes are checked so that they do not re-broadcast a RREQ by using a sequence number in their RREQ message. If the intermediate node has a route to the destination node, it will rebroadcast RREQ message to its neighbors. When the destination node receives RREQ message or the information inside the RREQ is fresher, a route reply (RREP) will be generated and routed back to the source node by using a unicast reverse path established by the corresponding RREQ message. Once the source node receives the RREP message, it will transmit the data packet

to the destination node.

2.4.3 Multi-path Routing

Multi-path routing provides three main advantages: fault-tolerance (or reliability of transmissions), load balancing and higher aggregate capacity.

In term of fault-tolerance, when a transmission link is broken, instead of discovering a new route, the alternative remaining route will be used to continuously forward the packet from the source node to the destination node without too much delay. By using load balancing, the traffic can be distributed over multiple routing paths in order to reduce the congestion. Furthermore, multiple routing paths are able to be used to transmit data packets simultaneously and hence increase the network capacity. However, compared with single-path routing, the overhead of multi-path routing is larger to maintain multiple routing paths records [19].

In [20], a multipath routing protocol called the *Ad Hoc On-demand Multipath Distance Vector (AOMDV)* extends the *AODV* protocol by computing multiple routing paths during a route discovery process, and it is used for highly dynamic ad hoc networks where the link failure and the route break occur frequently (e.g. mobile ad hoc networks). Unlike the *AODV*, duplicate copies of *RREQ* message are not immediately discarded at the intermediate nodes, and are examined to see whether this provides a new node-disjoint route path to the destination. The simulation result in [20] shows that the *AOMDV* offers a significant delay reduction in high dynamic ad hoc networks. However, although different routing paths may be discovered for a node pair, only a single route is active. By introducing a novel load balancing approach to allocate the traffic among the disjoint paths, L. Bononi and M. D. Felice [21] developed a new routing protocol called *Concurrent Separated AOMDV (CS – AOMDV)*. Based on cross-layer measurements, the traffic-path allocation scheme is a function of the number of hops and congestion level of each path.

The *AODV Multipath (AODVM)* [22] is another extension of the *AODV* protocol. The main difference between *AOMDV* and *AODVM* is that node-disjoint paths are established during the route request (e.g. *RREQ*) procedure in *AOMDV*, as well as the route reply (e.g. *RREP*) procedure in *AODVM*. In [23], a novel routing protocol, *AODVM with Path Selection Probabilities (AODVM – PSP)*, is developed based on path selection probability. The route discovery process is same as the *AODV* and the multiple route set up is same as *AODVM*. The rout-

ing probability is a function of transmission delay time and can be connected to many various expressions of the very same transmission delay time. The route is selected according to the routing probability. Based on this approach, the protocol adapts its route path selection activity to the varying data traffic status. The simulation results in [23] identify that *AODVM – PSP* has improved significantly when compared to other routing protocols in high load networks.

In [24], three multi-path routing protocols are looked at: *SMR* [25], *AOMDV* and *AODVM*. All three protocols are examined in *NS2* and the corresponding performance is compared. Based on the experiment results, reference [24] identifies that *AOMDV* protocol is more robust and performs better than others in most of the simulations, especially the high-mobility networks. *AODVM* gives the best performance in static networks with high node density, high load and low mobility. *SMR* performs best load balancing in low-density, low-load networks.

2.5 Cross-layer Design in Ad Hoc Networks

Unlike traditional layered network models (e.g. OSI model), cross-layer design aims to exchange state information between layers in order to obtain the performance gain. Protocols are designed by exploiting interactions and state information flowing among non-adjacent layer of the network architecture stack. A number of overview and surveys [26] [27] [28] [29] on cross-layer design in ad hoc networks are presented to show the development in this area. Reference [30] discusses the potential risk in cross-layer design, and it points out that the interaction among different layers exploited by the cross-layer design may affect the whole system.

In [31], a framework is proposed for cross-layer design to utilize energy in wireless ad hoc networks. The proposed framework provides a joint power control and scheduling solution by considering both physical and MAC layers in order to minimize the total transmission power based on QoS requirements (e.g. data rate or BER). Reference [32] studies the interaction between MAC and routing protocols by considering different parameters. A cross-layer feedback control mechanism is developed in [33] to dynamically adjust the network topology in order to improve the quality of video transmission in multi-hop ad hoc networks. Furthermore, subject to control the transmission rate and packet loss, a set of coding strategies are also proposed based on the channel condition. In [34], a cross-layer optimization framework is designed in multi-hop multi-cast wireless mesh networks, in which the

data routing sub problem is decomposed at the networks layer, and a power control sub problem is considered at the physical layer with a set of Lagrange dual variables coordinating interlayer coupling. After studying the cross-layer design in theory, the cross-layer concept has been examined and evaluated in [35] [36] [37] for the implementation purpose.

2.6 System Model

2.6.1 Wireless Communication System Model

This thesis focuses on the CDMA system at the physical layer and therefore a simple wireless communication system model is proposed in this section. It assumes that there is a large-scale radio channel between the transmitter and the receiver. In addition, the fading and the channel is not frequency-selective, so the received power is only related to the total transmission power at the transmitter and the propagation path loss. Given a distance between a transmitter and a receiver, the propagation path loss, h , is defined in [38] and measured in dB as follows.

$$h = 40 \log_{10} d + 20 \log_{10} f - 20 \log_{10} h_t h_r \quad (2.4)$$

where f is the carrier frequency in Gigahertz, and h_t and h_r are the heights of the transmitter and the receiver respectively. Based on the equation (2.4), once the carrier frequency and the height of transmitter and receiver are fixed, the path loss becomes a function of distance, e.g. $h(d)$.

At the input of the receiver, the noise distorts the waveform of received signals. Additive White Gaussian Noise (AWGN) is a simple mathematical model to study the background noise (which consist of thermal and equipment noise) with constant spectral density and a Gaussian distribution of amplitude in communication systems when there is no fading and frequency-selective, so only the AWGN is considered as the background noise here for modelling as follows

$$P_{noise} = kT_n BW \quad (2.5)$$

$$T_n = T_0 + T_e = T_0 + T_0(10^{\frac{NF}{10}} - 1) = T_0 \cdot 10^{\frac{NF}{10}} \quad (2.6)$$

where k is the Boltzmann constant, BW is the bandwidth, T_0 is the background temperature, T_e is the error temperature and NF is noise figure which is dependent on the property of receivers.

A CDMA system is assumed to operate over a transmission bandwidth BW , with a spreading factor N_{SG} and a T_c chip period. The total energy available per symbol period is $E_T = N_{SG}T_cP_T$. Given E_T at the transmitter, propagation path loss h calculated from the equation (2.4), and noise variance $\sigma^2 = \frac{N_0}{2}$ (where N_0 is the noise power spectral density), the total received SNR_T is measured at the receiver as follows.

$$SNR_T = \frac{10^{\frac{-h}{10}} E_T}{2\sigma^2} \quad (2.7)$$

However, the system capacity, C , is based on the assumption of being error-free, and is usually referred to as the Shannon bound or theoretical capacity. In practical transmissions, there is always a system gap between the theoretical capacity and the bit rate. The bit rate achieved by Pulse Amplitude Modulation (PAM) and Quadrature Phase Shift Keying (QAM) is found experimentally with heuristic theoretical justification [39] as follows.

$$b = \frac{1}{2} \log_2 \left(1 + \frac{SNR}{\Gamma} \right) \quad (2.8)$$

Where b is the discrete bit rate, Γ is the system gap which equals to the theoretical capacity time the inverse of the practical discrete bit rate. Based on equation (1.1) and (2.9), if the SNR becomes reasonably large (normally 20dB or more), the capacity will shows a semi-log plot. However, this rule cannot be applied to a low SNR range. On this basis, if uncoded PAM and QAM transmissions are within a reasonable SNR range, there exists an approximate gap value which relates to the corresponding BER [39].

$$\Gamma = -\frac{\ln 5BER}{1.5} \quad (2.9)$$

Based on the equation (2.9), the system gap is only determined by the BER requirement. In order to minimize the system gap and therefore enable the bit rate close to the theoretical capacity, coding schemes are used by either adding redundancy bits or repeating original bits during transmissions. In other words, coding techniques are used to achieve lower BER value under given constant SNR or reduce the SNR requirement under given constant BER measurement comparing to uncoded systems. The cost of this advantage is a complex encoder and a more complex decoder in communication systems. In the following chapters, $\Gamma = 8$ (which is related to $BER = 1 \times 10^{-6}$) is used for all performance evaluations.

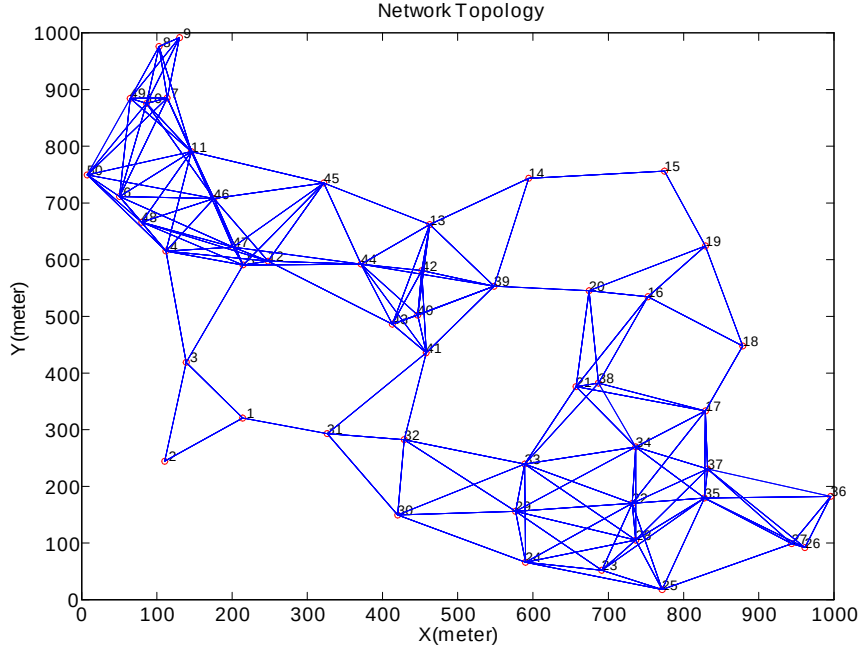


Figure 2.2: A 50-node Network Topology

2.6.2 Network Topology

A wireless ad hoc network with N randomly distributed nodes on an x-y grid is considered and each node is confirmed in order to keep at least one neighbor (Figure 2.2). In figure 2.2, the number inside the area indicates the node ID. Two nodes which are within a transmission range between each other are connected. It is assumed that all nodes in the network are immobile and multi-hop transmissions are required. A source node is able to aggregate the data and transmit the data packet to a destination node. A group of nodes in which data are relayed from the source node to the destination node is called a routing path. If the network is sufficiently large and densely connected, it is possible to have multiple routing paths between the source node and the destination node. Here, all selected multiple paths are assumed to be node-disjoint, which means no nodes other than the source and the destination nodes are used in more than one routing path.

In an ad hoc network, the transmission range is defined as the maximum range in which the signal can be detected at the receiver. At the MAC layer, the maximum carrier sensing range is defined as the range in which the transmission power can be sensed, and it is normally twice the maximum transmission range.

2.7 Conclusion

In this section, the background related to this thesis and the system models are given. Different versions of released IEEE 802.11 standards are presented. As this thesis focuses on the physical, MAC and network layers, wireless communications, MAC protocols and routing schemes in ad hoc networks are presented in detail. Furthermore, the development of cross-layer design is also discussed. Finally, the system model used in this thesis is given.

Chapter 3

Two-Group Rate Allocation

3.1 Introduction

Currently, CDMA technology has been widely applied in mobile communications (e.g. 3G [5]) and ad hoc networks (e.g. IEEE 802.11 [1]). In CDMA systems, narrowband data signals are multiplied by spreading codes which have a much higher rate than the data rate. All users in CDMA systems share the same carrier frequency and each user owns one specific sequence of spreading codes to spread the bandwidth and to distinguish the desired signal from others. The desired signal is detected only through these known codes and all other received signals are considered as interference. In traditional CDMA networks, the IS-95 and IEEE 802.11 for example, only one significant sequence is applied to multiply with the data stream for a user, and this is known as a single-code CDMA transmission.

However, the traditional single-code CDMA system does not satisfy the high data rate demand of fast developing wireless communications. In order to achieve high data rates for future CDMA networks, the multi-code CDMA [6][7] associated with resource allocation techniques is developed and widely recognized by many research studies to provide more system capacity when compared with a single-code CDMA.

Multi-code CDMA systems were designed by splitting a high data rate into smaller data rates using only one serial-to-parallel converter. These multiple small data streams are encoded with different spreading codes and transmitted with the same carrier frequency in parallel channels for a single user. To date, the multi-code CDMA technique has been considered in the downlink of HSDPA [5]. In HSDPA systems, all channels use a spreading factor $N_{SG} = 16$ and the Orthogonal Variable

Spreading Factor (OVSF) codes (which provide good cross correlation) are used for downlink data transmissions. In order to achieve a large dynamic range of the HSDPA link adaptation and to maintain a good spectral efficiency, a user may simultaneously utilize either $K=5, 10$, or 15 multiple spreading codes in parallel. By employing 15 multiple spreading codes, 16QAM modulation scheme and $3/4$ coding rate, the maximum theoretical data rate achieved in HSDPA systems is up to 10.7Mbps . The details of HSDPA system will be presented and discussed in section 3.5.

3.2 Multi-code CDMA System Capacity

In a multi-code CDMA system, Welch's Bound is defined in [40] as a system capacity upper bound and it is the sum of the specific capacities achieved by all sub-receivers. Based on the Shannon theory and the system gap concept [39], the practical system capacity in multi-code CDMA systems is defined as follows.

$$C_T = \sum_{k=1}^K C_k = \sum_{k=1}^K \frac{1}{2} \log_2 \left(1 + \frac{SNIR_k}{\Gamma} \right), \quad k = 1, 2, \dots, K \quad (3.1)$$

where $SNIR_k$ is the SNIR value measured at the input of k^{th} receiver, K is the total number of spreading codes available ($K = 1$ means single-code CDMA system and $K > 1$ means multi-code CDMA system), k is the index number of spreading codes for $k = 1, \dots, K$, and Γ is the system gap. Based on the equation (3.1), for given total number of spreading code, K , and system gap, Γ , the multi-code CDMA capacity is dependent on the $SNIR$ measured at each subreceiver.

The Signal to Noise and Interference Ratio (SNIR) measured at the input of k^{th} receiver is expressed as followed when there is no multipath effect.

$$SNIR_k = \frac{h_k p_k |\omega_k^H s_k|^2}{\sum_{j \neq k} h_k p_j |\omega_k^H s_j|^2 + \sigma^2 |\omega_k^H \omega_k|}, \quad k = 1, 2, \dots, K \quad (3.2)$$

Where h_k is the channel gain (or path loss) of the k^{th} propagation channel, p_k is the transmission power allocated at the k^{th} transmitter and the total transmission power is $P_T = \sum_{k=1}^K p_k$, s_k is the spreading codes used at the k^{th} transmitter, ω_k is the receiver decorrelator coefficient in the k^{th} channel, σ is the noise variance, and $\sum_{j \neq k} h_k p_j |\omega_k^H s_j|^2$ is the Multiple Access Interference (MAI) factor. Based on the equation (3.2), two parameters at the transmitter, transmission power (p_k) and spreading codes (s_k), affect the SNIR value measured at the receiver.

Given transmission power, p_k , at each subchannel, the maximum SNIR of the desired signal is achieved when $\sum_{j \neq k} h_k p_j |\omega_k^H s_j|^2 = 0$. However, if these spreading codes are not orthogonal to each other perfectly, there will exist a MAI (e.g. $\sum_{j \neq k} h_k p_j |\omega_k^H s_j|^2 > 0$), which decreases the SNIR value at the receiver and therefore reduces the related system capacity. Based on this, the optimal spreading code design aims to maximize SNIR through minimizing MAI at the receiver. The chip rate of a spreading code is the number of pulses generated per second, and it is normally much larger than the symbol rate. The ratio of the symbol period to the chip period is known as the spreading factor (or spreading gain) as follows.

$$N_{SG} = \frac{T_{symbol}}{T_c} \quad (3.3)$$

Given spreading codes $\bar{s} = [s_1, \dots, s_K]^T$ and total power at the transmitter, the power allocation aims to assign available power (p_k , $k = 1, 2, \dots, K$) to radio channels to maximize system capacity. Strong transmission power provides good quality of target signal, so the power allocated in channels interact with each other and therefore results in MAI for other channels. In order to achieve the maximum capacity in multi-code CDMA systems, the transmission power needs to be efficiently distributed. In practical communication systems, power allocation is normally associated with the modulation and coding scheme (or otherwise known as adaptive modulation and coding) in order to achieve the discrete bit transmissions. To date, the research study on resource allocation in multi-code CDMA systems fall into two main areas: power allocation and optimal spreading codes design. If spreading codes show perfect orthogonal property or the transmission power is efficiently allocated, the system capacity will be maximized. The related work on power allocation and optimal spreading codes design will be presented in section 3.3 and 3.4.

3.3 Power Allocation in Multi-code CDMA Systems

3.3.1 Related Work

Power allocation aims to maximize the sum capacity in multi-code CDMA systems through distributing the transmission power at all sub-transmitters. Based on the generalized processor sharing concept, in [41], a resource allocation approach is developed to achieve a high transmission rate. Imran [42] developed one discrete

resource allocation approach based on the waterfilling concept through combining power control and coding scheme. Reference [43] studies the greedy rate scheduling in multi-code CDMA systems to maximize the downlink throughput. In addition, this greedy algorithm provides a polynomial time complexity of $O(N)$ (where N is the number of users), which means that the computational complexity increases as a linear function of N , so it is able to be implemented in real multi-code CDMA systems due to its low complexity of design.

A significant gain on transmission power reduction is achieved by using an unequal power allocation approach with a Minimum Mean Square Error (*MMSE*) receiver in HSDPA systems [44]. The MMSE coefficient is updated iteratively in order to reduce the inter-code interference. Although this joint power control and MMSE receiver approach [44] eliminates the problem on residual received SNR in HSDPA systems, it still operates on an equal data rate allocation mechanism. Due to the fact that a lot of energy is required to increase the data rate to the next high level data rate for all parallel channels in HSDPA systems, this approach causes inefficient energy utilization and therefore results in residual energy in HSDPA systems.

Given the limited resource, e.g. the limited amount of number of spreading codes and transmission power, in HSDPA systems, a two-best user scheduling algorithm is developed in [45] by considering a power control allocation scheme to maximize the total network throughput. In addition, an Adaptive Proportional Fairness (APF) selection scheme is also developed to provide fairness between users. However, this power allocation approach also only considers one type of modulation and coding scheme, and hence results in residual energy in multi-code CDMA systems as well. Such equal rate and power control solutions are widely applied to maximize the data rate in multi-code CDMA systems for both single-user [48] and multiple-user systems [46]. Different scheduling algorithms are developed to improve the transmission data rate by taking into account the channel condition, the limited number of multiple spreading codes, the type of modulation and coding schemes, the available transmission power [46] and the maximum achievable bit rates [47].

Another direction to consider in terms of joint power allocation, is one taking into account other areas of thought. Kwan and Leung [46] introduce a joint power and rate control algorithm by allocating different power levels to different traffic classes for satisfying different SNR requirements. In [47], the resource allocation is jointly (or sequentially) optimized through formulating the radio resource on the down-

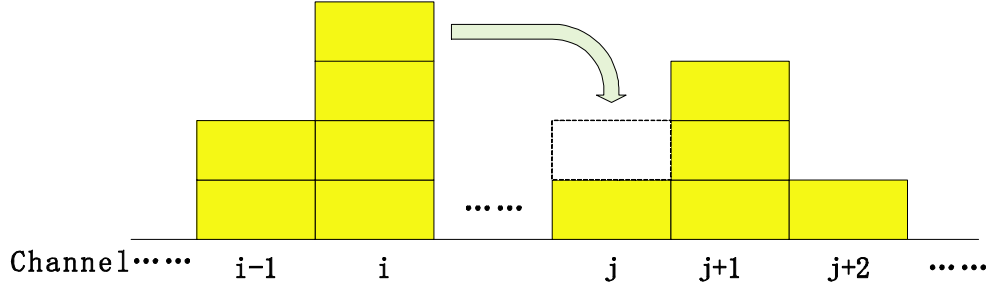


Figure 3.1: Discrete Bits Loading in Multi-tone Systems

link of CDMA network as a mixed integer non-linear programming problem. The availability of perfect channel state information (CSI) is assumed and used as the feedback information for multiuser scheduling. Furthermore, a sequential optimization (SO) procedure is developed to reduce computational complexity. However, this resource allocation approach incurs a high bandwidth overhead.

In [48], an Evolutionary Simulated Annealing (ESA) approach is proposed to jointly optimize the number of multiple spreading codes and power levels for all users. Unlike reference [48], reference [49] presents another joint algorithm for multi-codes by using adaptive modulation and coding schemes. Furthermore, a maximum Carrier-to-Interference ratio (C/I) scheduling algorithm is proposed to increase the expected channel condition at the input of the receiver in order to improve the network throughput in multi-code CDMA systems.

3.3.2 Discrete Bits Loading

One optimization application called discrete bit loading developed in [50] and [51], provides a practical approach to maximize sum capacity in multi-tone systems. The discrete bit loading approach derives an optimal bit allocation which supports the maximum information units or consumes minimum energy under respective constraints. It swaps multiple fixed granularities between channels, in which the information granularity value is defined as the smallest reducible values of data bits loaded on each channel.

With regard to the discrete bit loading approach, there is an attempt for one granularity bit to be unloaded at each channel to discover the channel with the maximum decreasing energy. Similarly, there is another attempt for one granularity bit to be loaded at each channel and a channel with the minimum incremental energy is proposed to be found. The incremental energy and decreasing energy of the k^{th}

channel at iterative steps are calculated as follows.

$$e_{decre,k} = E(b_k) - E(b_k - \beta), k = 1, 2, \dots K \quad (3.4)$$

$$e_{incre,k} = E(b_k + \beta) - E(b_k), k = 1, 2, \dots K \quad (3.5)$$

where β is the granularity (or the minimum information unit available to be transmitted), $E(b_k)$ is the energy required when b_k bits are loaded in the k^{th} channel, while $E(b_k - \beta)$ (or $E(b_k + \beta)$) is the energy required when $b_k - \beta$ (or $b_k + \beta$) bits are unloaded (loaded) in the k^{th} channel, and e_{decre} (e_{incre}) is the decreasing (incremental) energy. When the *MPAM* modulation scheme is used, the energy required to provide b discrete bits is given as follows

$$E(b) = \Gamma \sigma^2 (2^{2b} - 1) \quad (3.6)$$

where σ is the noise variance and Γ is the system gap.

Based on equations (3.4) and (3.5), if the decreasing energy of the i^{th} channel is big than the incremental energy of the j^{th} channel, the efficient discrete loading solution is found and the related granularity bit, β , is proposed to be moved from channel i to channel j (figure 3.1). This process iterates for all sub-channels until no efficient discrete loading exists (equation (3.7)).

$$Max[e_{decre,i}] < Min[e_{incre,j}] \quad (3.7)$$

In [42], the discrete bit loading is modified and examined in multi-code CDMA systems when random codes are applied at low SNR range. Although random codes have excellent auto-correlation properties to overcome the multipath effect, the cross-correlation property is poor and therefore results in MAI at the input of the receiver. In order to reduce the MAI and therefore improve the system capacity, a discrete bit loading approach associated with a power control approach is developed in [52]. Both rate adaptive discrete bits loading and margin adaptive bits loading are designed for multi-code CDMA systems to achieve different objectives. Given amount of energy at the transmitter, the rate adaptive discrete bit loading aims to achieve the maximum bit rate as close to the system capacity in multi-code CDMA systems as possible. From an alternative perspective, however, when a required transmission bit rate is fixed, margin adaptive discrete bit loading proposes to minimize the energy consumption. However, these approaches [42] only provide good performance at the low SNR range, but not at the high SNR range. Another weakness of this approach is that it requires huge computational complexity for bit loading calculation and also a complex receiver design (MMSE filter and decoder).

In order to achieve good performance within high SNR range, extension work known as prior codes selection is developed [53]. Firstly, a large number of random codes are generated. Secondly, the sum cross-correlations of codes are calculated. Then the codes, which have the worst sum cross-correlations, are deleted from the list of random codes. This process iterates until the number of codes equals to the number of channels in multi-code CDMA systems. The performance of this approach has been examined in [53] to show its enhancement within high SNR range, e.g. 30dB.

3.4 Optimal Spreading Code Design in Multi-code CDMA Systems

In this section, two optimal spreading code designs, Ulukus WBE codes and Grassmanian codes are presented and examined.

3.4.1 Ulukus WBE Codes

Ulukus WBE codes [54] aim to optimize the system capacity through minimizing the Total Squared Correlation (*TSC*) of spreading codes. This approach iteratively updates spreading codes by using MMSE receivers and feedback loops from the receiver to the transmitter. Given the knowledge spreading codes and measured noise at the receiver, the MMSE coefficient c_k at the k^{th} receiver is calculated through the following approach:

$$c_k = \frac{A_k^{-1} s_k}{(s_k^H A_k^{-2} s_k)^{\frac{1}{2}}} \quad (3.8)$$

$$A_k = \sum_{j \neq k}^{N_{SG}} s_j s_j^H + \sigma^2 I_{N_{SG}} \quad (3.9)$$

where s_k is spreading codes allocated in the k^{th} channel, σ^2 is the noise variance, and $I_{N_{SG}}$ is one $N_{SG} \times N_{SG}$ diagonal matrix where N_{SG} is the length of spreading codes (or spreading factor). Next, the MMSE coefficient c_k is forwarded to the transmitter as the feedback information and replaces the spreading codes s_k at the k^{th} transmitter as follows.

$$\bar{S} = [s_1 \cdots s_{k-1} \ c_k \ s_{k+1} \cdots s_K] \quad (3.10)$$

where $\bar{S}$ is a vector of spreading codes proposed to be allocated in multiple channels. This process is iterated until the difference between any pair of receiver coefficient and spreading code is sufficiently small. Each update decreases the TSC of the set of spreading codes and gradually produces a better set of spreading codes. If the number of codes is less than, or equal to, the spreading factor, e.g. $K \leq N_{SG}$, the spreading codes will be converged to a set of orthogonal codes. However, when the required number of spreading codes is larger than the spreading factor, e.g. $K > N_{SG}$, the system becomes overloaded and the system capacity achieved by Ulukus WBE codes sharply falls. The performance of Ulukus WBE codes in both under-load and overload multi-code CDMA systems will be examined and discussed in section 3.4.3.

3.4.2 Grassmannian Codes

In [55], one design method of complex optimal spreading codes is introduced based on the conference matrix [56]. Firstly, assuming $N_{SG} = 2^\beta$, $\beta \in \mathbb{N}$, one $2N_{SG} \times 2N_{SG}$ skew symmetric conference matrix $C_{2N_{SG}}$ is generated iteratively in such a way as follows.

$$C_2 = \begin{bmatrix} 0 & -1 \\ 1 & 0 \end{bmatrix} \quad (3.11)$$

$$C_{2n} = \begin{bmatrix} C_n & C_n - I_n \\ C_n + I_n & -C_n \end{bmatrix}, n = 2, 4 \cdots N_{SG} \quad (3.12)$$

Where I_n is one $n \times n$ diagonal matrix. The correlation matrix of spreading codes R can be generated as follows.

$$R = J\alpha C_{2N_{SG}} + I_{2N_{SG}}, \alpha = \frac{1}{\sqrt{2N_{SG} - 1}} \quad (3.13)$$

Let $V = [v_1, v_2 \cdots v_{2N_{SG}}]$ be the eigenvectors of R , and U be the eigenvalue of R , and a $2N_{SG} \times N_{SG}$ spreading codes matrix $S_{2N_{SG}}$ is obtained through following approach.

1. do the eigendecomposition $R = S_{2N_{SG}}^H S_{2N_{SG}} = UVU^H$
2. order eigenvalues to find out the largest N_{SG} eigenvalues and corresponding eigenvectors.
3. truncate U to be $2N_{SG} \times N_{SG}$ matrix $\tilde{U}$, to preserve only N_{SG} eigenvectors corresponding to the largest N_{SG} eigenvalues in V .

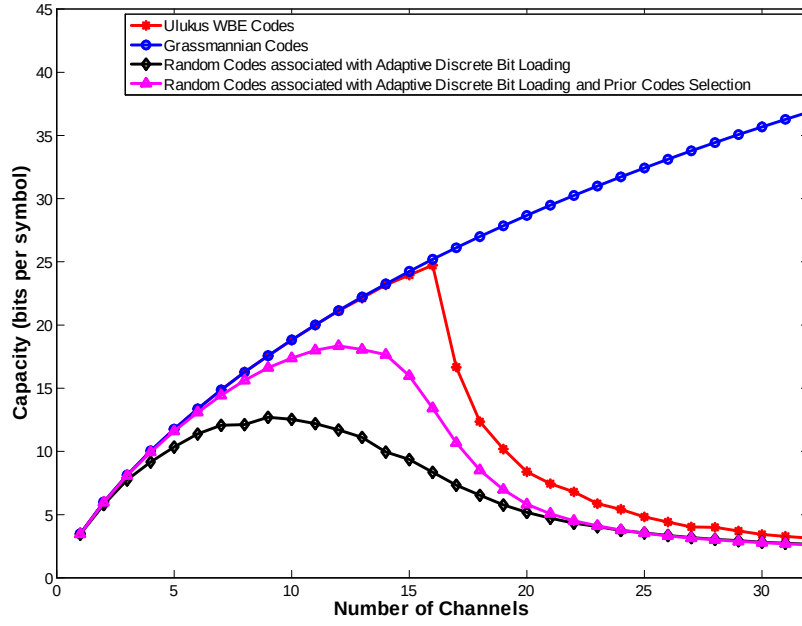


Figure 3.2: System Performance Comparison of Ulukus WBE codes, Grassmannian codes, random codes associated with rate adaptive discrete bits loading with and without prior codes selection

4. normalize each column of $\tilde{U}^H$ to get complex Grassmannian code set $S_{2N_{SG}}$.

$$S_{2N_{SG}} = V^{\frac{1}{2}} U^H = [s_1 \cdots s_{N_{SG}} \cdots s_{2N_{SG}}] \quad (3.14)$$

In this codes' matrix, all codes are complex where the codes have both real and imaginary parts, e.g. $s_k = c_k^I + jc_k^Q$, $k = 1, 2, 3 \cdots 2N_{SG}$, and the matrix generates $2N_{SG}$ spreading codes based on N_{SG} spreading factor. The interference is the same in each channel and is also minimized at the receiver. In conclusion, Grassmannian conference codes provide an approach to design complex spreading codes in two dimension modulation schemes in order to improve the system capacity in over-load multi-code CDMA systems. Based on equations (3.1) and (3.2), the capacity achieved by Ulukus WBE codes (equation (3.10)) and Grassmannian Codes (equation (3.14)) in multi-code CDMA systems can be calculated.

3.4.3 System Capacities Comparison

In this section, the performance of Ulukus WBE codes [54], Grassmannian codes [55], and random codes with rate adaptive discrete bits loading [42] in both underload

Simulation Parameters	
Maximum Available Channel (K)	32
Signal to Noise Ratio (SNR) in dB	30
Access Scheme	DSSS/CDMA
Propagation Path Loss (h)	1
Spreading Factor (SF)	16
Granularity (β)	1/3
BER	10^{-6}

Table 3.1: Simulation Parameters

($N_{SG} \geq K$) and overload ($N_{SG} < K$) multi-code CDMA systems are examined, while the MPAM modulation scheme is applied.

Here, the wireless propagation channel is assumed to be perfect, which means that there is no path loss, e.g. $h = 1$. As such, the SNR is simply equal to the ratio of the transmission power to the noise at the input of the receiver, and it is set as 30dB. The spreading factor $N_{SG} = 16$ and the maximum number of channels available is twice that of the spreading factor, e.g. $K = 32$. Regarding the system used Ulukus WBE codes and Grassmannian codes, equal power is allocated at each subtransmitter in multi-code CDMA systems which use Ulukus WBE codes and Grassmannian codes. The value of granularity β is set as 1/3 bit (which is able to be achieved by the coding scheme, for example, the Turbo code) in multi-code CDMA systems which use random codes and rate adaptive discrete bit loading. The required BER is set as 10^{-6} and the corresponding system gap is calculated based on the equation (2.9) given in section 2.6.1. Table gives all the parameters and the corresponding value used in the simulation.

Figure 3.2 gives the system capacities achieved by different techniques in multi-code CDMA systems. From the result, random codes associated with rate adaptive discrete bits loading gives poor performance, but the prior codes selection mechanism provides an enhanced solution in the under-load condition. Another weakness of this approach is that it requires huge computation for bit loading calculation and also a complex receiver design (MMSE filter and decoder). In underload multi-code CDMA systems, the system capacity achieved by both Ulukus WBE codes and Grassmannian codes is the same and up to the WBE bound of N_{SG} spreading factor. In overload systems, the system capacity achieved by the Ulukus WBE codes decreases. However, due to the complex spreading code design, Grassmannian codes generate $2 \times N_{SG}$ codes based on N_{SG} spreading factor, and the capacity achieved by Grassmannian codes increases and is up to the WBE bound of $2N_{SG}$

spreading factor. In addition, Ulukus WBE codes also require complex *MMSE* receiver design. Furthermore, Ulukus WBE codes interact with each other, which would suggest that they are not robust enough for communication systems.

Grassmannian codes, however, overcome the limitations of the Ulukus WBE codes to provide less complex system design, higher capacity upper bound, and is more robust for communication systems. Compared with random codes and WBE codes, Grassmannian codes perform much better in overload multi-code CDMA systems. Based on the result in Figure 3.2, the capacity achieved by Grassmannian codes in overload multi-code CDMA systems increases and up to 37 bit per symbol. However, the capacities achieved by both random codes and WBE codes decrease with the increase of number of channels used for transmissions.

3.5 High-Speed Downlink Packet Access (HSDPA) System

HSDPA [5] is a packet-based data service and enhances mobile wireless communication for WCDMA in 3G communication systems, and hence it is known as the 3.5G mobile communication system. HSDPA is one member of the High-Speed Packet Access (HSPA) family, and it is the downlink solution in HSPA standard. HSPA provides the data rate up to 14.4Mbit/s in the downlink (HSDPA) and 5.8Mbit/s in the uplink (High Speed Uplink Packet Access (HSUPA)). HSDPA improves the data rate and reduces the latency by using several novel techniques including shared channel transmission; multi-code CDMA transmission; fast Hybrid Automatic Repeat-request (HARQ); short Transmission Time Interval (TTI); and adaptive modulation and coding.

3.5.1 Features of HSDPA systems

HSDPA has introduced a new transport channel, the High-Speed Downlink Shared Channel ($HS - DSCH$), which uses channel-dependent scheduling to facilitate air interface channel-sharing between users. It includes three new physical layer channels which are the High Speed-Shared Control Channel ($HS - SCCH$); the High Speed-Dedicated Physical Control Channel ($HS - DPCCH$); and the High Speed-Physical Downlink Shared Channel ($HS - PDSCH$).

It is worth highlighting that in *HSDPA* systems, there is an application of multi-

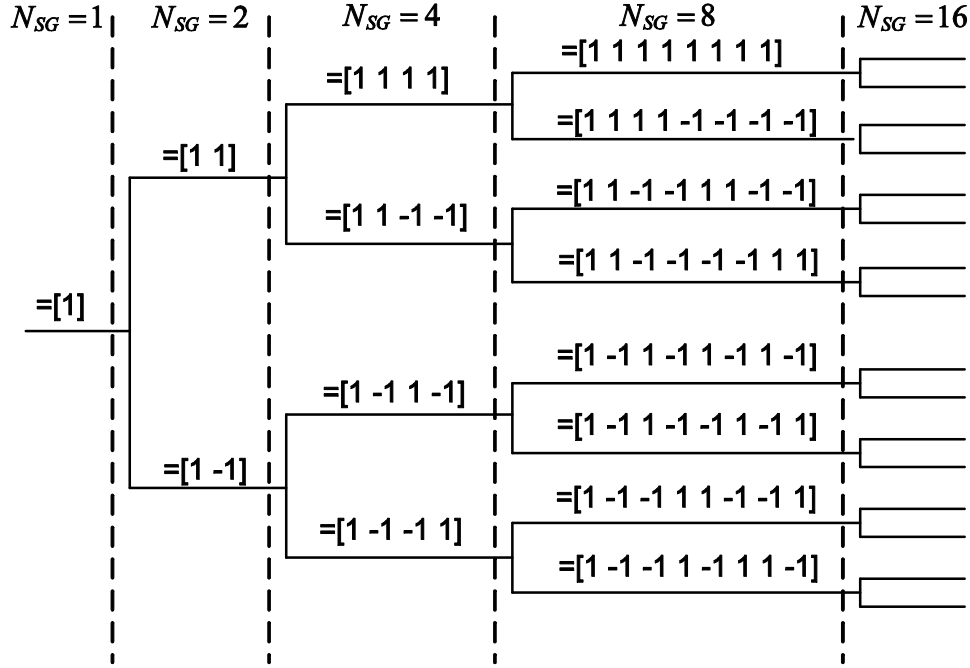


Figure 3.3: The Orthogonal Variable Spreading Factor (OVSF) Code Structure Tree

code CDMA communication. HSDPA systems allocate a total of $K = 5, 10$ or 15 spreading codes which all use the same QPSK and *MQAM* modulation schemes to one user in downlink transmissions. The adaptive modulation and coding technique is used to replace the power control in WCDMA. It adjusts the modulation scheme and coding rate based on the quality of signal at the input of the receiver, hence it provides different data rate options from 1.8Mbit/s to 14.4Mbit/s.

Another feature of the HSDPA system is the hybrid automatic repeat-request (*HARQ*) which is used to decrease latency and therefore improve the roundtrip time. The *HARQ* combines forward error-correcting coding and error detection. The error detection is based on the standard *ARQ* error-control approach, and it is achieved by adding the error correction bits (or redundant bits) into the frame to correct minor errors without retransmission.

3.5.2 Orthogonal Variable Spreading Factor (OVSF) Code

The OVSF code is applied in HSDPA systems to minimize the interference between users in underload conditions. In HSDPA systems, each user is allocated one or more unique and different OVSF codes for multi-code CDMA transmissions. The OVSF is a set of orthogonal spreading codes derived from a binary tree structure

which reflects the construction of Hadamard matrices.

$$C_{i+1}^{N_{SG}} = \begin{cases} C_i^{N_{SG}} C_i^{N_{SG}} & \text{if } x_i = 0 \\ C_i^{N_{SG}} (-C_i^{N_{SG}}) & \text{if } x_i = 1 \end{cases} \quad (3.15)$$

where N_{SG} is the spreading factor which is the length of spreading code and $N_{SG} = 2^m$, and i is the code index for $i = 0, 1 \dots m - 1$. There are two branches of binary sequence. $x_i = 0$ means the upper branch of the binary sequence and $x_i = 1$ means the lower branch of the binary sequence. $C_0^1 = [1]$ is defined as the root of the OVFS code tree. An example of an OVFS code tree is given in Figure 3.3. The OVFS code should satisfy the following condition to be orthogonal and therefore minimize the interference between users.

$$C_i^{N_{SG}} (C_j^{N_{SG}})^H = 1 \quad (3.16)$$

where $i, j \in [0, 1 \dots m - 1]$ and $i \neq j$.

3.5.3 Signal Processing in HSDPA Systems

In this section, a simple HSDPA system structure is designed to present the whole signal processing (Figure 3.4). The signal processing is divided into three stages which include signal processing at the transmitter, the radio channel, and the receiver as outlined below.

3.5.3.1 Signal Processing at the Transmitter

The source generates one or more binary data frames and sends them to the buffer. The total length of the frame is defined as $\sum_{k=1}^K dt_k$, where K is total number of channels used and dt_k indicates the number of data bits transmitted in the k^{th} subchannel. This data stream is then passed through a Series-to-Parallel (SIPO) converter and divided into multiple parallel sub-streams. These parallel data streams are then fed into Adaptive Modulation and Coding (AMC) units.

The first part of the AMC (or otherwise known as the rate control) unit is the channel encoder (e.g. the Turbo encoder), which adds redundancy bits into original data bits in order to recover the signal at the receiver. There is an assumption here that the coding rate is r_{coded} , and therefore the number of bits after coding is $\frac{dt_k}{r_{coded}}$ in the k^{th} subchannel. The coded binary bit stream is then sent into one modulation unit (e.g. $MQAM$) and mapped into symbols b_k with b_p bits per symbol in the k^{th} subchannel.

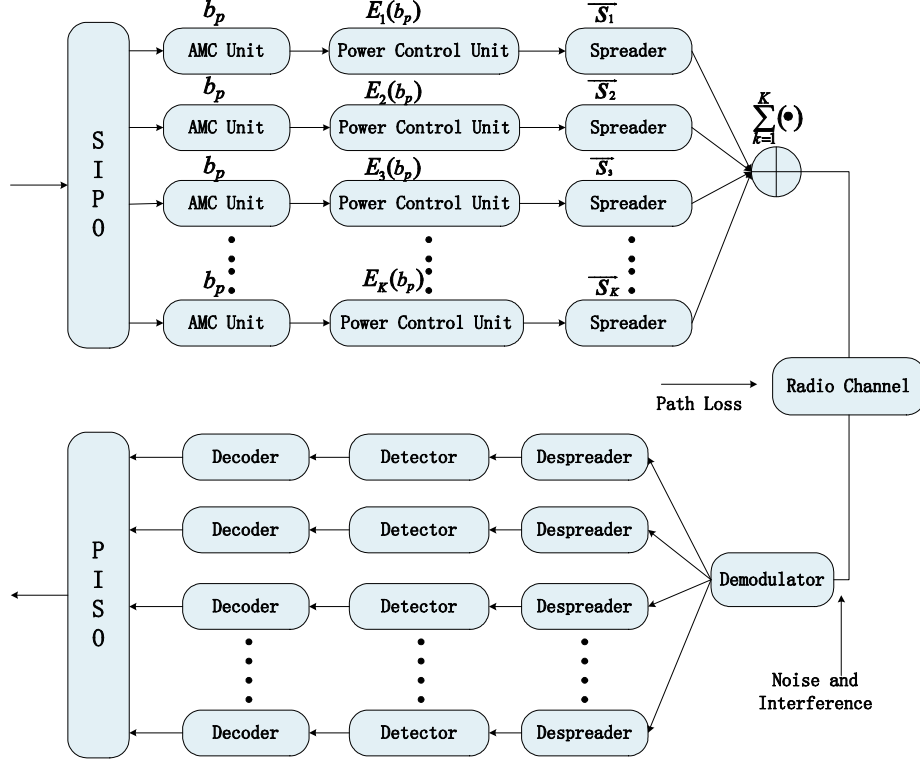


Figure 3.4: HSDPA System Structure

After modulation and coding, signal streams are sent into power (or amplitude) control units and allocated with power p_k (amplitude $A_k = \sqrt{p_k}$) in the k^{th} sub-channel, where the total power available $P_T = \sum_{k=1}^K p_k$ is always constrained. After allocating the power, the signals are presented as $\sqrt{p_k}b_k$ for $k = 1, 2 \dots K$, and the related energy required to transmit one symbol (which has b_p bits) is $E(b_p) = p_k T_s$, where T_s is the symbol rate and $T_{symbol} = N_{SG} T_c$.

Following on from that, signals are sent into spreader units to multiply with a highly significant sequence known as spreading codes. The assumption here is that a Direct Sequence Spread Spectrum (DSSS) system is at work in which each parallel channel uses specific spreading codes which length equals to N_{SG} , e.g., $\vec{s}_k = [s_{0,k}, \dots, s_{(N_{SG}-1),k}]^T$. These codes are transmitted with a chip rate $\frac{1}{T_c}$ and a unit energy. The signals, which pass through the spreader, are presented as $\sqrt{p_k}b_k s_k$. Finally, signals from all channels are summed up ($\sum_{k=1}^K \sqrt{p_k}b_k s_k$) and launched from the antenna with the same carrier frequency f .

3.5.3.2 Signal Processing in the Radio Channel

Once the signal processing is completed at the transmitter, signals are transmitted through the radio channel. In section 2.2.1, there is a discussion on the different wireless propagation models available for transmissions where it is assumed that the wireless propagation channel is a large-scale radio channel with no fading and which is non-frequency selective. The received amplitude of signal is given as $A_k = \sqrt{h_k p_k}$ for the k^{th} subchannel, where h_k is the path loss (or channel gain) for the k^{th} subchannel. The path loss is assumed to be known at the receiver due to the channel estimation. At the input of the receiver, received signals are corrupted by the noise and MAI, and are presented as $r_k = \sqrt{h_k p_k} b_k s_k + I_{MAI} + I_n$, where I_{MAI} is the MAI factor and I_n is the noise.

3.5.3.3 Signal Processing at the Receiver

When signals are received successfully, the carrier frequency, f , is removed from the received signals. Parallel signals are then despread through despreaders with the filter coefficient, w_k , for the k^{th} channel, and the corresponding outputs are presented as $\sqrt{h_k p_k} b_k (w_k^H s_k) + w_k^H \partial$. After despreading, parallel signals are demapped from symbols to bit sequences $\sum_{k=1}^K \hat{b}_k$ through detectors. It then follows that the binary sequences are fed into parallel decoders for estimating the original data sequences. Next, the estimated multiple data sequences, $\sum_{k=1}^K \hat{dt}_k$, are passed through a Parallel-to-Serial converter (PISO). Through comparing the recovered data sequences, $\sum_{k=1}^K \hat{dt}_k$, with original data sequences, $\sum_{k=1}^K dt_k$, the BER value is measured at the output of receiver.

3.5.4 Femtocell HSDPA Network

Another implementation of the HSDPA system is the Femtocell. Femtocell (also known as the home base station) is a short-range, low-cost and low-power cellular communication implementation. It achieves better indoor capacity and coverage from a small cell size. Femtocell is installed at home and in the office, and it connects to the service operator networks through a broadband gateway over the internet. In [57] and [58], an overview of the concept and development of the Femtocell is given. However, the high level modulation scheme, for example the 64QAM, does not perform very well in indoor environment due to the strong multipath effect and wall absorption. With regard to this problem, reference [59] evaluates the performance

of different modulation schemes (such as the QPSK, 16QAM and 64QAM) with the HSDPA systems in Femtocell indoor networks. The simulation results identify that only the equalizer receiver benefits from the 64QAM. From the operator's point of view, reference [60] analyses the Femtocell network architecture and signalling protocol options at implementation. Since 2009, a number of operators, such as Vodafone, have deployed the Femtocell station and also launched the related services.

3.6 Energy Utilization in HSDPA Systems

3.6.1 Problem Addressed: Residual Energy

In the previous section, a review of the HSDPA system was given. However, there are some limitations that ought to be looked at in existing HSDPA systems. One such constraint in the HSDPA system is that there are only three options available as far as the use of channels for transmissions is concerned, the available options are $K = 5, 10$, or 15 . In addition, all parallel channels are assigned the same modulation scheme, e.g. *QPSK* and *16QAM*. The received *SNR*, which satisfies the low level modulation scheme requirement but does not satisfy the next level modulation scheme requirement, will cause energy waste. All of these result in inefficient energy utilization and residual energy, which is the remaining and unused energy within the system. It is a problem that became the driving force for this section of the thesis, which focuses on the residual energy and related rate allocation in HSDPA systems.

3.6.2 Equal Energy and Rate Allocation in HSDPA Systems

The HSDPA system employs equal energy allocation to load equal bit rate in each channel with orthogonal multiple spreading codes. In a multi-code CDMA transmission, the source node allocates spreading codes into K parallel channels while $K \leq N_{SG}$. In each sub-channel, a given number of bits are modulated into symbols through modulation scheme, such as the *16QAM*. The bit rates for a multi-code CDMA system are usually taken from a set such that $b_k \in \{b_p | p = 0, \dots, P\}$ where p is the bit rate (or modulation) level, and $b_p = \beta p$ for $p = 0, \dots, P$ are the allowed discrete rates, where the granularity, β , is the minimum bit information available to be transmitted and it is defined as $\beta = b_{p+1} - b_p$ for $p = 0, \dots, (P - 1)$. The granularity is assumed to have a constant value when the modulation and coding scheme

are fixed (e.g. $\beta = 2$ is for uncoded *QPSK* and *MQAM* modulation schemes). Due to the fact that the bit rate allocated in all channels of the HSDPA system is the same, the required energy $E(b_p)$ for each channel $k = 1, \dots, K$ is related to the bit rate b_p for $p = 0, \dots, (P - 1)$ when the two-dimensional modulation scheme, such as *QPSK* and *MQAM*, is used.

$$E(b_p) = \frac{2\Gamma\sigma^2}{10^{\frac{-h}{10}}}(2^{b_p} - 1) \quad (3.17)$$

Based on the channel conditions, the HSDPA system applies the link adaptation (also known as adaptive modulation and coding) approach to allocate the equal bit rate across all parallel channels and to provide a wide range of discrete bit rates in order to satisfy the given QoS requirements of users. In order to achieve equal bit rate allocation, equal energy ($E_k = \frac{E_T}{K}$ for $k = 1, \dots, K$, where E_T is the total energy available at the transmitter) is assigned to all parallel channels. It is assumed that there is no fading and that the channel is non frequency-selective, hence the SNR measured at the input of each sub-receiver is the same, e.g. $SNR_k = \frac{SNR_T}{K}$ for $k = 1, \dots, K$. Due to the fact that the energy required to transmit b_p bits at each sub-transmitter, $E(b_p)$, satisfies $E(b_p) \leq \frac{E_T}{K}$ in HSDPA systems, the related SNR at each sub-receiver is also satisfied at $SNR(b_p) \leq \frac{SNR_T}{K}$ and is demonstrated as follows.

$$SNR(b_p) = \frac{10^{\frac{-h}{10}} E(b_p)}{2\sigma^2} \quad (3.18)$$

Based on the equations (2.7) and (3.17), the bit rate b_p achieved at each sub-channel is related to the SNR_T as outlined.

$$b_p = \log_2\left(\frac{SNR(b_p)}{\Gamma} + 1\right) \quad (3.19)$$

$$b_p \leq \log_2\left(\frac{SNR_T}{\Gamma K} + 1\right) \quad (3.20)$$

Where the bit rate levels are $b_p \in [2, 4, 6]$ and $\beta = 2$ for uncoded *QPSK* and *MQAM* transmissions in HSDPA systems. In a symbol period of transmission, the total bit rate, B , achieved in *HSDPA* systems is

$$B = \sum_{k=1}^K b_k = K b_k \quad (3.21)$$

The data rate r_{data} is then identified by using the following equation for the bit rate achieved in HSDPA systems.

$$r_{data} = \frac{BW}{N_{SG}} B \quad (3.22)$$

Where BW is the bandwidth used in HSDPA systems.

However, the equal energy and bit rate allocation results in inefficient resource utilization in HSDPA systems. The following section will therefore explore the related energy utilization and will formulate the problem of HSDPA systems through studying residual energy.

3.6.3 Residual Energy in HSDPA Systems

In the previous section, the equal energy allocation for all parallel channels in HSDPA systems was discussed. Given the total amount of energy, E_T , at the transmitter, the maximum bit rate achieved in each sub-channel is b_p . However, the equal allocated energy $\frac{E_T}{K}$ in each subchannel is not able to support a high bit rate level, e.g. $p + 1$, and it also satisfies $\frac{E_T}{K} \leq E(b_{p+1})$. On this basis, the following relationship between the total available energy E_T at the transmitter and the energy required for transmitting both b_p and b_{p+1} in HSDPA systems is given as

$$\begin{aligned} KE(b_p) &\leq E_T < KE(b_{p+1}) \\ KE(b_p) &\leq E_T < KE(b_p + \beta) \end{aligned} \quad (3.23)$$

It is at this stage that the incremental energy [61] in HSDPA systems ought to be considered. The incremental energy is the extra energy required to upgrade a given bit rate (e.g. b_p) to the next high-level bit rate (e.g. b_{p+1}). Based on the equation (3.17), the incremental energy required to transmit β extra bits from p level bit rate to $p + 1$ level bit rate is the difference between $E(b_p + \beta)$ and $E(b_p)$ as follows.

$$\begin{aligned} e(b_p + \beta) &= E(b_p + \beta) - E(b_p) \\ &= \frac{2\Gamma\sigma^2}{10^{\frac{-h}{10}}}(2^\beta - 1)2^{b_p} \end{aligned} \quad (3.24)$$

Based on the equations (3.23) and (3.24), there exists a total residual energy, e_R , which is the remaining amount of energy not to be utilized when the maximum bit rate is achieved. In a HSDPA system, the residual energy is upper-bounded by the incremental energy in each channel multiplied by the number of parallel subchannels as follows.

$$\begin{aligned} 0 &\leq E_T - KE(b_p) < K(E(b_p + \beta) - E(b_p)) \\ 0 &\leq e_R < Ke(b_p + \beta) \end{aligned} \quad (3.25)$$

Based on the equation (3.25), when using an equal energy and rate allocation approach in HSDPA systems, the residual energy, e_R , results in an inefficient usage

PHY Layer Parameters	
<i>Temperture (Kelvin) (T_0)</i>	290
<i>Noise Factor (NF)</i>	10
<i>Access Scheme</i>	<i>DSSS/CDMA</i>
<i>Carry Frequency (f)</i>	<i>2.4GHz</i>
<i>Spreading Factor (N_{SG})</i>	16
<i>Transmission Power (P_T)</i>	10dBm
<i>Received Packet Threshold</i>	$-75dBm$
<i>Carrier Sensitivity Threshold</i>	$-87dBm$

Table 3.2: PHY Layer Parameters

of the limited available energy and is upper-bounded by an incremental energy, $Ke(b_p + \beta)$. Accordingly, the problem in HSDPA systems can be expressed as the residual energy, e_R , will usually result in inefficient energy utilization, therefore reducing the overall data rate in HSDPA systems.

With this in mind, the following section will therefore examine the performance of the HSDPA system with the aim of identifying the root of the inefficient energy utilization problem in HSDPA systems.

3.6.4 Performance Evaluation

3.6.4.1 Bit Rate Allocation in HSDPA Systems

The parameters and related values used in this chapter for all results are given in table 3.2. Figure 3.5 shows the system capacity and discrete bit rate achieved in a HSDPA system. The black line indicates the system capacity of multi-code CDMA where 15 parallel channels are used. Other lines with different colors show the discrete bit rate achieved in HSDPA systems, where multiple modulation schemes (i.e. QPSK ($b_p = 2$), 16QAM ($b_p = 4$), and 64QAM ($b_p = 6$)) and parallel channels ($K = [5, 10, 15]$) are applied to achieve the maximum discrete bit rate.

This result shows that the bit rate displays a step-wise characteristic, and there is a large gap between the system capacity and bit rate achieved in HSDPA systems at the same SNR, especially when the bit rate increases. It is observed that at certain bit rates that are up to a 6 dB increase in the total SNR do not provide any increase for the bit rate. In order to avoid a bit rate reduction, it is important that the bit rate increases continuously with the increase of total received SNR.

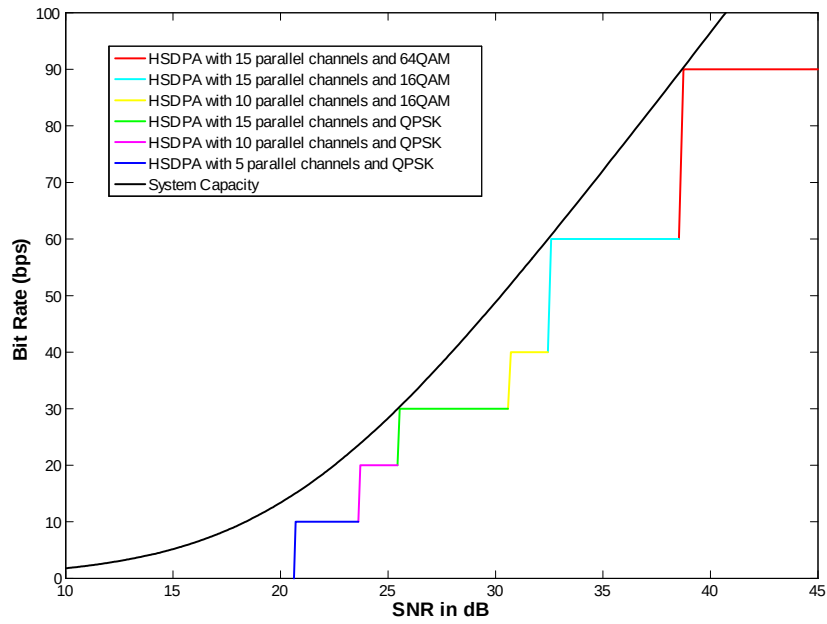


Figure 3.5: System capacity and bit rate achieved in HSDPA systems in different SNR zones

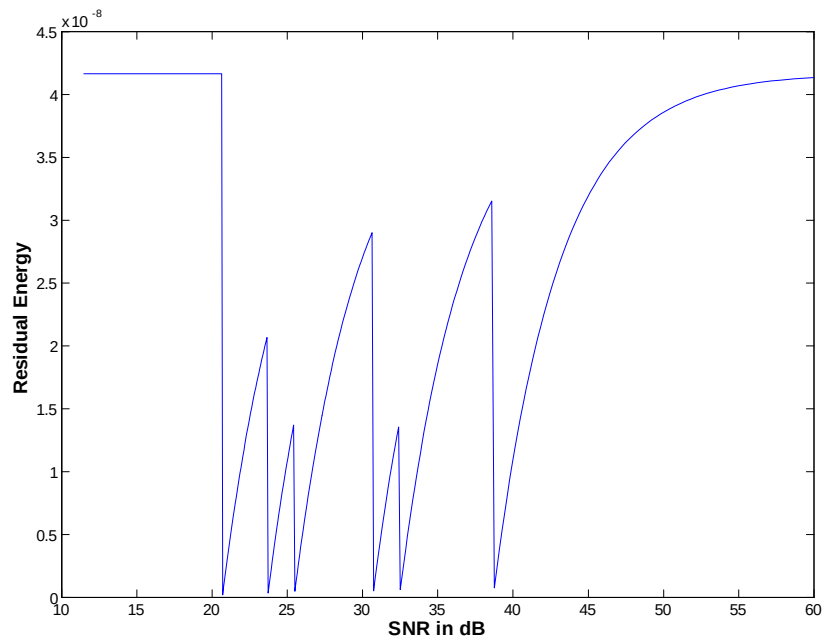


Figure 3.6: Residual Energy as a Function of SNR in HSDPA Systems

3.6.4.2 Residual Energy Allocation in HSDPA Systems

Figure 3.6 shows the residual energy as a function of the SNR in HSDPA systems. With an increase of SNR, all available energy, E_T , remains as the residual energy and is not utilized at all until the SNR is strong enough to support the minimum achievable bit rate in HSDPA systems, e.g. 10bps. As such, it could be said that as long as transmission power used at the transmitter is constant, then the shorter the distance between the transmitter and the receiver, the larger the received SNR will be.

On analysis of the results, two problems emerge, which both result in inefficient energy utilization in HSDPA systems. The first problem is that there are only three available options ($K = 5, 10$ or 15) for the parallel channels that can be used for transmissions. The second problem is that only one modulation scheme is applied to all parallel channels in a transmission. Taking these two problems into consideration and with the aim of resolving the issue of the poor efficiency of energy utilization and also to improve the data rate in HSDPA systems, two resource allocation approaches have been developed: the channel adaptive approach and the two-group allocation.

3.7 Channel Adaptive HSDPA Systems

One constraint in HSDPA systems is the restricted number of channels used for transmissions in terms of there being only three options available, $K = 5, 10$ or 15 , which results in inefficient energy utilization. In order to solve this problem, a channel adaptive approach is developed in HSDPA systems to ease this constraint to efficiently utilize the energy. In the following section, the rate allocation and residual energy in channel adaptive HSDPA systems will be studied.

3.7.1 Rate Allocation in Channel Adaptive HSDPA Systems

The total bit rate achieved in channel adaptive HSDPA systems, B_{CA} , simply equals to the number of parallel channels multiplied by the bit rate achieved in each sub-channel as follows.

$$B_{CA} = nb_p \quad (3.26)$$

Where $n = 1, 2, \dots, K$.

As the same modulation scheme is applied to all subchannels, based on the equation (3.19), the SNR required to transmit b_p at each subchannel is given as follows.

$$SNR_{CA}(b_p) = \Gamma (2^{b_p} - 1) \quad (3.27)$$

Once the total received SNR at the receiver is known, the number of parallel channels used for transmissions is obtained as follows.

$$\begin{aligned} n &= \left\lfloor \frac{SNR_T}{SNR_{CA}(b_p)} \right\rfloor \\ n &= \left\lfloor \frac{SNR_T}{\Gamma (2^{b_{p+1}} - 1)} \right\rfloor \end{aligned} \quad (3.28)$$

Based on the equation (3.26) and (3.28), the total bit rate achieved in channel adaptive HSDPA systems is given.

$$B_{CA} = b_p \left\lfloor \frac{SNR_T}{\Gamma (2^{b_{p+1}} - 1)} \right\rfloor \quad (3.29)$$

3.7.2 Residual Energy in Channel Adaptive HSDPA Systems

In channel adaptive HSDPA systems, the total available energy, E_T , falls between two adjacent total energies utilized for n and $n + 1$ parallel channels as follows.

$$nE(b_p) \leq E_T < (n + 1)E(b_p) \quad (3.30)$$

Based on this, the total residual energy, e_R^{CA} , in channel adaptive HSDPA systems is given below.

$$\begin{aligned} 0 &\leq E_T - nE(b_p) < (n + 1)E(b_p) - nE(b_p) \\ 0 &\leq e_R^{CA} < E(b_p) \end{aligned} \quad (3.31)$$

Based on the equation (3.31), the residual energy upper bound in channel adaptive HSDPA systems is the energy required to transmit b_p in one channel, i.e. $E(b_p)$,

In the next section, the performance of channel adaptive HSDPA systems will be examined in order to identify the enhancement in both energy utilization and bit rate in HSDPA systems.

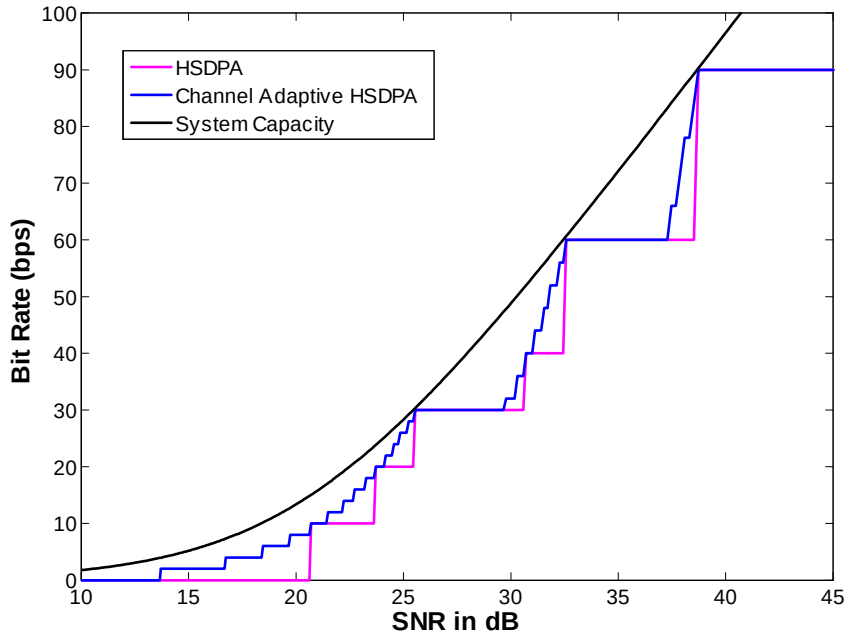


Figure 3.7: System capacity and bit rate achieved in HSDPA and Channel Adaptive HSDPA systems

3.7.3 Performance Evaluation

3.7.3.1 Bit Rate Allocation in Channel Adaptive HSDPA Systems

Figure 3.7 demonstrates the system capacity and the bit rate achieved in both the original HSDPA system and the channel adaptive HSDPA system, respectively. Compared with the original HSDPA system, the result shows that channel adaptive HSDPA improves the bit rate. Due to the fact that there are more channel options available for transmissions in HSDPA systems, the residual energy upper bound decreases so that the required amount of energy is sufficient to transmit b_p in one channel, $E(b_p)$. Within the low SNR range (such as $\text{SNR} < 26\text{dB}$), the bit rate achieved in channel adaptive HSDPA systems, in particular, is closed to the system capacity. However, the channel adaptive approach does not significantly improve the bit rate within the high SNR range (such as $26\text{dB} < \text{SNR} < 39\text{dB}$).

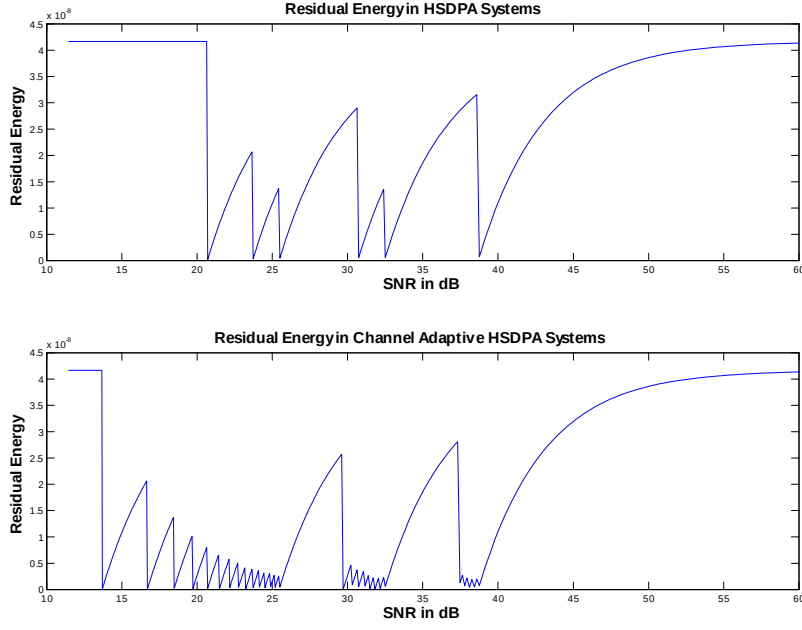


Figure 3.8: Residual Energy as a Function in HSDPA and Channel Adaptive HSDPA systems

3.7.3.2 Residual Energy Allocation in Channel Adaptive HSDPA Systems

Figure 3.8 compares the residual energy in both the original and channel adaptive HSDPA systems. From the result, it is clear that the channel adaptive approach starts to reduce by a large amount of residual energy within the low SNR range. Nevertheless, the energy utilization is still not efficient enough within the high SNR range (e.g. 26dB–39dB) as a large amount of residual energy still remains in HSDPA systems. In Figure 3.8, when the total SNR is strong enough to achieve the minimum bit rate transmission in channel adaptive HSDPA systems (i.e. 14dB), the residual energy starts to decrease. However, when the total energy falls between two adjacent total energies utilized for n and $n + 1$ parallel channels, the residual energy (Figure 3.8) will rise with the increase of the total SNR, and it will fall down until it satisfies the energy requirement for $n + 1$ parallel channels. In addition, when the total energy falls between two adjacent modulation levels (e.g. level p and level $p + 1$) for all parallel channels, the residual energy (Figure 3.8) will also increase and fall down until it satisfies the energy requirement for the higher level modulation (e.g. level $p + 1$).

Based on the analysis of results, the channel adaptive approach is identified to efficiently utilize energy within low SNR range and therefore enhances the bit rate in HSDPA systems. However, there is still a large amount of residual energy existing within high SNR range in channel adaptive HSDPA systems.

3.8 Two-group Rate Allocation HSDPA Systems

As there is a very limited amount of total available energy for transmissions, in order to achieve more efficient energy utilization in HSDPA systems, a two-group resource allocation approach is developed with a view to reduce the residual energy upper bound in HSDPA systems. Known as the total SNR at the output of the receiver, instead of allocating the bit rate equally across all parallel channels, two different bit rates are allocated between two groups of channels in order to enhance the achievable bit rate in HSDPA systems. One advantage of the two-group allocation approach is that it is simple to implement in existing HSDPA systems and does not require complex infrastructure being upgraded. In addition, two-group allocation does not require huge computational complexity to process signals.

In the following sections, the rate allocation and the residual energy in two-group HSDPA systems are studied and examined respectively.

3.8.1 Rate Allocation in Two-group HSDPA Systems

Here, it is assumed that all $K = N - 1$ optimal spreading codes are available for any nodes in networks. These spreading codes are allocated into two groups of channels (Figure 3.9): where m channels make up the first group with a high bit rate (e.g. b_{p+1}), and $(K - m)$ channels make up the second group with a low bit rate (e.g. b_p).

In a symbol period, the bit rate b_{p+1} with a modulation level of $p + 1$ ($b_{p+1} = b_p + \beta$) is allocated to all m channels in the first group, and the bit rate b_p with a modulation level of p ($b_p = \beta \cdot p$, where $p \in [1, 2, 3]$ and $\beta = 2$) is allocated to all remaining $(K - m)$ channels in the second group. It is further assumed that the available bit rate b_k allocated in the k^{th} channel is related to the *MQAM* modulation scheme and is drawn from a given set such that $b_k \in \{b_p | p = 0, \dots, P\}$. The total bit rate B_{TG} achieved by two-group allocation HSDPA systems is demonstrated below.

$$B_{TG} = mb_{p+1} + (K - m)b_p \quad (3.32)$$

The transmission energy E_k is adjusted for each channel $k = 1, \dots, K$ such that

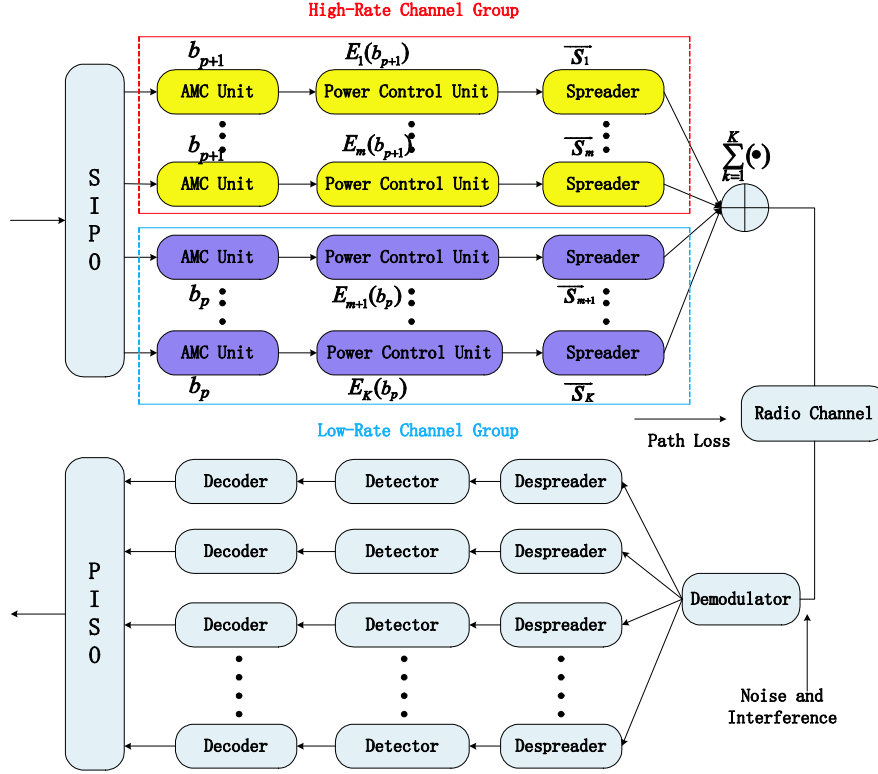


Figure 3.9: Two-Group HSDPA System

m channels have the same transmission energy E_{p+1} and the remaining $(K - m)$ channels have the same transmission energy E_p .

It is assumed that each node transmits at the same maximum total energy E_T for a distance d between the transmitter and the receiver when having a transmission path loss, h , given by equation (2.4), the total number of bits B_{TG} given by equation (3.32) is maximized as best as possible where the total required energy, E_t will be less than the available total energy, E_T as exemplified below.

$$\max_{b_p, m} (mb_{p+1} + (K - m)b_p) \quad (3.33)$$

for $0 \leq p \leq P$ and $0 \leq m \leq K - 1$

Given E_T at the transmitter, the system gap value Γ , propagation path loss h calculated from equation (2.4) and noise variance σ^2 , the total received SNR_T is given by the equation (2.7). Based on the Shannon Theory, the bit rate maximization problem is related to the total received SNR_T .

subject to

$$0 \leq SNR_T - \Gamma(m(2^{b_{p+1}} - 1) + (K - m)(2^{b_p} - 1)) < \Gamma(2^{b_{p+1}} - 2^{b_p}) \quad (3.34)$$

Based on equations (3.33) and (3.34), the maximization problem of two-group allocation works to find the bit rate level as represented by p , and the related number of channels for each group, indicated as $K - m$. In order to identify the bit rate level and related number of channels, the low level bit rate b is equally loaded across all parallel channels. Here, the problem is to find the modulation level p of bit rate $b_p = p\beta$ for $p = 0, \dots, P$ through equal SNR allocation as follows.

$$p = \min \left(\left\lfloor \frac{\log_2 \left(\frac{10^{-h}}{2KN_0\Gamma} E_T + 1 \right)}{\beta} \right\rfloor, P \right) \quad (3.35)$$

$$p = \min \left(\left\lfloor \frac{\log_2 \left(\frac{SNR_T}{K\Gamma} + 1 \right)}{\beta} \right\rfloor, P \right) \quad (3.36)$$

From equations (3.34) and (3.36), when trying to find the modulation level p for low level bit rate b_p , the number of channels, m , can be obtained by using the following equation.

$$m = \left\lfloor \frac{E_T - KE(b_p)}{e(b_p)} \right\rfloor \quad (3.37)$$

$$m = \left\lfloor \frac{\frac{SNR_T}{\Gamma} - K(2^{b_p} - 1)}{2^{b_{p+1}} - 2^{b_p}} \right\rfloor \quad (3.38)$$

Once p and m are identified using equation (3.36) and (3.38), the total bit rate, B_{TG} , achieved in two-group allocation HSDPA systems can be calculated from the equation (3.32).

3.8.2 Residual Energy in Two-group HSDPA Systems

Known as the achievable bit rate in two-group HSDPA systems, the related residual energy utilization is studied in this section. In order to achieve the proposed bit rate in all parallel channels, the energy $E(b_{p+1}) = E(b_p + \beta)$ is allocated to each channel of the first group with m channels, and the energy $E(b_p)$ is allocated to each channel of the second group with a $(K - m)$ channel. The total available energy required for transmissions, E_t , is

$$\begin{aligned} E_t &= mE(b_p + \beta) + (K - m)E(b_p) \\ &= KE(b_p) + me(b_p + \beta) \end{aligned} \quad (3.39)$$

Based on the equation (3.39), the $m\beta$ additional bit rate is achieved by using extra incremental energy $me(b_p + \beta)$ from the residual energy in HSDPA systems. The total available energy at the transmitter falls between two scenarios as follows.

$$\begin{aligned}
mE(b_p + \beta) + (K - m)E(b_p) &\leq E_T < (m + 1)E(b_p + \beta) + (K - m - 1)E(b_p) \\
KE(b_p) + me(b_p + \beta) &\leq E_T < KE(b_p) + (m + 1)e(b_p + \beta)
\end{aligned} \tag{3.40}$$

In two-group HSDPA systems, the energy $E_T - KE(b_p)$ falls between two adjacent total incremental energies $me(b_p + \beta)$ and $(m + 1)e(b_p + \beta)$ as follows.

$$me(b_p + \beta) \leq E_T - KE(b_p) < (m + 1)e(b_p + \beta) \tag{3.41}$$

The total residual energy, e_R^{TG} , in two-group HSDPA systems is given.

$$\begin{aligned}
0 &\leq E_T - KE(b_p) - me(b_p + \beta) < (m + 1 - m)e(b_p + \beta) \\
0 &\leq E_T - E_t < e(b_p + \beta) \\
0 &\leq e_R^{TG} < e(b_p + \beta)
\end{aligned} \tag{3.42}$$

Given that each node transmits at the same E_T , equations (3.25) and (3.42) identify that the two-group allocation reduces the total residual energy upper bound from $Ke(b_p + \beta)$ (equation (3.25)) to $e(b_p + \beta)$ (equation (3.42)) in HSDPA systems, and the bit rate achieved in HSDPA systems increases by $m\beta$ as well.

In the next section, the performance of two-group HSDPA systems will be examined in order to identify the enhancement on both energy utilization and bit rate in HSDPA systems.

3.8.3 Performance Evaluation

3.8.3.1 Bit Rate Allocation in Two-group HSDPA Systems

Figure 3.10 demonstrates the bit rate against the total SNR curve for the proposed two group rate allocation method. According to this result, the bit rate achieved in two-group HSDPA systems increases continuously with the increase of total received SNR. Compared with the original HSDPA system, the proposed two-group rate allocation method provides up to 30bps enhancement at certain total SNR. Within the low SNR range (e.g. 14dB–26dB), the performance achieved by two-group HSDPA systems and channel adaptive HSDPA systems are the same. However, within the high SNR range (e.g. 26dB–39dB), the two-group allocation approach significantly outperforms the adaptive channel method in HSDPA systems and obviously improves the bit rate, as well as minimizing the gap between the system capacity and bit rate achieved in HSDPA systems.

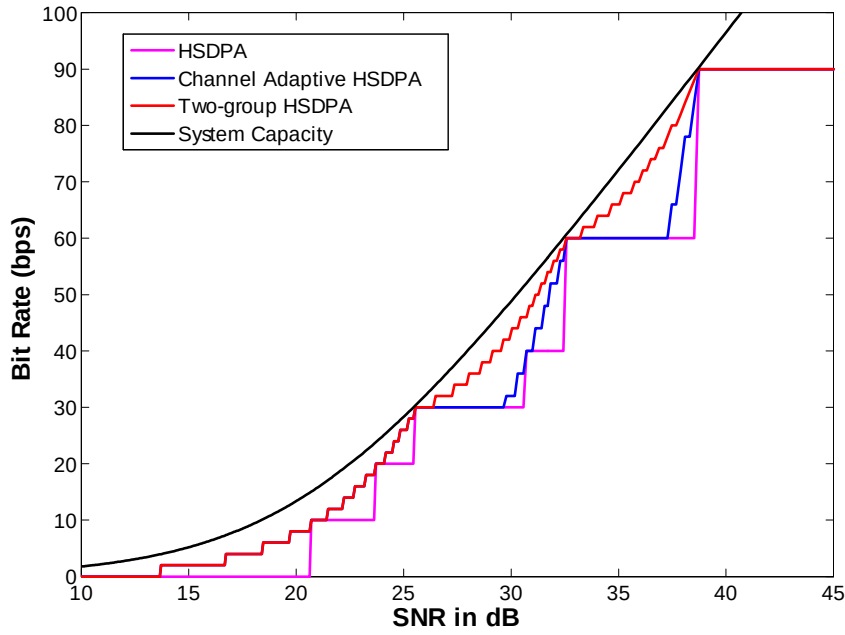


Figure 3.10: System capacity and bit rate achieved by HSDPA, Channel Adaptive HSDPA and Two-group HSDPA systems

3.8.3.2 Residual Energy Allocation in Two-group HSDPA Systems

In Figure 3.11, the residual energy existing in HSDPA systems, channel adaptive HSDPA systems and two-group HSDPA systems are compared. The remaining residual energy in both channel adaptive HSDPA and two-group HSDPA systems is the same within low SNR ranges (i.e. $\text{SNR} < 26\text{dB}$). However, the two-group allocation approach obviously reduces the residual energy in HSDPA systems within high SNR ranges, i.e. $26\text{dB} < \text{SNR} < 39\text{dB}$.

Figure 3.12 indicates that the average residual energy as a function of the SNR from 21dB to 39dB in HSDPA systems, channel adaptive HSDPA and two-group HSDPA systems. The average residual energy in two-group HSDPA systems is below 5×10^{-9} within all SNR ranges. The channel adaptive HSDPA system perform just as well as the two-group HSDPA system within part of the SNR range, as like with 21dB to 25dB and 30dB to 32dB. However, within part of the SNR range, for instance, with 26dB to 29dB and 33dB to 37dB, the channel adaptive HSDPA systems do not utilize the residual energy efficiently. Within the high SNR range, the ratio of average residual energy existing in two-group HSDPA systems to the average residual energy existing in the original HSDPA systems is up by 10%. The

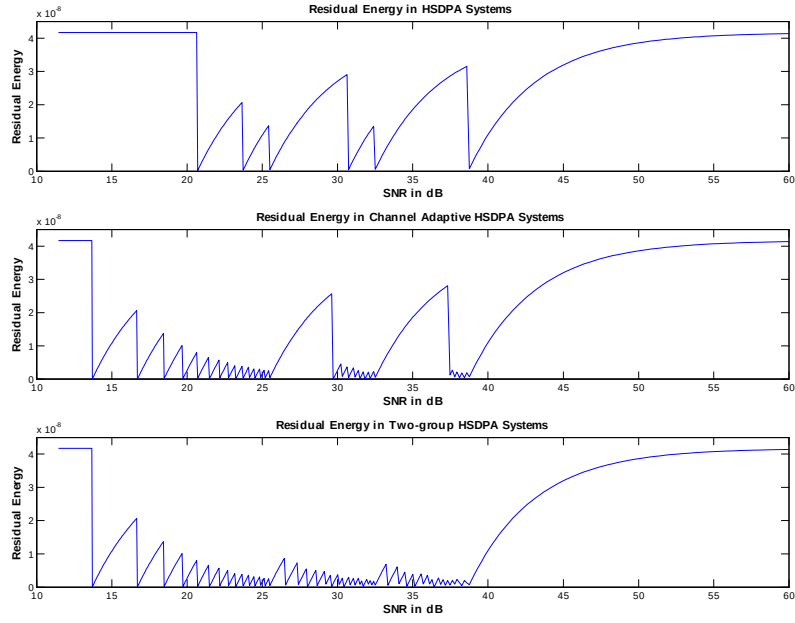


Figure 3.11: Residual Energy as a function of SNR in HSDPA, Channel Adaptive HSDPA and Two-group HSDPA Systems

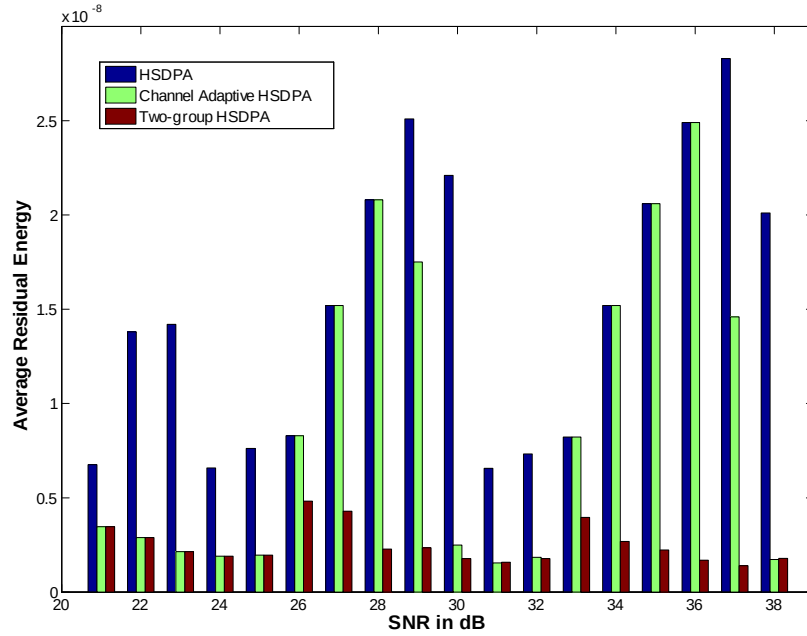


Figure 3.12: Average Residual Energy as a function of SNR in HSDPA, Channel Adaptive HSDPA and Two-group HSDPA Systems

result identifies that two-group allocation approach significantly reduces the residual energy and therefore provides an efficient energy utilization solution for the HSDPA system.

3.9 Conclusion

In this chapter, the multi-code CDMA and related resource allocation techniques (e.g. power allocation and optimal spreading code design) are studied. The chapter then focuses on the HSDPA system, which is an application of the multi-code CDMA. However, the energy is not efficiently utilized in HSDPA systems, and hence this results in a residual energy problem. In order to efficiently utilize energy and to improve the data rate in HSDPA systems, two approaches are developed, which are known as the channel adaptive method and the two-group allocation.

Based on the radio channel condition (or measured SNR at the receiver), the channel adaptive method adjusts the number of channels that can be used in HSDPA systems and two-group allocation, which in itself assigns the bits into two channel groups with different modulation schemes. The results identify that both the channel adaptive method and the two-group allocation reduce the residual energy in HSDPA systems and improve the data rate. Furthermore, both of these approaches are simple to implement within the existing HSDPA systems.

Chapter 4

Energy-aware Multipath Routing

4.1 Introduction

Unlike mobile cellular systems, wireless ad hoc networks do not operate from a centralized point of administration. Communication in ad hoc networks works to forward information from one node to another through a routing path, so that significantly more additional energy is consumed by the relay node. However energy is extremely limited in ad hoc networks, such as sensor networks. It is for this very reason, when taking into consideration the capacity of the network resource, which the resource allocation has been developed in such a way as to maximize the amount of transmitted information and to also extend the network life.

With regard to resource management in energy-constrained networks, many different aspects of this area have been closely studied, including work around energy-aware routing, MAC-aware routing, and multipath routing. It is suggested that energy-aware routing can be used to find the best low energy cost routing path based on different factors, such as the residual energy that remains in the battery, the energy drain rate and the battery sensitivity. MAC-aware routing aims to find a routing path that minimizes the chances of retransmission, and which will efficiently manage the channel access, thus reducing energy consumption. Multi-path routing is another area of significant interest as it offers three main advantages: fault-tolerance (also known as reliability of transmissions), load balancing and higher aggregate capacity. In this chapter, the energy-aware routing is examined alongside a further investigation into multipath routing and related load balancing.

4.2 Related Work

4.2.1 Energy-aware Routing Protocols

To date, many research studies have examined energy-aware routing protocols closely with the intention to uncover the best efficient energy utilization solutions that can, in turn, be used to maximize transmitted information bits, and which will extend the network lifetime.

Reference [62] offers a comprehensive study and review of such energy-aware routing protocols. In [63], a performance study on power-aware routing in ad hoc networks is presented in details, with a particular focus on the evaluation and comparison of the five main different energy-aware routing protocols, listed as follows: *AODV* [3]; Minimum Total Power Routing (*MTPR*) [64]; Minimum Battery Cost Routing (*MBCR*) [65]; Min-Max Battery Cost Routing (*MMBCR*) [65] and Conditional Max-Min Battery capacity Routing (*CMMBCR*) [66]. *MTPR* [64] selects the routing path with the minimum total transmission power consumption. This approach tends to select the routing path with the largest number of hops since the transmission power is inversely proportional to the distance. *MBCR* [65] designs the path metric by considering the sum of the inverse of the battery capacity for all intermediate nodes. However, the node, which has little remaining battery capacity, may be still chosen and therefore its related energy will diminish quickly. In order to solve this problem and provide fairness throughout the network, *MMBCR* was developed. *CMMBCR* was designed through combining both *MTPR* and *MMBCR*, and switching *MTPR* and *MMBCR* based on different conditions. In order to examine and evaluate these five protocols, an energy consumption model was built in [63]. In addition, reference [63] also proposes a novel enhancement to enable the flow admission in ad hoc networks. Reference [67] also gives an experiment-based evaluation on the performance of different energy-aware protocols: *MTPR*, *MBCR*, *MMBCR*, *CMMBCR*, *MDR* and Time Delay On-demand Routing (*TDOD*) [68].

Power-aware routing is another area of research interest in terms of energy utilization in ad hoc networks and extending the network lifetime. With this in mind, a Power-aware Routing Optimization (*PARO*) [69] was developed and with the aim to minimize the transmission power by using the power consumption as the cost function in routing metric. A destination-based Power-aware Source Routing (*PSR*) protocol [70] was also developed to calculate the link cost, which considered the transmission power and remaining battery energy at the destination node. The

routing path with the minimum cost is also selected at the destination node.

Since the power control is a major issue for CDMA transmissions, the power-aware or interference-aware routing protocols are exploited in CDMA ad hoc networks. A CDMA based MAC Protocol associated with power control known as Controlled Access CDMA (*CA-CDMA*) is proposed in [71] for mobile ad hoc networks. It takes into account the multiple access interference (MAI) in order to solve the near-far problem and therefore enhances the network performance. The information on path loss is obtained from control packets, e.g. RTS/CTS, and is used to dynamically bind the transmission power at the nodes. The required transmission power is adjusted for the data packet in order to achieve the interference-limited simultaneous transmissions for the neighbouring node of the receiver. The simulation results identify the significant improvement of the CA-CDMA over the IEEE 802.11. In [72], a cooperative approach is used to develop two interference-aware routing schemes through improving the interference distribution in CDMA ad hoc networks. It takes into account not only the link cost but also the potential interference caused by the neighbouring nodes on the route.

A minimum consumed energy routing algorithm design is discussed in [73]. Rather than applying the routing path discovery mechanism based on hops or delay, reference [73] develops a routing metric based on energy consumed at nodes and uses the routing path to provide the minimum total energy consumed. The link cost in the routing metric takes into account signal strength, battery levels at the nodes and power consumption for transmissions. However, it does not consider the remaining energy at nodes, which means that the energy in the intermediate nodes will be consumed quickly and this will result in some nodes dying quicker than others in networks.

In order to resolve this problem, the battery information and related residual energy should be taken into account for routing metric designs. In [74], an energy-aware and delay-aware routing algorithm called Energy and Delay-Constrained (*EDC*) routing, is proposed by considering residual energy and buffer information (e.g. current size of the queue) in cost function metric design for route discovery. It aims to extend the network lifetime and minimize the end-to-end delay. The residual energy information is used to more efficiently load traffic, and the buffer information is used to avoid the routes which have congested nodes. Another power-aware protocol known as Traffic load and lifetime Deviation based Power-aware Routing protocol (*TDPR*) [75] is developed in order to extend the overall network lifetime

as well. TDPR considers the battery residual energy, the transmission power, the traffic load at nodes, all intermediate nodes lifetimes, and the deviation between nodes lifetime on active routes for the route discovery process. The proposed node lifetime prediction function takes into account both the battery's residual energy and transmission power at intermediate nodes, to calculate the accurate lifetime of a given node. Route maintaining schemes can also re-establish a reliable path based on the node lifetimes and mobility.

E. Gelenbe developed a Power-aware Ad Hoc *CPN* (*AHCPN*) [76] as an extension of Cognitive Packet Networks (*CPN*) [77] by employing smart packets to exploit the use of unicasts and broadcasts for discovering routing paths. In order to find both a low-delay and energy-efficient routing path, a new routing metric is designed in the smart packet to consider both the energy remaining at the nodes and path delay. Packets choose nodes with more remaining battery energy based on the Path Availability probability. The simulation results in [76] show that *AHCPN* is able to distribute the network traffic in an efficient way and therefore extends the battery lifetime at nodes. In addition, *AHCPN* also dynamically discovers and maintains the routing path without having to make an immense number of broadcasts. .

In [78] [79], a new routing called Min-Max Battery Capacity Routing (*MMBC*) is developed, in which the routing metric is obtained through calculating the summation of the inverses of the remaining energy at the intermediate nodes along the routing path. Furthermore, a Min-Max algorithm is proposed to achieve fairness in terms of the use of energy of nodes. This is said to be achieved through the avoidance of the nodes with the least remaining energy as intermediate nodes for the routing path discovery.

By contrast an algorithm unlike *MMBC* was developed in [78] to select the routing path which consumes minimal energy and instead maximizes the minimal residual energy in networks. In [80], a routing algorithm called Maximum Residual Packet Capacity (*MRPC*) is put forward in order to extend the operational lifetime. This algorithm uses both the residual energy and the expected energy spent in the routing metric. *MRPC* considers the packet transmission energy as the link character which affects energy consumption and packet retransmission. The routing path, where the intermediate node has the smallest link cost, is considered to have the maximum lifetime.

Instead of only considering the routing path with the least energy consumption,

in [81] [82], the routing problem is formulated as a linear problem to maximize the network lifetime through minimizing the total energy consumption of a routing path. In regards to the routing path selection, the energy consumption between nodes is balanced in proportion to their energy reserves. The work in [81] solves this linear problem for a single destination case, and it is extended [82] to the multicommodity case which has a set of destinations.

Another concept called Energy Drain Rate for the energy-aware routing design is introduced in [83]. This energy drain rate concept is represented as the speed of energy consumption. In [83], a Minimum Drain Rate (MDR) algorithm is developed, and the energy drain rate is computed by an exponential weighted moving average method to give the estimated energy consumption. The link cost function of *MDR* is defined as the ratio of the residual battery energy to the drain rate at each node in a network.

4.2.2 MAC-aware Routing Protocols

MAC-aware routing is another popular research topic in terms of designing routing protocols. Studies tend to focus on the network contention and aims to find a routing path with high throughput or with a low probability of retransmission by efficiently managing the channel access. A survey [84] on MAC-aware routing is given by Alexander Ende to show the development in this area. Reference [85] proposes a routing metric called Expected Transmission Count (ETX) to find routing paths, which provide high throughput, in multi-hop wireless networks. It considers the effect of wireless link loss and transmission interference between successive links along a route as follows.

$$ETX = \frac{1}{p_{data} \times p_{ack}} \quad (4.1)$$

where p_{data} is the probability of successful data packet transmission, and p_{ack} is the probability of successful *ACK* packet transmission. A route *ETX* is the sum of the link *ETX*, e.g. $\sum_{i \in J} ETX_i$, where i is the link along the routing path J . For a given node pair, the routing path with the minimum route *ETX* provides the largest throughput. This metric is designed to avoid routing paths with lossy links. However, this approach is too simple to be applied in long path multi-hop networks and also does not consider the physical layer effect and contention at the MAC layer. In [86], another routing metric called Expected Transmission Time (*ETT*) is

developed based on the *ETX* concept, and it is defined as follows.

$$ETT = ETX \times \frac{L}{r_{data}} \quad (4.2)$$

where L is the data packet size and r_{data} is the data rate. The sum of *ETT* of all links along a given routing path is considered as the transmission delay. Given a routing path, the link with the same data rate is grouped and the related *ETT* is obtained for each group. The group (e.g. X) with the largest sum *ETT* is considered as the "bottleneck" group which provides the lowest data rate as a result of maximum transmission delay. For a routing path, the routing path throughput is normally dominated and limited by the "bottleneck" group. By introducing a tunable parameter $\theta \in [0, 1]$. Reference [86] offers a tradeoff solution between the throughput and transmission delay known as the *Weighted Cumulative ETT* (*WCETT*) and which can be described as follows.

$$WCETT = (1 - \theta) \sum_{i \in J} ETT_i + \theta X \quad (4.3)$$

In [87], another new routing metric called *Expected Data Rate* (*EDR*) is developed for accurately finding high-throughput paths in multi-hop ad hoc networks. A transmission interference model is designed by considering the IEEE 802.11 DCF. Three important observations are introduced: transmission contention degree as a function of packet loss, exponentially increased contention window size due to the packet loss, and possible concurrent transmissions over non-interfering links. Compared with the previous routing metric research, *EDR* is more accurate and relates to the existing MAC standard: IEEE 802.11 DCF. The expression of *EDR* of link k is given as follows.

$$EDR(k) = \frac{r_{\max}}{ETX(k) + I(k)} \quad (4.4)$$

where $r_{\max}$ is the ideal maximum data rate, $ETX(k)$ is the *ETX* of link k , and $I(k)$ is the total sum Transmission Contention Degree (*TCD*) of link k [87]. In the equation (4.4), $ETX(k)$ is used to account for the throughput reduction due to the packet loss, while $I(k)$ is used to account for the throughput reduction due to the time-sharing with the contending links. The expected routing path is the one with the minimum value of all links and *EDRs* along the routing path. Reference [89] develops an Expected Throughput (*ETP*) solution as an extension of *EDR* by considering the contention effect from other links along the routing path to the target link as follows.

$$ETP(k) = p_{data}^k p_{ack}^k r_{data}^k = \frac{p_{data}^k p_{ack}^k}{\sum_{j \in P, j \neq k} \frac{1}{r_j}} \quad (4.5)$$

where r_{data}^k is the expected data rate achieved at link k , r_j is the nominal bit rate of link j , P is the set of all links along given routing path that contend with link k .

However, there are some limitations to these MAC-aware routing matrices as mentioned earlier. Firstly, *ETX* is only related to the transmission loss. Although both *EDR* and *ETP* start to take into account the IEEE 802.11 MAC DCF, the models are still not accurate enough to give the network contention at the MAC layer. For example, in an ad hoc network the contention is not only resulted by links along the routing path. Secondly, only the transmission data rate is considered by these routing matrices, but not the throughput. Finally, these routing matrixes are based on the assumption that all links along the routing path have an equal number of transmission opportunities.

In [88], a MAC-aware Routing with Load Balancing (*MaLB*) routing algorithm is developed. Based on *ETP* [89], *MaLB* uses a greedy approach and takes into account the interaction between links to enhance the network throughput through avoiding congested links. *MaLB* also provides an optimal multi-hop association of nodes through offering a fair traffic distribution between the gateways in networks. The transmission delay for all nodes to its associated gateway is defined as the cost function as follows.

$$D(T) = \sum_i \frac{|T_i|^2}{ETP_{l_i}} \quad (4.6)$$

where $|T_i|$ is the number of nodes in the subtree with root i and ETP_{l_i} is the *ETP* at link l from node i . The routing algorithm aims to find a routing path which has the minimum cost function D . Compared with *ETX* [85], the simulation results identify that *MaLB* results in more than 60% throughput improvement.

In [90] [91], a MAC aware routing matrix called Medium Time Metric (*MTM*) is developed to select a routing path with high throughput by considering the medium time at the MAC layer. The multi-hop routing path throughput is defined as follows.

$$S = \frac{L}{\sum_{i=1}^H T_i^m} \quad (4.7)$$

where T_i^m is the medium time of the i^{th} link, and H is the maximum number of hops for a routing path.

Through assigning a weight, which is based on the medium time defined in [90] [91], to each link in the network, a routing matrix is designed and the routing path with the minimum sum of the medium time of all links along the routing path is selected. Based on this route selection mechanism, a routing path with the highest

throughput is found. However, one problem for reactive protocols is the fact that nodes discard duplicate *RREQ* messages. In order to solve this problem, reference [92] proposes that every node keeps track not only of their 1-hop neighbours, but also their 2-hop neighbours. In addition, a multi-rate route discovery procedure is also developed in [92] based on the *MTM* approach [90] [91].

In relation to the MAC-aware routing in multi-rate multi-hop ad hoc networks, reference [93] studies a novel topology control scheme, which uses multiple high rate links to replace low rate links, as long as the indirect communication condition is satisfied as follows.

$$T_{ac}^m + T_{cb}^m + \eta < T_{ab}^m \quad (4.8)$$

where T_{ac}^m is the medium time from node a to node c , and η is the increasing contention due to indirect communication. Reference [93] also suggests a method to calculate the η value and identifies that its value is small.

4.2.3 Multipath Routing Protocols

As multiple routing paths provide a number of benefits on fault-tolerance, load balancing and higher aggregate capacity in networks, the research on multipath routing protocols has become a hot topic for routing studies and analysis.

A multipath routing algorithm is proposed for the *AODV* protocol in [94], and it utilizes the mesh structure to provide multiple routes without the extra overhead. The alternate routing path is used only when the primary route is broken. The simulation results in [94] identifies that this algorithm provides the robust and enhanced reliability for the existing *AODV* protocol. A new routing protocol known as the Caching and Multipath (*CHAMP*) Routing Protocol is proposed in [95] by using cooperative packet caching and shortest multipath routing in order to reduce the packet loss that results from the route breakage in high density and mobile ad hoc networks. Each node maintains a small buffer for caching data packets and retransmission of the data when breakdown occurs. A multipath routing path discovery mechanism is also developed to select the non-disjoint shortest multipath routes at equal distance. The mechanism identifies two routing paths with optimal solutions in terms of forwarding packets to the destination, as well identification of further routes that do not gain any benefits. The simulation results show that *CHAMP* outperforms *AODV* and *DSR*, especially in dynamic and high load ad hoc networks.

The load balancing is another point of focus with the multipath routing protocol study. In [96], a multiple routing protocol known as Multipath Source Routing (*MSR*) is proposed by extending the *DSR* route discovery and route maintenance mechanism. Based on the measurement, a load distribution scheme between multiple routes is also developed. The simulation results show that *MSR* improves the throughput and the end-to-end delay, but decreases the data rate with few overheads. A dynamic load-aware based load balancing routing (*DLBL*) algorithm is proposed in [97]. *DLBL* balances load and reduces the congestion during route discovery by avoiding congested routes to shorten the transmission end-to-end delay. *DLBL* also maintains multiple routes in order to provide alternative routes when the primary route is broken, and this enhances the reliability and robustness of networks.

Some multipath routing schemes have also been proposed by considering the fading effect. This is exemplified in [98], where Multi-Route Path Selection (*MRPS*) diversity scheme is introduced. It chooses the intermediate node which has the best current channel condition as the next hop for transmissions in order to reduce the fading effect. The channel state information (*CSI*) is obtained through *RTS/CTS* packets. Based on the same concept, a new routing metric called the Route Outage Probability (*ROP*), which uses multiple routing paths in order to minimize the packet loss due to fading, is developed in [99]. *ROP* prefers the routing path which has low packet loss probability and additional hops. In order to increase the probability of packet successful transmission, it is suggested that *ROP* is used in single route (*SR*), multiple route (*MR*) and *MRPS* based on a cross-layer protocol stack.

4.2.4 Energy-aware Multipath Routing Protocols

By considering multiple routing paths in energy-aware routing design, extra benefit can be gained. In [100], an Energy Aware Routing (*EAR*) protocol is developed and is intended to extend the network lifetime. *EAR* discovers the multiple routing path by using localized flooding of request messages, and assigns a probability based on the residual energy and the energy for transmissions at the nodes along the routing path. It maintains all multiple routing paths, but selects the routing path with higher residual energy and probability for packet transmissions.

Lu and Wong also developed an Energy-Efficient Multipath Routing Protocol (*EEMRP*) [101] to discover multiple node-disjoint paths between the source and the destination nodes. The traffic is then distributed across selected routing paths through a load balancing algorithm. The multiple routing paths are discovered based

on the link cost function, which takes into account both the residual energy at the nodes and the number of hops. By introducing a load balance ratio (fairness index), the traffic is distributed to the discovered multiple routing paths. The simulation results identify that EEMRP provides an efficient energy utilization, lower average transmission delay and control overhead. However, the link cost function only takes into account the ratio of the initiation to residual energy, as well as a hop count from the source node, but does not consider the transmission distance from the source node.

Similar to EEMRP, a distributed, scalable and localized multipath routing algorithm, called Energy-aware Multipath Routing Algorithm (*EMRA*) [102], is proposed to find node-disjoint paths for a node pair by only using localized information. Through considering network reliability, a load balancing algorithm is developed in order to distribute traffic across node-disjoint multiple paths. The algorithm assumes that each node in the network can acquire its residual energy level, failing probability and local information. Three main factors are taken into *EMRA* to select the required routing path, these are: node lifetime, energy-efficient transmission and transmission delay.

Unlike EEMRP, in [102], the transmission distance in multi-hop networks is discussed, and it identifies that determining an idea on the transmission distance is very important for designing energy-aware routing path selection algorithm in multi-hop ad hoc networks. Compared with *EEMRP* [101], the simulation results in [102] show that *EMRA* provides higher efficient energy utilization, lower control overhead and a shorter average delay. Furthermore, by considering network reliability, *EMRA* is available to use in unreliable environments for packet delivery. However, although the transmission distance is discussed in [102], it still does not consider the effect of physical layer parameters, such as data rates or transmission power with regard to the energy-aware design.

Another multipath energy-aware routing called Multipath Minimum Energy Routing (*MMER*) is proposed in [103] in order to minimize the total energy consumption in networks and therefore extend the network lifetime. An energy cost function is designed by taking account into total traffic flowing over links. An optimization problem is formulated in [103] and a gradient projection method is used to solve this optimization problem. Based on this gradient projection method, the *MMER* aims to minimize the sum of the link energy cost through the optimal distribution of traffic amongst multiple disjoint paths. The simulation results identify that

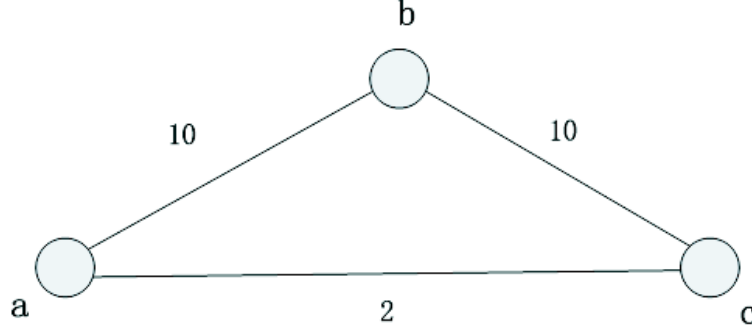


Figure 4.1: A Simple Multi-rate Ad Hoc Network

MMER performs well in static and low mobility ad hoc networks and achieves longer network lifetime.

4.3 Problem Addressed

Multiple data rates are widely applied in both wireless communications systems and Ad Hoc networks. Different transmission rates are achieved depending on the quality of radio channel. The rate adaptation scheme at devices dynamically selects and switches the data rates based on the channel estimation. In the IEEE 802.11, the multi-rate capability is provided for WLANs (table 2.1). The existing popular routing scheme, AODV, provides the shortest path solution to find the routing path with the minimum number of hops. However, AODV is developed based on the equal-rate ad hoc networks, and not the multi-rate ad hoc networks. In a multi-rate ad hoc network, the routing path with the minimum number of hops may not necessarily be the shortest path with the minimum delays or with the least energy consumption.

Figure 4.1 demonstrates a simple multi-rate ad hoc network. The data rate for the links between node a and node b, and node b and node c, is 10, but the data rate for the link between node a and node c is only 2. Here, the amount of bits required to be transmitted from node a to node c, is M . If the routing path, J_{ac} , is the only link l_{ac} , the transmission time is $\frac{M}{2}$ and the related energy consumed is $\frac{P_T \times M}{2}$. If the routing path, J_{ac} , includes links l_{ab} and l_{bc} , the transmission time is $\frac{M}{10} + \frac{M}{10} = \frac{M}{5}$, and the related energy consumed is $\frac{P_T \times M}{5}$. This simple example identifies that AODV may not provide a minimum transmission time nor the least

energy consumption solution.

4.4 Problem Formulation

Given a source node and destination node in a network, a total of M bits are required to be transmitted over T seconds through Z routing paths linking from the source to the destination. Regarding path z , a packet with L_z bit is used for each hop and $\bar{T}_{i,j}$ is defined as the service time between node i and j which are adjacent to each other for $i, j \in J_z$ where J_z represents a set of numbers denoting all nodes in path z for $z = 1, \dots, Z$. The required total data rate between the source and the destination for transmissions is given by

$$R = \frac{M}{T} \quad (4.9)$$

bit per second (bits/s). If a total of m_z bits are transmitted over T seconds using path z where $M = \sum_{z=1}^Z m_z$, the maximum MAC rate over path z is

$$R_{MAC,z,MAX} = \frac{L_z}{\max(\bar{T}_{i,j})_{i,j \in J_z}} \quad (4.10)$$

bit per second. The required MAC power to achieve $R_{MAC,z,MAX}$ for path z is then expressed as

$$P_{MAC,z} = \frac{P_T N_{SG} T_c m_z}{T} \sum_{i,l \in J_z} \frac{1}{r_{i,l}} \quad (4.11)$$

where P_T is the transmission power, N_{SG} is the spreading gain, T_c is the chip period and $r_{i,l}$ is the bit rate required for the transmission between node i and l , where $i, l \in J_z$. Therefore, the amount of energy consumed over T seconds is given as

$$\begin{aligned} E_z &= P_{MAC,z} T \\ &= P_T N_{SG} T_c m_z \sum_{i,j \in J_z} \frac{1}{r_{i,j}} \\ &= m_z E_{b,z} \end{aligned} \quad (4.12)$$

which is required to transmit m_z bits from the source to the destination node over path z where

$$E_{b,z} = P_T N_{SG} T_c \sum_{i,j \in J_z} \frac{1}{r_{i,j}} \quad (4.13)$$

is the energy consumed per bit for path z .

Therefore, once the energy consumption for each routing path is known, the total consumed energy is proposed to be minimized

$$\min(E_T) = \min \left(\sum_{z=1}^Z m_z E_{b,z} \right) \quad (4.14)$$

On this basis, the total data rate in bits per second is achieved as

$$R = \frac{M}{T} = \sum_{z=1}^Z \frac{m_z}{T} \quad (4.15)$$

In order to solve this margin adaptive optimization problem which minimizes the total energy, E_T , required for all routing paths $z = 1, \dots, Z$, the least consumed amounts of energy per bit, $E_{b,z}$ must be identified as suggested in the equation ((4.14)). Therefore, $E_{b,z}$ is used as the cost function of the path metric in this proposed multipath routing algorithm. Subject to minimizing $E_{b,z}$, the bit rate between all two adjacent active nodes, i and l in path z are required to be optimized, based on the equation (4.13), for $i, l \in J_z$.

Once the least amount of energy consumed per bit, $E_{b,z}$, is discovered for all possible disjointed routing paths $z = 1, \dots, Z$, the number of bits which should be allocated to path z need to be determined. In order to achieve a load balancing, an equal amount of energy is allocated for each path, such as $E_1 = \dots = E_z = \dots = E_Z$. Given the total amount of bits, M , the energy consumed at each path is given as follows.

$$\begin{aligned} M &= \sum_{z=1}^Z m_z \\ M &= \sum_{z=1}^Z \frac{E_z}{E_{b,z}} \\ E_z &= \frac{M}{\sum_{z=1}^Z \frac{1}{E_{b,z}}} \end{aligned} \quad (4.16)$$

Based on the equation (4.16), the number of bits allocated to routing path z is given as follows.

$$\begin{aligned} m_z &= \frac{E_z}{E_{b,z}} \\ m_z &= \frac{M}{E_{b,z} \sum_{z=1}^Z \frac{1}{E_{b,z}}} \end{aligned} \quad (4.17)$$

After formulating the energy minimization problem, a proposed modified multipath routing discovery scheme will be presented in the following section.

4.5 Disjoint Multiple Routing Paths Discovery

Before commencing the multi-path routing discovery, a connectivity or adjacency matrix, A , is formed to identify all possible links in an ad hoc network. For a given N -node ad hoc network, the adjacency matrix is an $N \times N$ dimensional square matrix. If there exists a link between the i^{th} and the j^{th} nodes, then $A(i, j) = 1$, otherwise $A(i, j) = 0$. The set of nodes adjacent to node V of the topology is denoted by $adj(V)$, of which the elements are known as the neighbours of node V , and the number of neighbours represents the degree of node V . Therefore, the maximum number of disjoint paths between a source and a destination node is the minimum value between the degrees of the source and the destination nodes.

Here, in regard to the multipath routing scheme, only the disjointed routing paths will be considered. When the source node starts to send the aggregated data to the destination node, the source node unicasts a REQUEST message to all neighbouring nodes with a distinct route ID. One of the neighbouring nodes which has the minimum path metric will be chosen to be forwarded with the REQUEST message. The routing path table is then updated with the information of the selected node, which will not be considered for selection when the rest of the routing paths are sought to ensure disjointed multi-path routing in the future.

However, all possible routing paths should be identified to obtain the low cost routing paths. One possible way of doing this is by using a Trellis-hop diagram approach, which will be discussed in the following section.

4.5.1 Trellis-hop Diagram Approach

Given an N -node network topology, all possible routing paths can be identified using a trellis-hop diagram. The diagram is organized by arranging $N \times N$ nodes. In each column, the nodes are ordered based on their ID, and the same node ID repeats in the row. Based on the link between nodes from the network topology, the routing paths are identified by drawing the links from the source node to its neighbouring nodes, which are denoted in the next column. Next, the same process starts from the neighbouring node until the drawing of routing paths reach the destination node.

An example of a 6-node network topology is given in figure 4.2 and the corresponding trellis-hop diagram is produced in figures 4.3 to 4.6. From these figures, eight routing paths, which pass through the neighbouring node 2 of the source node, are found; nine routing paths pass through the neighbouring node 3 of the source

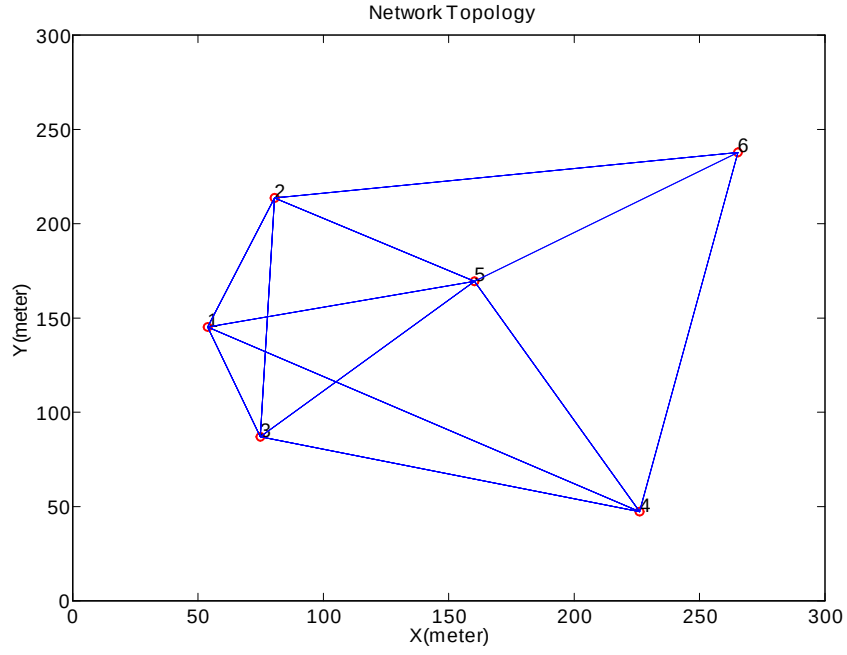


Figure 4.2: A 6-node Network Topology

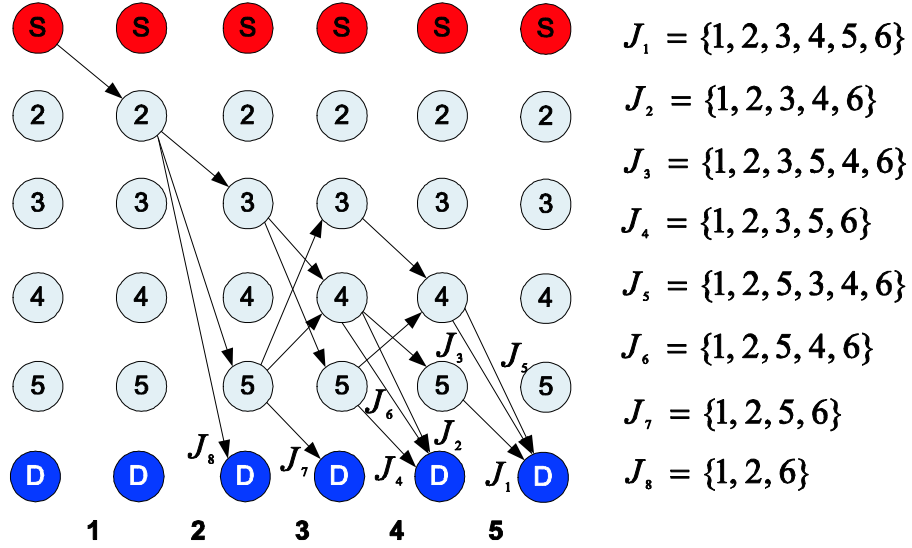


Figure 4.3: The First Neighbor of Source Node for Trellis-hop Diagram

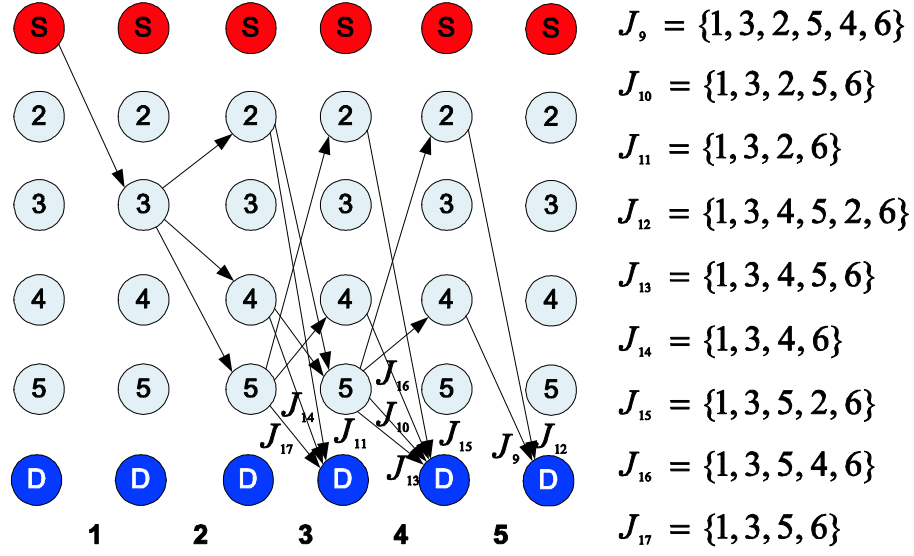


Figure 4.4: The Second Neighbor of Source Node for Trellis-hop Diagram

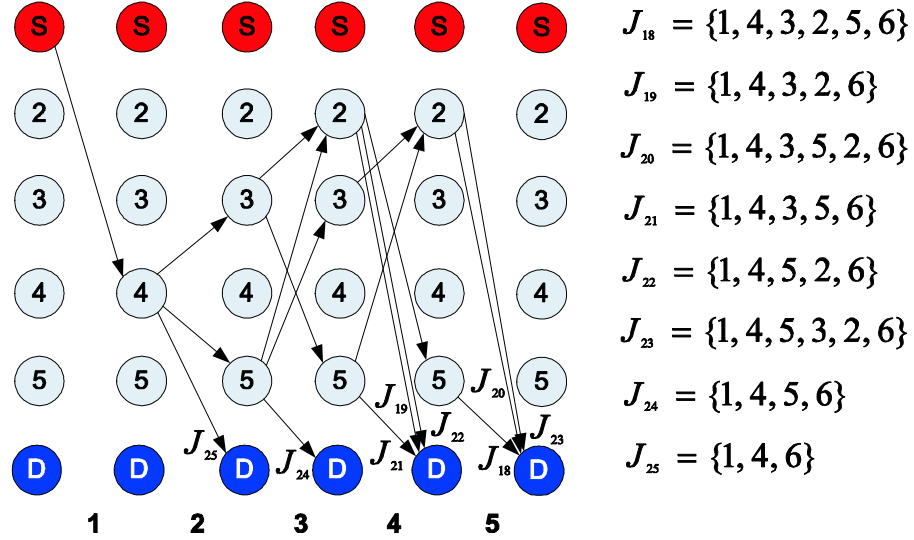


Figure 4.5: The Third Neighbor of Source Node for Trellis-hop Diagram

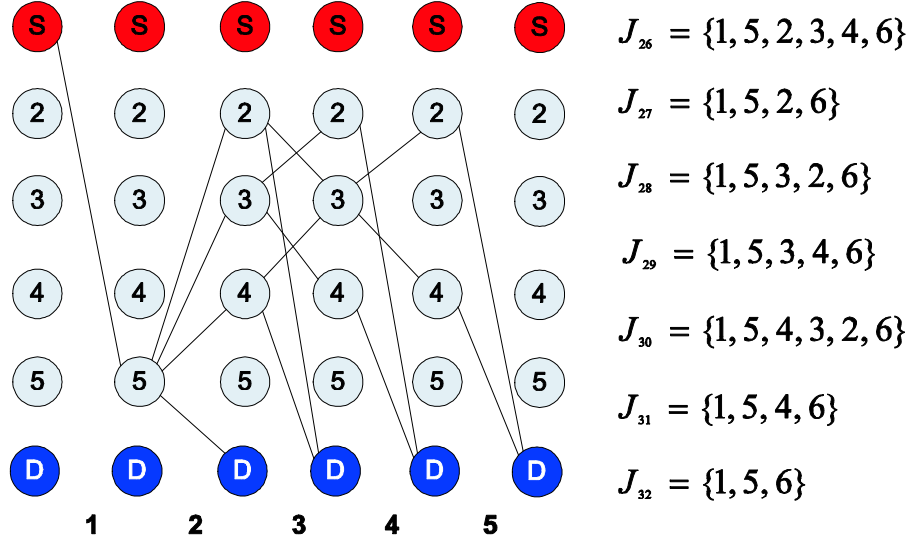


Figure 4.6: The Fourth Neighbor of Source Node for Trellis-hop Diagram

node; eight routing paths and seven routing paths pass through the source node's neighbouring node 4 and node 5, respectively. After completing the trellis-hop diagram, there are 32 routing paths in total that need to be identified. The cost of each routing path can be calculated based on the equation (4.13), and the one with the least cost will be identified.

However, it is obvious that this routing path discovery scheme has extensive computational complexity to discover all possible routing paths existing for a given node pair in a network. In order to solve this problem, a modified Viterbi algorithm is developed and presented in the following section.

4.5.2 Modified Viterbi Algorithm

The modified Viterbi algorithm aims to reduce the computational complexity of the routing path discovery process of the Trellis-hop diagram approach. This algorithm groups the link-joint routing paths and selects the routing path with the least cost and removes other routing paths from the routing table. Figure 4.7 gives a simple example to explain the modified Viterbi algorithm. At node 5, uncompleted routing paths A, B, and C have the same next hop ID, node 6, and therefore all of them pass through the link (5, 6). The cost of these link-joint uncompleted routing paths is compared and the uncompleted routing path with the least cost is chosen, e.g. uncompleted routing path A, and other uncompleted routing paths B and C are

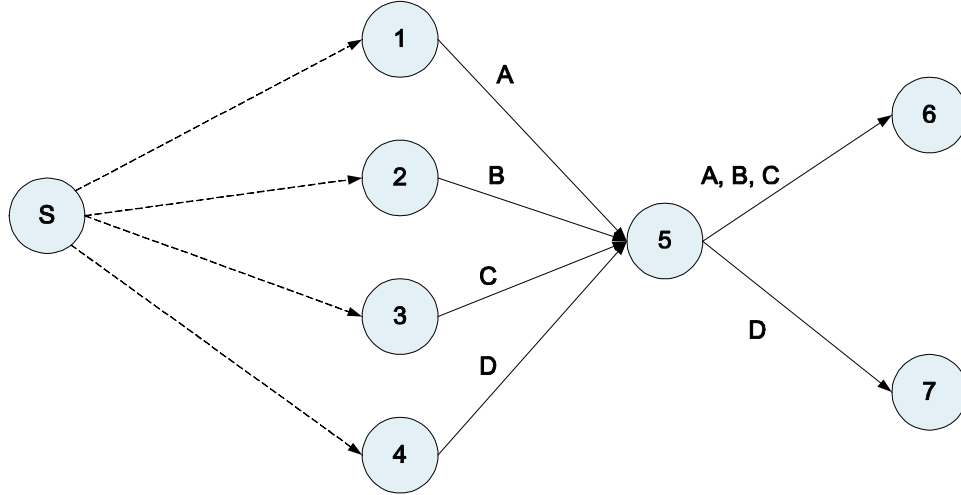


Figure 4.7: Modified Viterbi Algorithm

deleted from the routing table.

In order to implement the modified Viterbi algorithm, a "hop level checking" method is developed. It groups the routing paths based on different number of hops, k , where $k = 1, 2, \dots, N - 1$. The routing paths discovery process starts from the lowest hop level, G_1 , in which the link cost is calculated using equation (4.13) for all uncompleted routing paths. This calculation process is repeated for all hop levels. At each hop level, the routing paths which arrive at the destination node are considered as the completed routing paths and recorded in the completed routing path table before being removed from the uncompleted routing path table. If there are more than one uncompleted routing paths which pass through the same link between the current node and the next node, these routing paths are marked as the link-joint uncompleted routing paths. Their costs are compared and the uncompleted link-joint routing path which has the least path cost is selected, and other link-joint uncompleted routing paths are removed from the routing table. This "hop level checking" process is terminated when all possible low cost routing paths are found. There are three main routing path tables for the "hop level checking" process and these are: (1) the completed routing path table, (2) the link-joint uncompleted routing path table, and (3) the uncompleted link-disjoint routing path table.

Here, an example is given based on the network topology given in figure 4.2. Figures 4.8 to 4.12 show the process of hop level checking. Firstly, the source node starts to search for its neighbours within one hop, e.g. node 2, node 3, node 4 and node 5. This first-hop level checking process results in a group of uncompleted

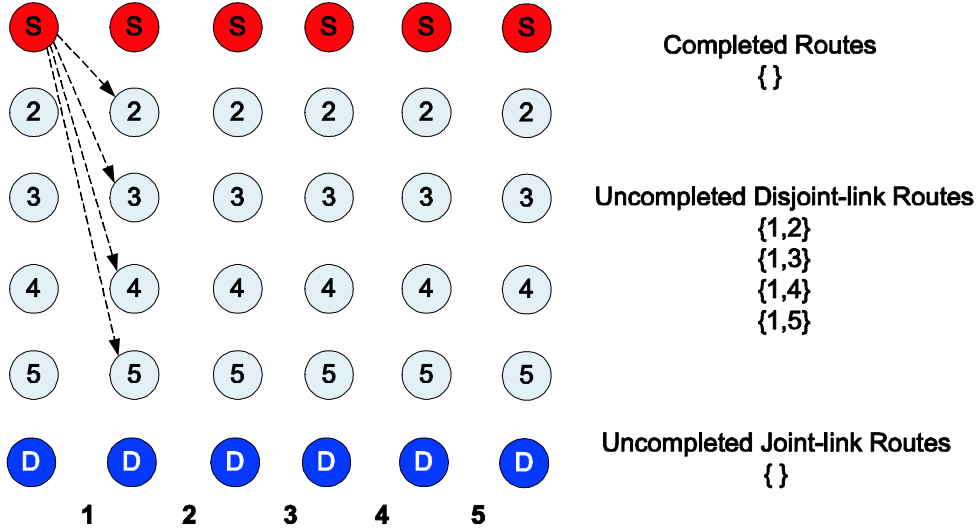


Figure 4.8: The First Hop Level for Routing Paths Discovery

disjoint-link routes, i.e. $\{1,2\}$ $\{1,3\}$ and $\{1,4\}$. Next, the source node's neighbouring nodes search their neighbours, which have a two-hop distance from the source node. After this process, three completed routes, which reach the destination node, are found and recorded in the completed route table, e.g. $\{1,2,6\}$, $\{1,4,6\}$ and $\{1,5,6\}$. The uncompleted disjoint-link routes table is then updated (Figure 4.9) and no uncompleted joint-link routes is found during the second-hop level checking process. After running the third-hop level checking, the completed route table and the uncompleted disjoint-link route table are updated. In addition, uncompleted joint-link routes are found, such as $\{1,2,3,5\}$ and $\{1,4,3,5\}$. Through comparing the sum cost of links along the routing path, the routing path with the least cost is selected and rest of them are removed from the uncompleted route table. This process continues for the fourth-hop level checking and fifth-level checking until all completed routes were discovered. From Figure 4.12, twenty-three completed routing paths are discovered after all the hop-level checking processes. Compared with the Trellis-hop diagram approach, the modified Viterbi algorithm reduces the computational complexity of routing path discovery for a node pair $\{1,6\}$ by more than 25% in the network topology given in figure 4.2. With the increase of the number of nodes and links in a network, the benefit gained by modified Viterbi algorithm will significantly increase.

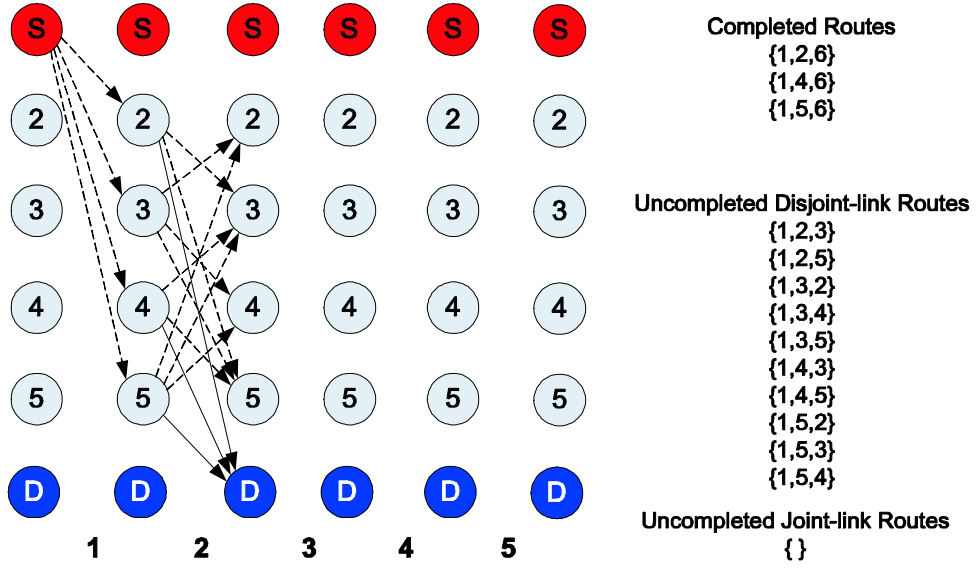


Figure 4.9: The Second Hop Level for Routing Paths Discovery

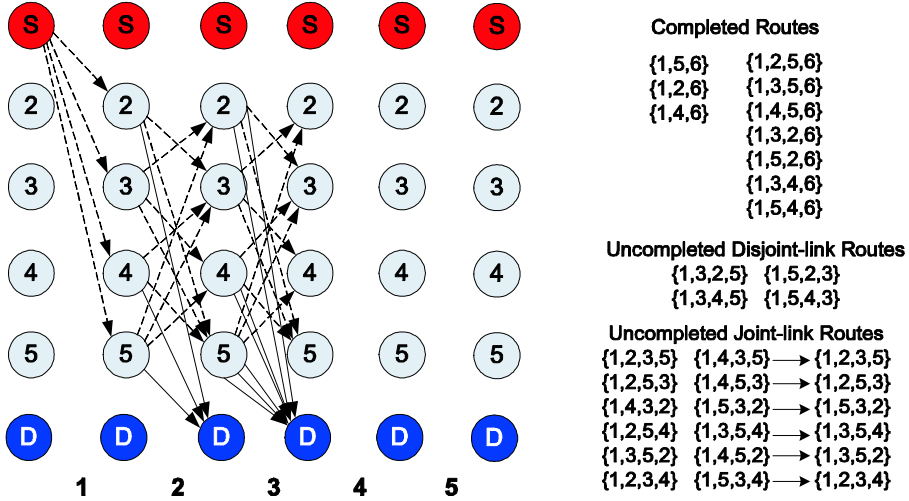


Figure 4.10: The Third Hop Level for Routing Paths Discovery

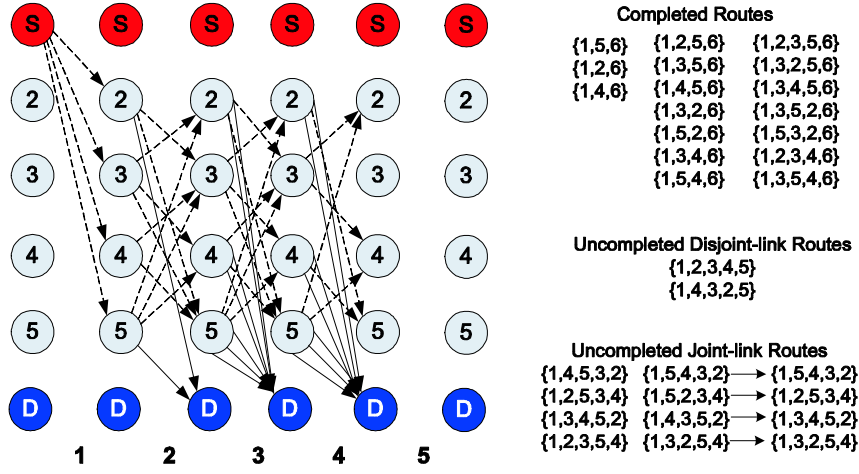


Figure 4.11: The Fourth Hop Level for Routing Paths Discovery

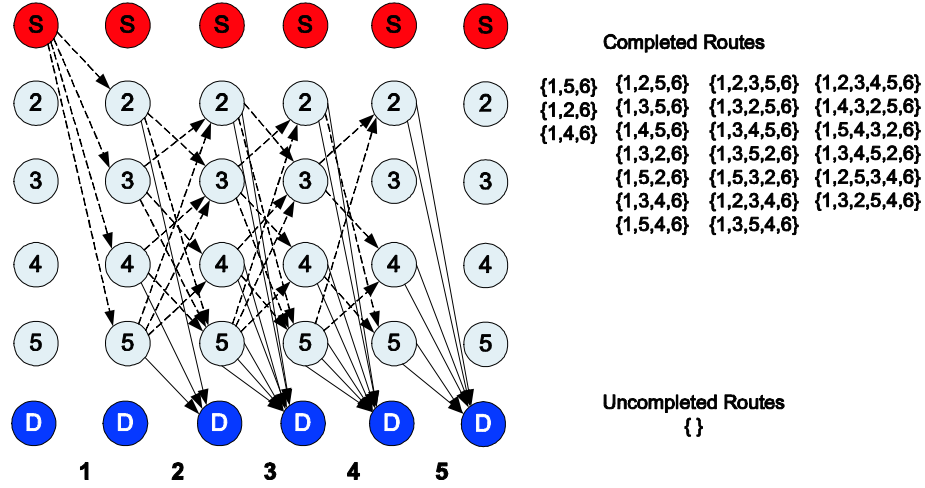


Figure 4.12: The Fifth Hop Level for Routing Paths Discovery

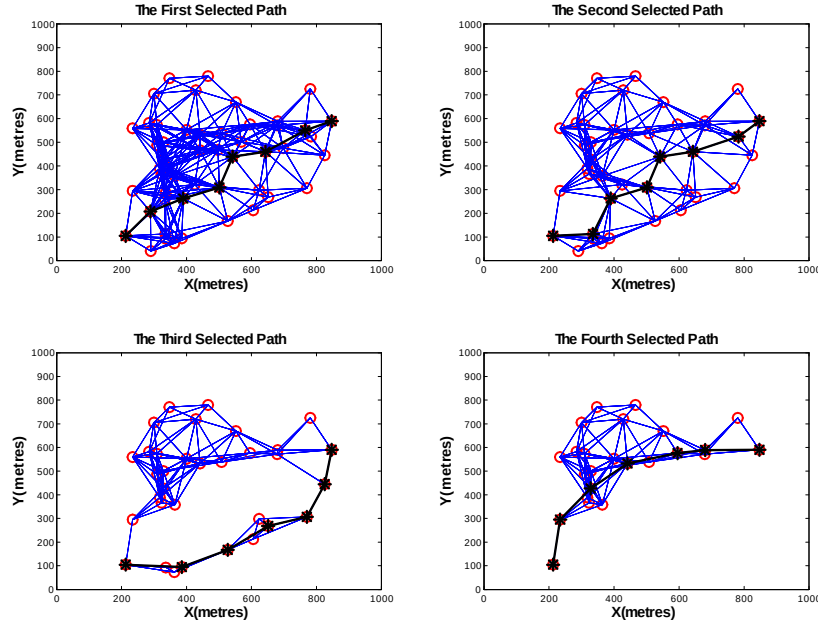


Figure 4.13: Disjoint Multiple Routing Discovery

4.6 Energy-aware Single-path Routing (EASR) and Energy-aware Disjoint Multi-path Routing (EADMR)

After running the proposed routing path discovery scheme, if the energy consumption is the only issue that needs to be considered here, the routing path with the least cost will be selected from the completed route table and defined as the energy-aware single-path routing (EASR). As regards the energy-aware disjoint multipath routing (EADMR), there are many combination options from the completed route table. Here, a network topology restructuring approach is developed to find the EADMR for a node pair given in a network topology.

In order to ensure disjoint multiple routing paths, one low cost routing path is selected from the completed routing paths table. Next, all nodes along the chosen routing path are removed from the network topology and a new topology is restructured. The routing discovery is then run again to find the remaining disjointed routing paths. This process continues until all disjointed routing paths are found. Here, the network topology plotted in Figure 2.2 is used as an example. Given a node pair (27, 44), a group of low cost disjoint routing paths is found (figure 4.13): [27-29-23-20-16-10-12-42-41-46-44]; [27-25-26-30-9-40-37-48-44]; [27-24-32-33-34-35-

36-47-44]; [27-28-21-19-14-7-6-50-11-43-44].

In order to extend the network lifetime, different groups of low-cost disjoint routing paths can be generated. For example, there exists another disjointed routing path group for the network topology given in Figure 2.2: [27-29-23-30-39-9-40-37-46-44]; [27-25-24-32-33-34-38-35-36-47-44]; [27-26-20-16-10-12-42-41-48-44]; [27-28-21-19-14-7-6-50-11-43-44]. With different disjointed routing paths groups, the network lifetime is increased since more options of routing paths exist, especially at the event of path failures.

4.7 Performance Evaluation

4.7.1 Low Cost Routing Path Distribution

In this section, the low cost routing path distribution is studied in ad hoc networks. Given the transmission power, P_t , chip rate, T_c , and spreading gain, N_{SG} , at each node, the cost of a routing path is dependent on both the data rate and the number of hops along the routing path as given in the equation (4.13). In this experiment, ten 50-node different network topologies are generated and one node pair with minimum 5 hops is given in each network topology. Given a threshold of energy consumed per bit, such as $1.5\text{e-}9$ J/bit, the discovered routing paths, which have the same number of hops, are accounted for and plotted in Figure 4.14. The result shows that the low energy consumption routing path requires 7 to 25 hops for 50-node networks and 8 to 38 hops for 100-node networks. In addition to this, it can also be observed that the routing paths with a small number of hops may not result in low energy consumption, and this identifies the problem addressed in Section 4.3.

4.7.2 EASR vs AODV

Given node pairs under different scenarios, figure 4.15 gives the energy consumed per bit of EASR and AODV which are examined in five different network topologies. The routing path with the least cost is chosen for both EASR and AODV, while the two-group HSDPA system is compared with the original HSDPA system. It clearly shows that the EASR has much less energy consumption compare with AODV. The two-group allocation, which provides data rate enhancement at the physical layer, provides further energy consumption reduction for both EASR and AODV.

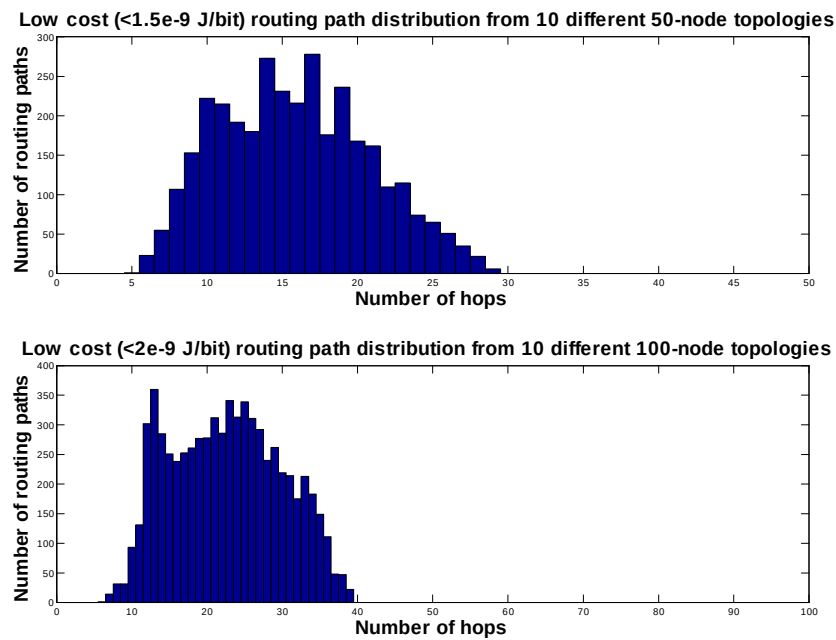


Figure 4.14: Low Cost Routing Paths Distribution

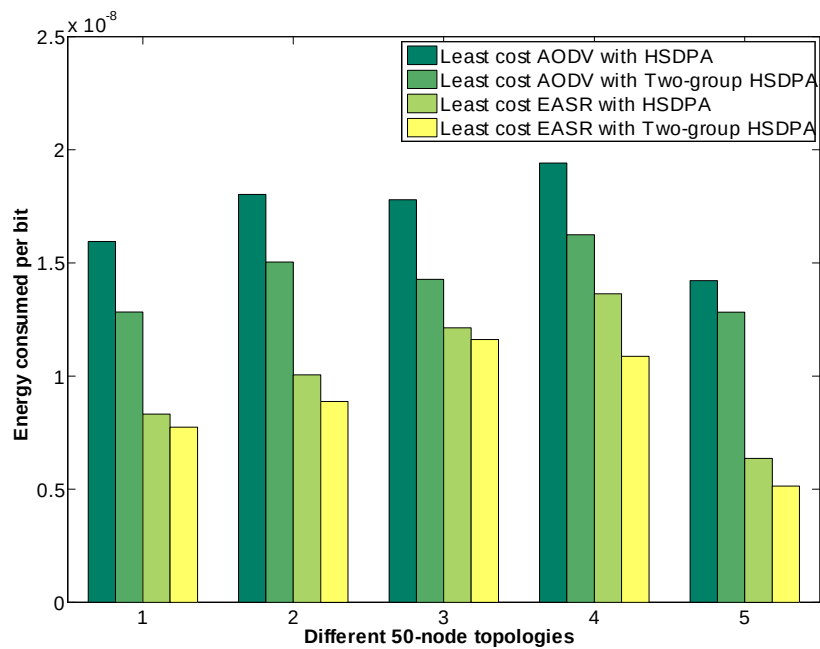


Figure 4.15: Performance of AODV and EASR

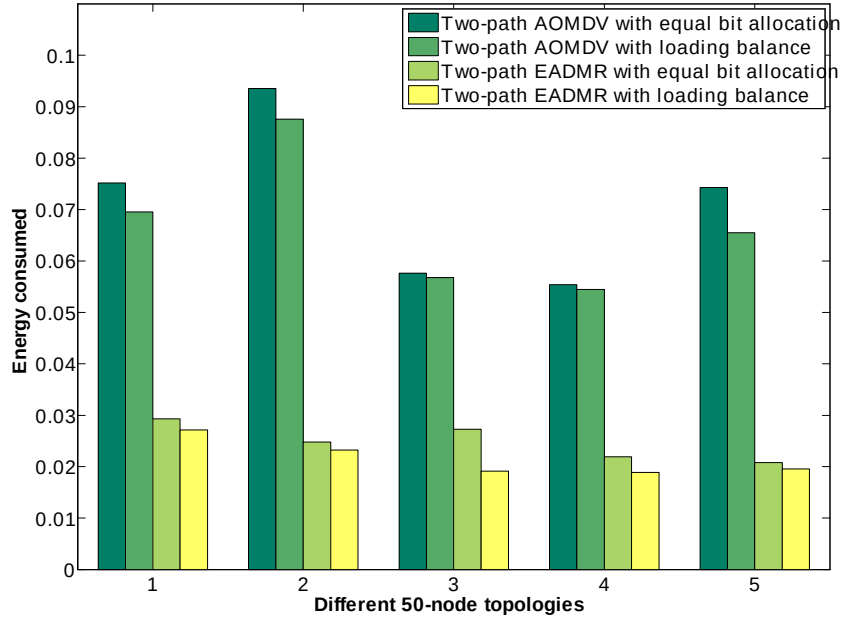


Figure 4.16: Performance of AOMDV and EADMR

4.7.3 EADMR vs AOMDV

A comparison between the multi-path AODV (AOMDV) and the proposed EADMR schemes was also studied. Without the loss of generality, two routing paths are selected for both AOMDV and EADMR, with 300kbytes of information bits fed. Figure 4.16 illustrates the amount of energy consumed for both AOMDV and EADMR in five different topologies. It can be seen that the EADMR requires much less energy than the AOMDV due to lower path cost. When load balancing is incorporated, the bits are allocated more efficiently and therefore the energy consumption is further reduced. However, if the path cost for different routing paths are closed, the load balancing approach does not show any significant improvements.

4.8 Conclusion

In this chapter, the energy-aware routing was studied. After reviewing the related work on energy-aware routing, MAC-aware routing, multipath routing and energy-aware multipath routing, the energy minimization problem in multi-rate ad hoc networks was addressed and formulated. In order to extend the network lifetime through finding routing paths which consume low energy in ad hoc networks, an

energy-aware routing scheme is developed. With regard to the proposed energy-aware routing scheme, the energy consumed per bit, $E_{b,z}$, was defined as the cost function of path metric and a trellis-hop diagram approach was developed to find all possible routing paths in an ad hoc network for any given node pair. In addition, a modified Viterbi algorithm was developed to reduce the computational complexity of the routing path discovery process. Furthermore, a network topology restructuring approach was proposed to find energy-aware disjoint multipath routing and a load balancing approach was developed for further energy consumption reduction.

The performance of proposed energy-aware single-path routing (EASR) and existing routing scheme (AODV) were compared, and the result identified that EASR consumes less energy. The performance of two-group and single-group HSDPA in a same routing scheme were also compared and the result shows that two-group HSDPA provides a low energy consumption solution due to its data rate enhancement at the physical layer. Furthermore, the proposed energy-aware disjoint multipath routing (EADMR) is examined and compared with multipath AODV (AOMDV). The result obviously claims the significant improvement on energy consumption reduction of EADMR. Finally, the load balancing technique was examined and identified that it is capable of further reducing the energy consumption in multipath routing.

Chapter 5

MAC DCF Service Time

5.1 Introduction

Based on the OSI seven-layer model presented in section 2.1, it can be said that where wireless networks are concerned, the physical layer is used to manage the movement of packets from the MAC layer to the radio channel at the transmitter, as well as managing the movement of packets from the radio channel to the MAC layer at the receiver. The MAC layer is used to control the nodes as to access channels with the purpose of achieving fairness in networks, thus avoiding collisions. In other words, the PHY layer determines how to transmit frames; the MAC layer decides on which nodes to use in networks and when to transmit; and finally, the network layer discovers which routing path to use and also determines how to forward the information from the source to the destination.

However, the data rate achieved at the physical layer only gives the upper bound of network throughput. Increasing the network throughput upper bound does not result in the same proportion on its related network throughput improvement. From a MAC layer perspective, many factors contribute to inefficient transmissions, throughput reduction and long delays, adding to problems such as overhead, packet drop and re-transmissions. On this basis, in order to achieve an efficient transmission and enhance the network throughput, the MAC layer study is essential.

In the previous chapter, the energy-aware disjoint multipath routing and related load balancing scheme were studied. When the total amount of bits, M , is distributed to each routing path based on the load balancing approach developed in equation (4.17), these information bits are divided into data packets for transmissions, which results in an end-to-end delay for a routing path. The end-to-end delay equals to the

time consumed starting from the first packet departure at the source node lasting until the last packet has arrived at the destination node. After allocating related information bits to multiple routing paths for transmissions, a problem is addressed on the management of the data packet transmission along the routing path in order to achieve a low end-to-end delay by considering the service time at the MAC layer.

5.2 IEEE 802.11 Distributed Coordination Function (DCF)

5.2.1 DCF Transmissions

With regard to the 802.11 MAC layer, the DCF is defined as the distributed channel access MAC protocol. The DCF provides two options for packet transmissions which consist of a two-way handshaking (or basic access) mechanism and an optional four-way handshaking (or RTS/CTS) mechanism. Both methods follow the backoff rule to control the channel access and avoid collision. Nodes in a network are required to listen to the channel state for a DCF Inter Frame Space (DIFS) interval before transmitting. If the channel is sensed to be busy during a DIFS interval, the node enters the backoff process and is prevented from transmitting. Here, the RTS/CTS mechanism, which provides better solutions to solve hidden terminal problems, is concentrated in this chapter.

Once the node is permitted to transmit packets, the source node will transmit a Request-to-Send (RTS) packet to the destination node. If this RTS packet is received correctly by the destination node and the channel is still free, a Clear-to-Send (CTS) packet responds to the transmitter. RTS/CTS packets contain information about the length of the data packets, so nodes that received the packets can be on standby for a sufficient time interval as needed for data packet transmissions. If the source node successfully receives a CTS packet, it is permitted to transmit the data packet. When the data packet is received successfully by the destination node, a corresponding acknowledgment (or ACK) packet is sent from the destination node to the source node. The transmissions of all packets (RTS/CTS/DATA/ACK) are separated by a time slot known as the Short Inter-Frame Space (SIFS). The SIFS allows time for switching the transceiver between sending and receiving nodes. Figure 5.1 demonstrates a typical DCF transmission process. The information inside these MAC control packets contains the IP address of the destination node and the

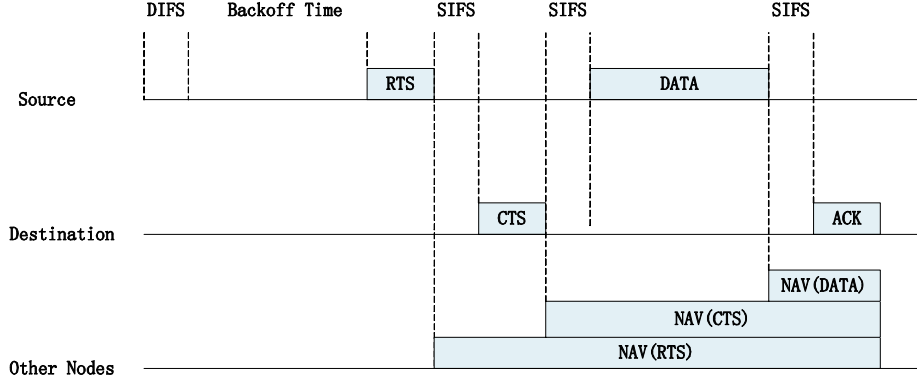


Figure 5.1: DCF Transmission

duration time of the transmission. Based on these information, source nodes' neighbours, update a list called the Network Allocation Vector (NAV), which maintains the remaining time for deferring nodes' transmission.

With regard to the successful transmission time, it is defined as the sum of time consumed for all packets transmissions and intervals during one full successful transmission. From figure 5.1, the successful transmission time in DCF is given as follows.

$$T_s = t_{DIFS} + t_{RTS} + t_{CTS} + 3 \times t_{SIFS} + 4 \times \delta + \frac{L}{r_{data}} + t_{ACK} \quad (5.1)$$

Where t_{RTS} , t_{CTS} and t_{ACK} are the time slots for control packet (RTS/CTS/ACK) transmissions. These control packets are transmitted through a basic rate in DCF, e.g. 1Mb/s, t_{SIFS} is the time interval between packet transmissions, δ is propagation delay and $\frac{L}{r_{data}}$ is the time required for a data packet transmission. If there is no contention between nodes in a network, e.g. peer-to-peer transmission, the service time equals to the successful transmission time, $\bar{T} = T_s$. However, in a large network, where contention cannot be avoided, the node then has to obtain permission to access the channel for transmissions, thus resulting in a waiting period known as the backoff time.

Here, it is assumed that the collision only occurs on the RTS packet, so the time consumed due to a collision, is defined as followed.

$$t_c = t_{RTS} + t_{DIFS} + \delta \quad (5.2)$$

This means that if the time duration exceeds t_c and the destination node still does not receive the RTS packet, a collision is considered to occur.

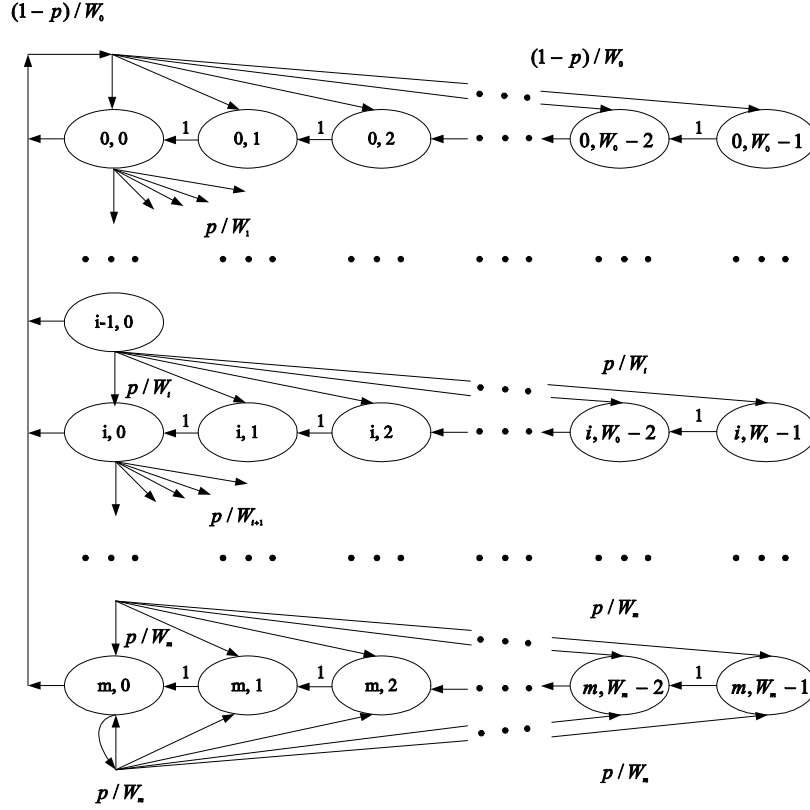


Figure 5.2: Markov Chain Model for Backoff Process (taken from [99])

5.2.2 DCF Backoff Process

In an ad hoc network, all nodes try to access the channel, therefore resulting in contention between nodes. The DCF is designed to control the right of accessing the channel in order to balance fairness and avoid the collisions in a given network. The method used in the DCF to control the contention between nodes is the backoff scheme.

At each node, there is a counter to decide when the node is permitted to transmit. After sensing the channel is idle for a duration as long as a DIFS interval, the counter is given a backoff time, whose range is uniformly chosen from $(0, W)$, where W is the contention window size. Next, this given backoff time starts to decrease until the channel is sensed to be busy again. A Markov chain model (figure 5.2) is developed in [104] for analyzing the backoff process in DCF. It assumes an ideal channel condition, with a fixed number of nodes and a saturation condition. In the Markov chain model as shown in figure 5.2, each row represents the backoff steps and each column represents the backoff stage.

5.2.2.1 Backoff Step

At the beginning of each backoff process, the minimum value $W_{\min}$ is chosen as the initial contention window size, and the initial value of the backoff time in the counter is uniformly chosen from $(0, W_{\min})$. As a discrete and integer time scale is adopted, the probability of a backoff time at the initial stage 0 is $\frac{1}{W_o}$. After sensing the channel is idle, the backoff counter starts to decrease, e.g. $W_{\min} - 1$ to $W_{\min} - 2$, with a successful probability $\frac{1-p}{W_o}$ for each decrement, where p is the probability of collision. If the channel is sensed to be busy, the counter is frozen until the channel is free again. When the backoff counter decreases to zero, the node is permitted to transmit a packet with a probability of successful transmission, $1 - p$.

5.2.2.2 Backoff Stage

When a collision occurs, the backoff process will enter to the next stage and the contention window size is doubled after every unsuccessful transmission attempt in order to reduce the probability of collision. At the stage i , the contention window size is $W_i = 2^i W_{\min}$, $i \in [0 \dots m]$, where m is the maximum stage value, and the backoff time will be uniformly chosen from the range $(0, W_{i+1})$. If the collision occurs on the final stage of the backoff process, the contention window size will not double again and will maintain the maximum value. If the transmission is successful at any backoff stage, the backoff process will return to the initial stage and the contention window size will be set to $W_{\min}$.

5.2.2.3 Transmission Probability

The main assumption in this Markov model is that at each transmission attempt, each packet has a constant and independent collision probability p , which is known as the condition collision probability. This Markov chain model is more accurate as long as W and N increase to larger value. Based on the backoff rule, all transmissions occur when the backoff time decreases to zero in the counter, regardless of the backoff stage. On this basis, the transmission probability, τ , is modelled in [104] as a function of collision probability, p .

$$\tau = \frac{2(1-2p)}{(1-2p)(W_{\min}+1) + pW_{\min}(1-(2p)^m)}, m = \log_2\left(\frac{W_{\max}}{W_{\min}}\right) \quad (5.3)$$

5.3 Related Work

At the MAC layer, protocols are designed to avoid collisions and hence save energy through reducing the number of retransmissions. In [105], a MAC protocol is studied in the multiple channels CSMA and Dynamic Channel Assignment with Power Control (DCA-PC) for reducing collisions and saving energy. With regard to the multiple channel CSMA protocol, the idle and busy tables are introduced to detect the channel state for avoiding collisions, and are based on the random selection and reservation schemes to choose the channels. By exploiting the multi-channel MAC protocol concept, a MAC protocol is designed for the multi-code CDMA scheme, as to dynamically allocate multiple codes [106]. The transmitter and receiver negotiate the number of spreading codes by exchanging RTS/CTS packets through contention-based common codes.

In order to investigate the DCF in multi-hop large ad hoc networks, a mathematical analysis model is developed in [12] [107]. The key feature of this model is that a node can be modelled individually and it provides an interaction to directly link physical parameters with the behavior of the MAC protocol. In addition, these models also consider the interdependency of nodes through the network topology which relates it to the network contention. As the DCF does not provide any QoS services, many research studies have focused on the priority backoff process and service differentiation. In [108], a priority backoff algorithm (PBA) for the IEEE 802.11 DCF was developed to properly assign network throughput between nodes in proportion to their priority. In order to achieve this, each node is supposed to collect the statistical data of other nodes' transmission as well as maintain a Sent Data Table for all nodes in a network. Before transmitting the data, the node would evaluate contention window sizes according to the statistical data and priority. A dynamic priority backoff algorithm (DPBA) was proposed in [109] as an extension of the PBA to consider dynamic priority in terms of the DCF performance.

Current DCF protocol does not consider packet size for medium access operations as all packets win the channel contention equally. However, long packets occupy the channel for a long time and therefore unfairness increases delay and jitter for short packets. In order to solve this problem, a fully distributed algorithm, which considers the packet size in the backoff process, is proposed in [110]. The results identify the significant improvement of the delay reduction of the small packets. In a multi-rate network, the low data rate link takes down the throughput of the routing path. In order to solve this problem, a simple and standard-compliant algorithm

is developed in [111] to provide fairness when utilizing the channel. An adaptive scheme is introduced to adjust the packet size based on the data rate, and therefore enable nodes to occupy the channel for equal amounts of time. For example, the node with a low data rate will reduce the packet size.

As an extension of the DCF, the EDCF provides QoS and differential services. To date, much research has focused on the performance analysis of the EDCF [112] [113] [114] [115]. In [113], the expected bandwidth of each traffic class for the service differentiation mechanism in the EDCF was studied and evaluated through an approximate analytical model. Another analytical model was introduced in [112] to calculate the traffic priority and throughput under the saturation conditions. Based on this analytical model, the effect on service differentiation for each contention parameter was analyzed. By focusing on the post-collision period, which was ignored by previous EDCF modelling studies, reference [114] proposed and validated an analytical model to accurately predict the saturation throughput of the EDCF. In [115], the behavior of the backoff entity of each of the access categories (ACs) in the EDCF was modelled as a two-state Markov chain, which was an extension of the Markov chain model developed by Bianchi in [104]. Unlike previous EDCF modeling work, the analytical model developed in [115] considered virtual collisions between different ACs; the influence of using different AIFS for different ACs on the saturation performance; delay and delay jitter.

In [116], the performance of the contention-based channel access mechanism of the IEEE 802.11 and 802.11e were investigated and evaluated. The maximum throughput achieved in the legacy DCF and the EDCF were compared with two different simulation scenarios. The results show that IEEE 802.11e provides an improvement for network performance in terms of throughput, delay and packet loss.

5.4 MAC DCF Service Time

In [107] and [12], Carvalho developed a DCF analytical model by studying different channel states. The main difference between the DCF mathematical model developed by Bianchi [104] and Carvalho [107] is that Carvalho's analytical model considers a large ad hoc network which requires multi-hops. Furthermore, another key feature of Carvalho's model is that the nodes are modelled individually as to provide an interaction to directly link physical parameters into the behaviour of

the MAC protocol. In addition, these models also consider the interdependency of nodes through the network topology.

At the MAC layer, the service time, $\bar{T}$, includes the successful transmission time T_s and the average backoff time $\bar{T}_B$.

$$\bar{T} = T_s + \bar{T}_B \quad (5.4)$$

In the previous chapter, large ad hoc networks, which require routing to forward information, are studied. Here, the Carvalho's model and related MAC DCF-based service time are investigated in large ad hoc networks. In the following sections, both the backoff time and interaction models developed in [107] and [12] are studies.

5.4.1 Backoff Time Model

The contention between nodes in a large network is complex due to a lot of variables and the random process, hence the backoff time is unpredictable. In [107], Carvalho develops a mathematical DCF model in order to analyze the backoff process and therefore obtain the average backoff time, $\bar{T}_B$, as follows. It is assumed that this model does not consider the maximum retransmission, which means that retransmission will continue until the frame is successfully received.

$$\bar{T}_B = \frac{\alpha (W_{\min}\beta - 1)}{2q} + \frac{1-q}{q}t_c, \text{ where } \alpha = t_i p_i + t_c p_c + t_s p_s \quad (5.5)$$

where α is the average time cost per event during a backoff process, t_i , t_c and t_s are the time spent on different channel states, e.g. idle, collision and success, and p_i , p_c and p_s are the probability of corresponding different channel states. $\beta = \frac{q-2^m(1-q)^{m+1}}{1-2(1-q)}$, where q is the successful transmission probability, $m = \log_2(\frac{W_{\max}}{W_{\min}})$, $W_{\min}$ is the initial (or minimum) contention window size and $W_{\max}$ is the maximum contention window size. The probability of different channel states p_i , p_c and p_s is then analyzed as follows.

The probability is that a channel which is busy due to some nodes within node i 's carrier sensing range R_i will transmit a frame while node i is in its backoff process is

$$p_{tr}^i = 1 - \prod_{j \in R_i} (1 - \tau_j) \quad (5.6)$$

where τ_j is the transmission probability of node j . On this basis, the probability that channel is idle within i 's carrier sensing range is simply given as follows.

$$p_i^i = 1 - p_{tr}^i \quad (5.7)$$

Next, the probability of a successful transmission occurring within i 's carrier sensing range is given and conditioned on the fact that at least one node in R_i has attempted to transmit.

$$p_s^i = n_i \tau_i \prod_{j \neq i, j \in R_i} (1 - \tau_j) \quad (5.8)$$

Finally, the probability of a collision occurs within i 's sensing range is given as follows.

$$\begin{aligned} p_c^i &= p_{tr}^i - p_s^i \\ &= 1 - \prod_{j \in R_i} (1 - \tau_j) - n_i \tau_i \prod_{j \neq i, j \in R_i} (1 - \tau_j) \end{aligned} \quad (5.9)$$

In the 802.11 standard, the value of the maximum and minimum contention window size, $W_{\max}$ and $W_{\min}$, are given. In terms of the time slot for different channel states, t_c and t_s were given in equation (5.2) and (5.1) respectively, and the idle timeslot, t_i , equals to 20μ second. Once the network topology is given, the nodes within i 's carrier sensing range R_i will be known. Based on equations (5.7) (5.8) and (5.9), the probabilities of different channel states p_i , p_c and p_s are all a function of τ , which is the transmission probability. In order to calculate the average backoff time, $\bar{T}_B$, from equation (5.5), two parameters, q and τ , which represent the behaviour of MAC protocol, should be known.

In an ad hoc network, the MAC protocol dynamically adjusts its behaviour to schedule the future retransmission in order to minimize the number of unsuccessful transmissions according to the feedback information. For example, if the contention between nodes in a network is high, the successful transmission probability, q , will tend to decrease. In order to achieve this, the MAC protocol will limit the transmission frequency at nodes to reduce the contention level through controlling the channel access right of nodes. When the successful transmission probability tends to be high, the MAC protocol will increase the transmission probability, and this will result in high contention levels again. Accordingly to this, the MAC protocol is considered as a dynamic system whose feedback information is the successful transmission probability, q , and the corresponding outputs is the scheduling rate, τ , e.g. $\tau = h(q)$. In [12], based on the equation (5.3) developed in [104], an approximate transmission probability, τ , is proposed as a function of the successful transmission probability, q , as follows.

$$\tau = \frac{2W_{\min}}{(W_{\min} + 1)^2} q \quad (5.10)$$

Here, in order to obtain the average backoff time, $\bar{T}_B$, from equation (5.5), the only unknown parameter is the successful transmission probability, q . The following section will then look at the successful transmission probability, q , which is studied through introducing an interaction model developed in [12] for multi-hop ad hoc networks.

5.4.2 Interaction Analysis Model

Unlike the single-hop ad hoc network, the multi-hop ad hoc network is more complex as the contention is not only dependent on the nodes within the carrier sensing range of the source node, but is also affected by the rest of nodes in a ad hoc network. In order to study the interdependence between nodes in a multi-hop network and obtain the successful transmission probability, q , an interaction model is developed in [12] as follows.

$$\underline{q} = (\mathbf{I} + \mathbf{\Phi})^{-1} \underline{\pi} \quad (5.11)$$

where $\mathbf{I}$ is the identity matrix, $\underline{q}$ is a vector of successful transmission probability of all active nodes in a given network, e.g. $[q_1 \ q_2 \ \cdots \ q_n]$, $\underline{\pi}$ is a vector of handshake successful probability of all active nodes in a given network, and $\mathbf{\Phi}$ is the interference matrix. This interaction model directly links the MAC parameter (e.g. the successful transmission probability, q) to the physical layer parameter and the network topology. Given a network topology, the neighbourhood distribution is provided for the interference matrix, $\mathbf{\Phi}$, which is given as follows.

$$\mathbf{\Phi} = \begin{bmatrix} 0 & \phi_{12} & \cdots & \phi_{1n} \\ \phi_{21} & 0 & \cdots & \phi_{2n} \\ \vdots & \vdots & \ddots & \vdots \\ \phi_{n1} & \phi_{n2} & \cdots & 0 \end{bmatrix}, \quad \phi_{ij} = \begin{cases} \frac{2W_{\min}}{(W_{\min}+1)^2} \pi_{ij} & \text{if } j \in V_i \\ 0 & \text{if } j \notin V_i \end{cases} \quad (5.12)$$

The non-zero elements in the interference matrix indicate the neighbourhood between nodes, and the corresponding value is determined by the handshake probability, π . For example, π_{ij} means that there is link between node i and node j .

In an ad hoc network, if a collision happens at the receiver, the whole frame will be dropped. Accordingly, the related probability of successful frame reception is given as follows.

$$p_f = (1 - BEP)^L \quad (5.13)$$

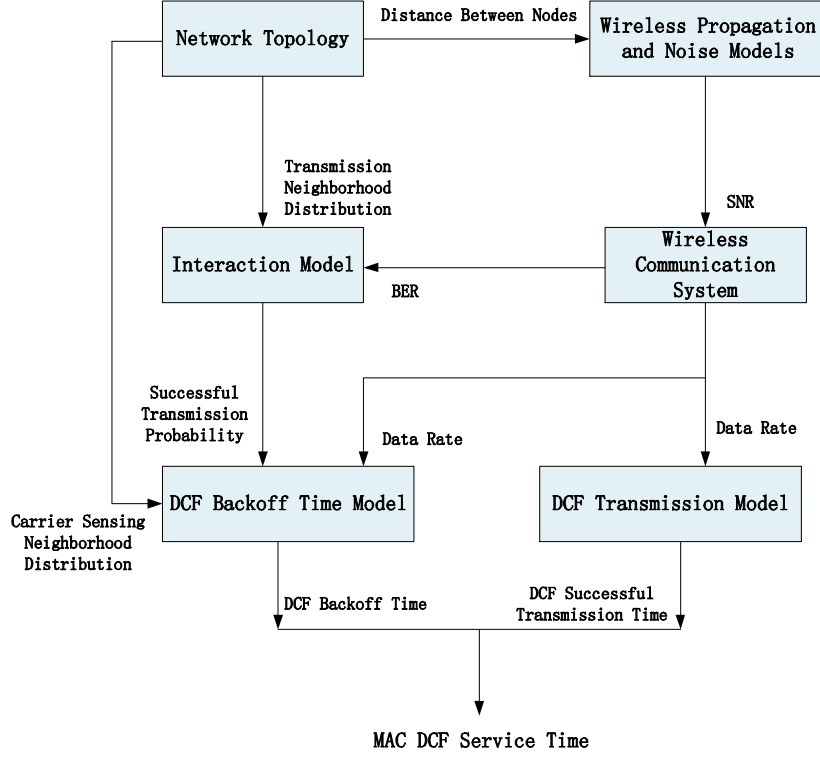


Figure 5.3: Network Flow

where L is the frame length and BEP is bit error probability which could be replaced by the physical-layer parameter, BER . A simple successful handshake process includes both a data packet transmission and a successful ACK packet transmission. On this basis, the successful handshake probability is given, if there exists a successful handshake process between node i and node j .

$$\pi_{ij} = p_f^{i,j} p_f^{j,i} \quad (5.14)$$

where $p_f^{i,j}$ is the successful frame probability of a data packet transmission from node i to node j , and $p_f^{j,i}$ is the successful frame probability of an ACK packet transmission from node j to node i .

Therefore, given a network topology and measured handshake successful probability, the successful transmission probability, q , can be obtained from equations (5.11), (5.12), (5.13) and (5.14). In the following section, different network models are analyzed to study the MAC service time through designing a network flow.

5.4.3 Network Models' Flow

In this section, all network models studied in previous sections and chapters are summarized in order to obtain the service time at the MAC layer. A network flow is designed in figure 5.3, which includes a network topology, a wireless propagation model, a noise model, an interaction model, a wireless communication system model, and a MAC DCF model.

Given a network topology, the location of all nodes is fixed, and therefore the transmission neighbourhood and the carrier sensing neighbourhood of nodes are known. Next, given the transmission power at the source node, P_T , and distance between a node pair, d , the SNR at input of receiver can be measured through the wireless propagation model and the noise model.

As only one of the neighbours within the source node's carrier sensing range is permitted to access the channel under the MAC DCF protocol control, the single user transmission scenario is considered in a wireless communication system. Once the SNR is measured at the receiver, the data rate, r_{data} , and related BER over a transmission link will be known from the wireless communication system model, such as the two-group HSDPA system.

Given the frame size, L , and the data rate, r_{data} , over a transmission link, based on the equation (5.1), the successful transmission time, T_s , in DCF is known. By using the BER over the transmission link and transmission neighbourhood between nodes as the input of the interaction model, the MAC parameter, successful transmission probability, q , is calculated.

Once the successful transmission probability, q , the data rate, r_{data} , and the carrier sensing neighbourhood distribution are known, the average backoff time, T_B , is calculated from the model discussed in the section 5.4.1.

To conclude, the DCF-based service time is obtained by summing of the average backoff time, T_B , and the successful transmission time, T_s .

5.4.4 Examination of the DCF Analysis Model

In [12] Carvalho had examined the accuracy of his DCF analytical model through comparing the results from MATLAB and the simulation results from Qualnet. However, in order to study the MAC DCF-based service time, the accuracy of this analytical models still needs to be examined.

In this section, the NS2 [117] (ns-2.31 version) is used as the network simulator

MAC Layer Parameters	
$W_{\min}$	32
$W_{\max}$	1024
<i>MAC Header (bytes)</i>	34
<i>ACK (bytes)</i>	38
<i>CTS (bytes)</i>	38
<i>RTS (bytes)</i>	44
<i>Idle Slot Time (μsec)</i>	20
<i>SIFS (μsec)</i>	10
<i>DIFS (μsec)</i>	50

Table 5.1: MAC Layer Parameters

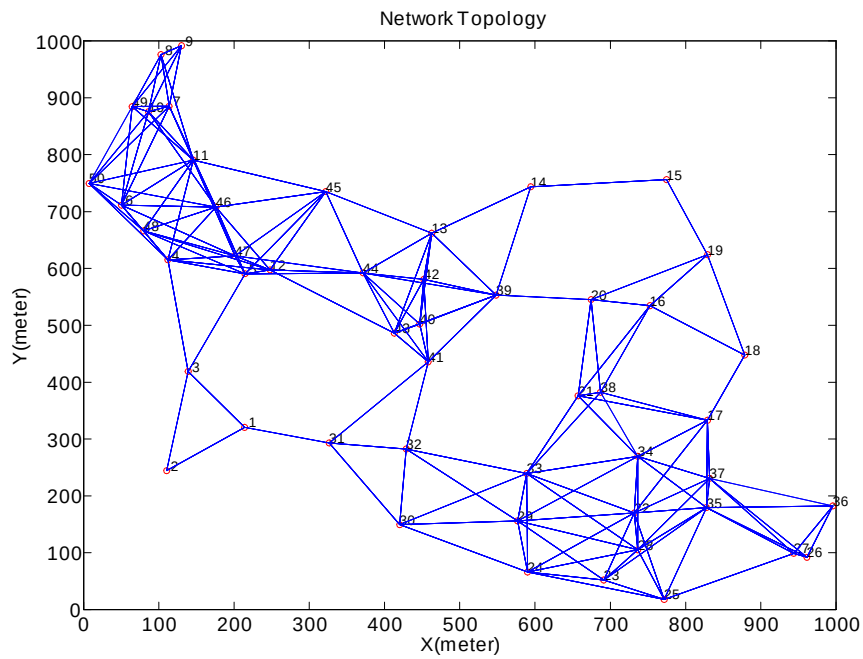


Figure 5.4: A 50-node Network Topology

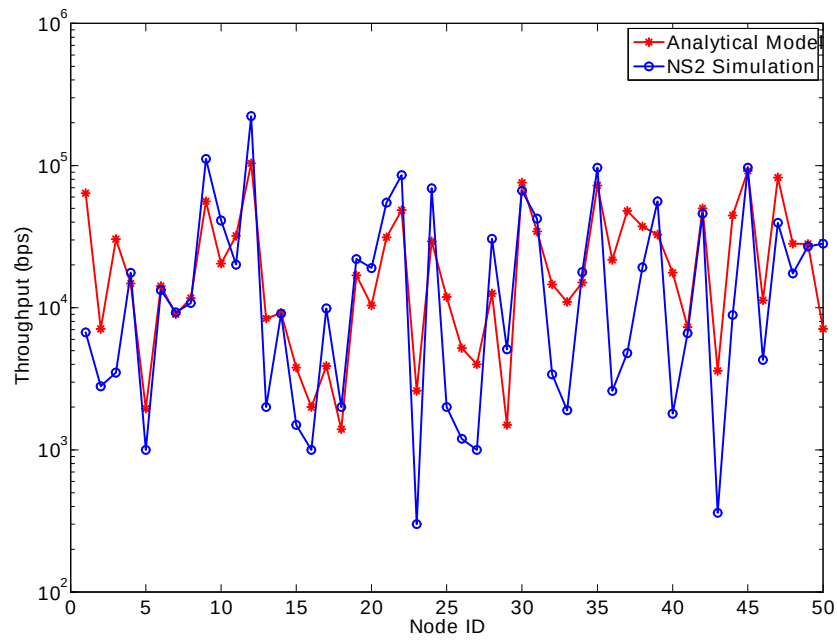


Figure 5.5: The Average Throughput Comparison of NS2 and Analytical Model at Individual Node

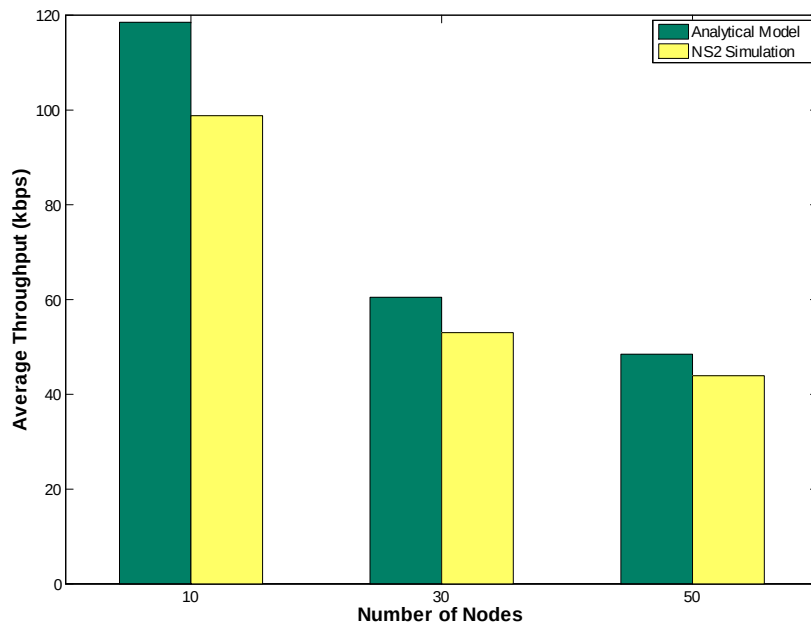


Figure 5.6: The Average Throughput Comparison of NS2 and Analytical Model with Different Network Sizes

and the analytical model is designed in MATLAB (7.2.0 version). All physical-layer and MAC-layer parameters given in table 3.2 and table 5.1 are used for the performance examination and evaluation in the following sections.

Due to the fact that the backoff algorithm is a random process, the experience is repeated for 10 seeds with different initial transmission periods in order to obtain the average throughput in the NS2. The initial transmission period is randomly chosen within the interval $[0 \ 1]$ second, and 500 seconds is set for each simulation running period in order to obtain the average throughput.

Based on the given network topology in figure 5.4, the average throughput generated at the individual node is examined. Figure 5.5 demonstrates the average throughput comparison produced in the MATLAB and in the NS2. The results show a good agreement between the theoretical and simulated throughput measurements, which indicate that the analytical model developed in [12] is accurate enough for modelling the MAC DCF-based service time in large ad hoc networks. Next, the average throughput for different network sizes generated in the MATLAB and the NS2 are examined and compared. The result identifies that the DCF analytical model tends to be more accurate with the increased number of nodes in a network, so it is often made available for use in large ad hoc networks.

5.5 Throughput Enhancement Achieved by Two-group Allocation

In this section, the throughput enhancement achieved by two-group allocation in HSDPA systems at the physical layer is examined. The payload (or data packet) size for an individual node is set at 1kbytes.

The throughput achieved by both single-group and two-group HSDPA systems at each node is plotted in figure 5.7. The result identifies that the two-group HSDPA improves the throughput at nodes compared with the single-group HSDPA. The reason why the two-group HSDPA system enhance the throughput is that two-group allocation provides the data rate enhancement, and therefore significantly reduces the data transmission period, T_{data} . However, in figure 5.7, the throughput achieved at some nodes is the same for both the single-group and the two-group HSDPA systems. The reason is that when the maximum data rate is achieved in the single-group HSDPA system, the two-group allocation is not able to further enhance the data rate in HSDPA systems, hence it cannot improve the throughput in an ad hoc

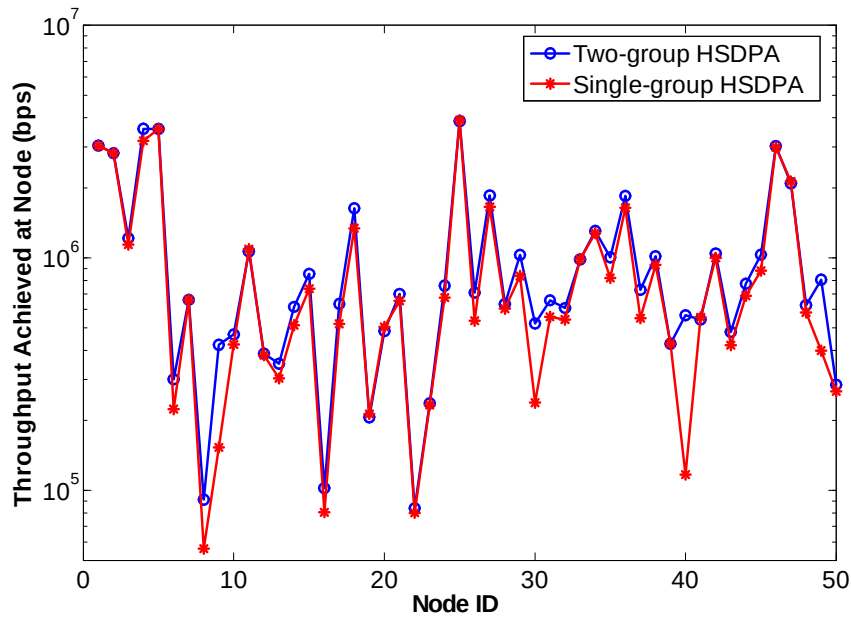


Figure 5.7: Throughput Achieved by Two-group and Single-group HSDPA at Each Node

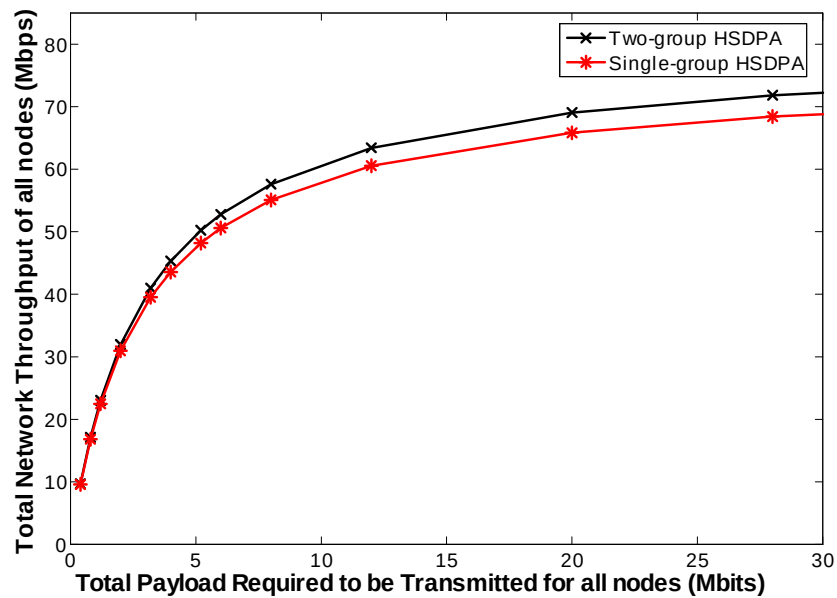


Figure 5.8: The Network Throughput Achieved as a Function of Total Payload

network.

Figure 5.8 demonstrates the total network throughput achieved in a given network topology plotted in figure 5.4. The result identifies that the two-group allocation significantly improves the total network throughput where the payload tends to be larger at nodes. Where the payload is large enough, the data transmission period, T_{data} , will hold a large proportion of the average service time. Accordingly, the two-group allocation approach significantly reduces the average service time when the payload is large, hence it obviously improves the network throughput.

5.6 Simplified MAC DCF Service Time Model

Based on the MAC DCF-based service time study in previous sections, the average service time, $\bar{T}$, is very complex and includes many variable parameters. In order to easily study the average service time in an ad hoc network, the average service time model [12] is simplified in this section.

Based on the equationn (5.5), the average backoff time, $\bar{T}_B$, is simplified by using the following approach.

$$\begin{aligned}\bar{T}_B &= \frac{(t_i p_i + t_c p_c + t_s p_s)(W_{\min} \beta - 1)}{2q} + \frac{1-q}{q} t_c \\ &= \frac{(t_i p_i + t_c p_c)(W_{\min} \beta - 1)}{2q} + \frac{1-q}{q} t_c + \frac{p_s (W_{\min} \beta - 1)}{2q} t_s \\ &= a + b t_s\end{aligned}\tag{5.15}$$

where $a = \frac{(t_i p_i + t_c p_c)(W_{\min} \beta - 1)}{2q} + \frac{1-q}{q} t_c$, $b = \frac{p_s (W_{\min} \beta - 1)}{2q}$, and $t_s = \bar{T}_S$. We then simplify the successful transmission time, $\bar{T}_S$, given in the equation (5.1) as follows.

$$\begin{aligned}\bar{T}_S &= t_{DIFS} + t_{RTS} + t_{CTS} + 3 \times t_{SIFS} + 4 \times \delta + \frac{L}{r_{data}} + t_{ACK} \\ &= c + \frac{L}{r_{data}}\end{aligned}\tag{5.16}$$

where $c = t_{DIFS} + t_{RTS} + t_{CTS} + 3 \times t_{SIFS} + 4 \times \delta + t_{ACK}$

Finally, the service time, $\bar{T}$, can be simplified as a linear function of payload (or data packet) size, L .

$$\begin{aligned}\bar{T} &= \bar{T}_B + \bar{T}_S \\ &= a + b \bar{T}_S + \bar{T}_S \\ &= a + (1+b)c + \frac{1+b}{r_{data}} L \\ &= X \cdot L + Y\end{aligned}\tag{5.17}$$

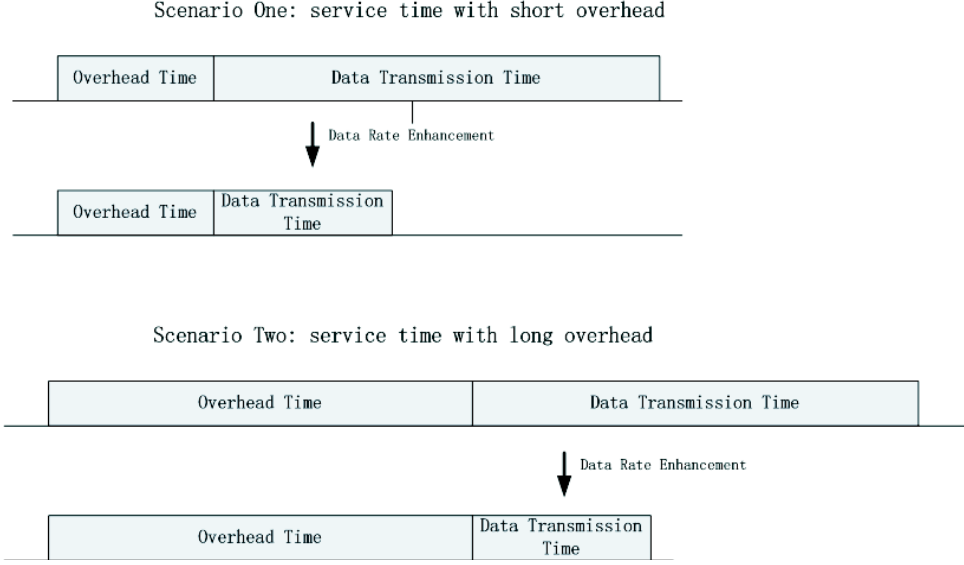


Figure 5.9: Service Time with Short and Long Overhead

where $X = \frac{1+b}{r_{data}}$ and $Y = a + (1+b)c$.

In the following sections, both the network throughput maximization and the end-to-end delay minimization are studied based on this simplified MAC DCF-based service time model given in equation (5.17).

5.7 MAC DCF-based Throughput Enhancement: Adaptive Payload Allocation

5.7.1 Problem Addressed

The throughput is defined as the payload to the related service time based on the equation (1.3). As regards the service time, the data transmission time is the only effective timeslot and others are concluded as the overhead, which results in the gap between the data rate and the throughput. The overhead includes MAC control packets (RTS/CTS/ACK), the header, (MAC/PHY header), the backoff period, and the interframe spaces (DIFS and SIFS). Given the payload (or data packet) size, the high data rate result in short data transmission time and therefore large network throughput (Scenario one in figure 5.9). However, the overhead time may be much longer within the total service time, and therefore the throughput cannot be significantly further improved even if the data rate increases (Scenario two in figure 5.9).

The control packet, which is part of the overhead, affects the service time and causes throughput reduction. In IEEE 802.11, all MAC control packets (e.g. RTS/CTS/ACK) are transmitted with the basic rate (i.e. 1Mbits). However, the payload is transmitted at the data rate, which may be much higher than the basic rate. Due to this fact, in a network with a high data rate, the data transmission time maybe less than the control packet transmission time, and therefore the control packet transmission time dominates the service time and limits the effective throughput.

In order to overcome the overhead effect, large payload size and aggregation schemes are examined in 802.11n standard as the MAC enhancement to improve the throughput [118] in a high data rate network. The concept of frame aggregation is to aggregate multiple payloads into a single transmission, and this reduces the number of access attempts to the medium and therefore significantly increases the throughput.

Here, the payload size allocation is studied to enhance the MAC DCF performance and therefore maximize the overall network throughput, $\sum_{i=1}^N S_i$, in an ad hoc network, when all nodes are active and each node is proposed to transmit a given data packet, while the total number of bits required to be transmitted in an ad hoc network, M_T , is given. It is assumed that the change of payload size in networks does not affect the contention (or backoff process) between nodes in a network.

In the following sections, an optimal solution called the Lagrange optimization approach, is proposed to allocate the payload size at each node in order to maximize the overall network throughput, $\sum_{i=1}^N S_i$. Next, another payload allocation solution known as the fixed payload transmission period approach is developed to provide a simple way to distribute the payload size at nodes and therefore improve the overall network throughput in an ad hoc network.

5.7.2 Lagrange Optimization Approach

In the mathematical optimization, the Lagrange method is used to find the maximum or minimum value of a function subject to certain constraints as follows.

$$\text{maximize } f(x, y) \quad (5.18)$$

$$\text{subject to } g(x, y) = c \quad (5.19)$$

Here, the total amount of information bits required to be transmitted, M_T , is given and equals to the sum of information bits allocated at each node in a given

network, e.g. $\sum_{i=1}^N L_i = M_T$. Here, the objective is to maximize the total network throughput, $\sum_{i=1}^N S_i$. Based on equations (1.3), (5.4) and (5.17), the throughput achieved at each node is a function of corresponding information bits allocated at each node, e.g. $S_i(L_i)$. Based on this, the optimization problem is concluded as follows.

$$\text{maximize } \sum_{i=1}^N S_i(L_i) \quad (5.20)$$

$$\text{subject to } \sum_{i=1}^N L_i = M_T \quad (5.21)$$

By introducing a variable, φ , known as the Lagrange multiplier, the Lagrange function is given as follows.

$$\sum_{i=1}^N S_i + \varphi \left(\sum_{i=1}^N L_i - M_T \right) = 0 \quad (5.22)$$

Let $\gamma = \sum_{i=1}^N S_i + \varphi \left(\sum_{i=1}^N L_i - M_T \right)$

By integrating the payload, L_i , as follows.

$$\begin{aligned} \frac{d\gamma}{dL_i} &= 0 \\ \frac{Y_i}{(Y_i + X_i L_i)^2} + \varphi &= 0 \\ L_i &= \frac{\left(\sqrt{-\frac{Y_i}{\varphi}} - Y_i \right)}{X_i} \end{aligned} \quad (5.23)$$

Here, the payload size at node i , L_i , is a function of Lagrange multiplier, φ , variable X_i and Y_i .

By integrating φ as follows.

$$\begin{aligned} \frac{d\gamma}{d\varphi} &= 0 \\ \sum_{i=1}^N L_i - M_T &= 0 \\ \sum_{i=1}^N \frac{\left(\sqrt{-\frac{Y_i}{\varphi}} - Y_i \right)}{X_i} &= M_T \\ \varphi &= - \left(\frac{\sum_{i=1}^N \frac{\sqrt{Y_i}}{X_i}}{M_T + \sum_{i=1}^N \frac{Y_i}{X_i}} \right) \end{aligned} \quad (5.24)$$

The Lagrange multiplier, φ , is a function of total information bits, M_T , variable X and Y for all nodes. Based on equations (5.23) and (5.24), the payload size

allocated at node i , L_i , is concluded as the function of known total information bits, M_T , variable X and Y for all nodes as follows.

$$L_i = \frac{1}{X_i} \left(\frac{\sqrt{Y_i} \left(M_T + \sum_{i=1}^N \frac{Y_i}{X_i} \right)}{\sum_{i=1}^N \frac{\sqrt{Y_i}}{X_i}} - Y_i \right) \quad (5.25)$$

The variable parameters, X and Y , are mainly related to the backoff process and the control packet (RTS/CTS/ACK) transmissions. Given a network topology and MAC protocol parameters, parameter X and Y will be known from the simplified DCF model discussed in the section 5.6.

Finally, once the total number of information bits required to be transmitted for all nodes in a transmission period, and the variable parameters, X and Y , are known, then the payload size allocated at each node can be calculated from the equation (5.25) and the corresponding throughput achieved at each node will also be obtained from equations (1.3), (5.4), (5.17) and (5.25).

5.7.3 Equal Payload Transmission Period Approach

In the previous section, network throughput maximization was studied through using the Lagrange optimization approach. In this section, the payload transmission period is studied and the related throughput enhancement approach is developed.

Given a data rate, r_{data}^i , for node i to transmit a payload, the payload size equals to the payload transmission period, T_{data}^i , times the data rate, r_{data}^i , as follows.

$$L_i = T_{data}^i \cdot r_{data}^i \quad (5.26)$$

Here, the payload transmission period is assumed to be the same for all nodes, e.g. $T_{data}^1 = \dots = T_{data}^i \dots = T_{data}^N = T_{data}$. As the total information bits required to be transmitted, M_T , is known, the equal payload transmission period is given as follows.

$$\begin{aligned} \sum_{i=1}^N T_{data} \cdot r_{data}^i &= \sum_{i=1}^N L_i \\ T_{data} \sum_{i=1}^N r_{data}^i &= M_T \\ T_{data} &= \frac{M_T}{\sum_{i=1}^N r_{data}^i} \end{aligned} \quad (5.27)$$

Based on the equation (5.27), equal payload transmission period, T_{data} , equals to the average overall network payload transmission period, which is the total payload,

M_T , to the total data rate, $R_T = \sum_{i=1}^N \cdot r_{data}^i$, in a given network. Once the payload transmission period, T_{data} , is known, the payload size allocated at node i is simply given as follows.

$$L_i = T_{data} \cdot r_{data}^i = \frac{M_T \cdot r_{data}^i}{\sum_{i=1}^N \cdot r_{data}^i} \quad (5.28)$$

The equal payload transmission period approach provides a simple solution to find the payload size allocated at nodes in a given network, as the parameters, which are only required to be known, include the data rate achieved at each active transmission link.

5.7.4 Performance Evaluation

In this section, the Lagrange optimization approach is compared with the fixed payload transmission period approach through examining their throughput enhancement performance. Next, the throughput enhancement achieved by both adaptive payload allocation at the MAC layer and two-group allocation at the physical layer are examined. Here, different total payloads, which are required to be transmitted in a transmission period, are allocated at all nodes in a given network topology as in figure 2.2. Each node randomly chooses a neighbouring node as its destination node for transmissions.

5.7.4.1 The Lagrange Optimization Approach versus the Equal Payload Transmission Period Approach

Figure 5.10 gives the total network throughput achieved at all nodes as a function of the total given payload. The result obviously shows that both the Lagrange optimization approach and the equal payload transmission period approach improve the total network throughput compared with the equal payload allocation (which allocates the same payload size at each node).

From the result, it is clear that the Lagrange optimization approach and the equal data transmission period approach achieve almost the same performance. On this basis, it means that the Lagrange optimization approach also achieve an equal payload transmission period at nodes through adjusting the payload size.

5.7.4.2 MAC enhancements on Network Throughput

Unlike the two-group allocation, the adaptive payload allocation results in throughput reduction at some nodes (Figure 5.11). However, the adaptive payload allocation

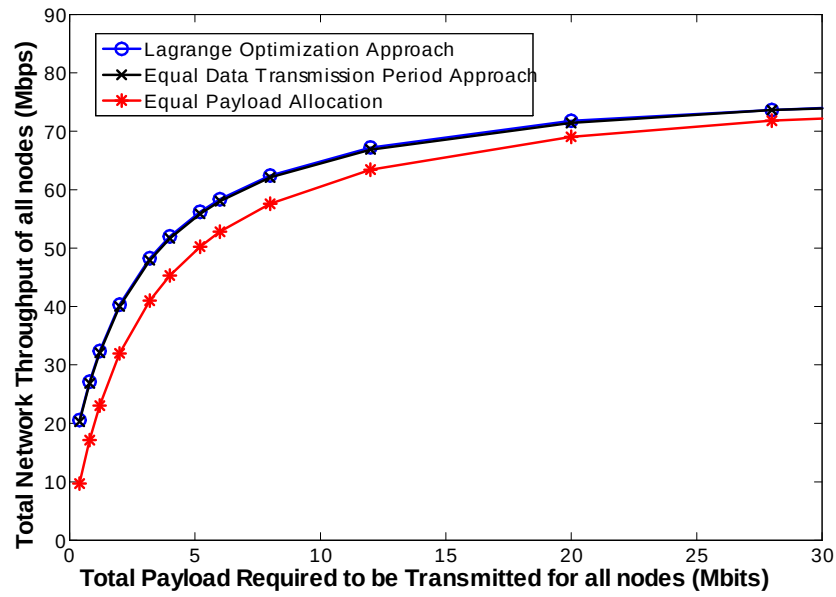


Figure 5.10: Lagrange Optimization Approach vs Equal Data Transmission Period Approach

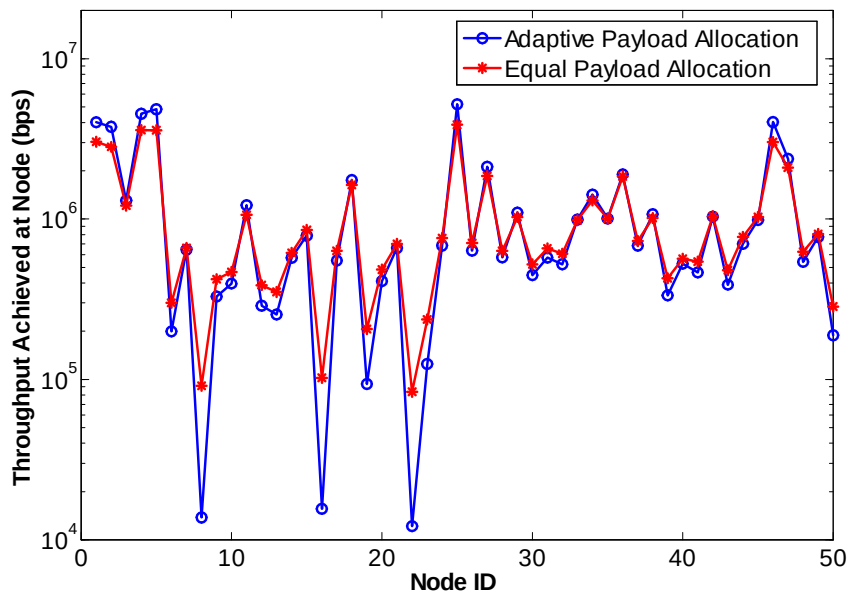


Figure 5.11: Throughput Achieved by Adaptive and Equal Payload Allocation at Each Node

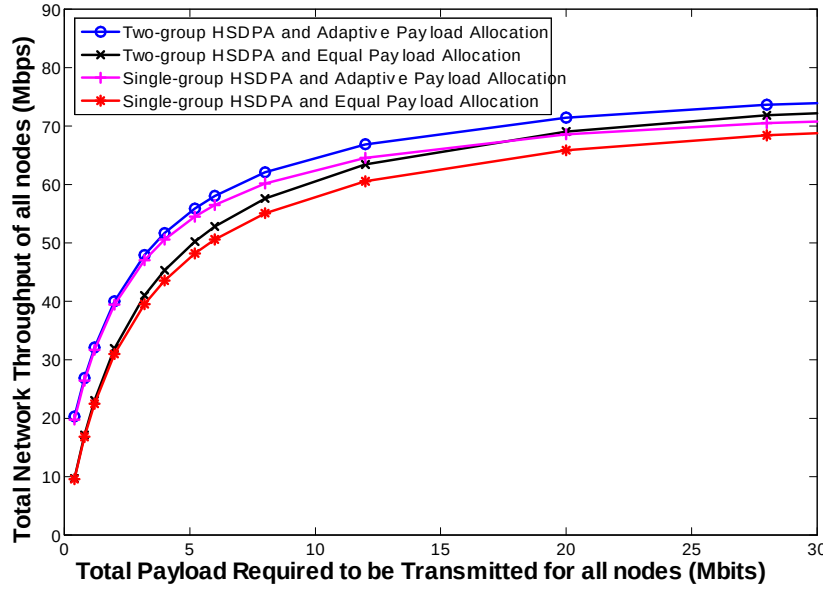


Figure 5.12: Total Network Throughput as a Function of Total Payload Required to be Transmitted for All Nodes

significantly enhances the total network throughput through distributing given total payload at nodes, especially within the small payload range. When the total payload tends to be larger, the throughput enhancement achieved by the adaptive payload allocation will not be obvious.

Figure 5.12 demonstrates the total network throughput of all nodes achieved by different schemes, such as the two-group HSDPA and the adaptive payload allocation. The results in Figure 5.12 identify that both the two-group allocation (the black line) and the adaptive payload allocation (the magenta line) improve the total network throughput of all nodes in a given network.

Due to the fact that the adaptive payload allocation (the magenta line) significantly enhances the total network throughput within the small payload range, and two-group allocation (the black line) significantly improves the total network throughput within large payload range, the combination of the adaptive payload allocation and the two-group allocation (the blue line) further enhances the total network throughput in any payload range. Compared with the original single-group HSDPA with the equal payload allocation, the combination solution enhances the total network throughput by 20% to 100%.

5.8 MAC DCF Based End-to-End Delay Minimization

In the previous section, the overhead issue is addressed and the related throughput enhancement is studied through adjusting payload size at nodes under given total amount of bits in a network. Different payload sizes are allocated at nodes based on the physical (i.e. data rate) and MAC parameters (i.e. transmission probability). However, when the routing scheme is considered in a network, the payload size should be kept the same at nodes which are along the routing path. Based on the energy-aware routing scheme and load balancing approach developed in chapter 4, the payload size allocation and related transmission management for a routing path will be studied in this section in order to achieve low end-to-end delay.

5.8.1 Problem Addressed

Given a total amount of bits required to be forwarded from the source to the destination, the end-to-end delay equals to the time consumed starting from the first packet departures at the source node lasting until the last packet has arrived at the destination node. If the transmission is continuous for a routing path (or full load routing path), the data packet will be transmitted and will not need to wait for the previous data packet to arrive at the destination

When a number of data packets are continuously transmitted from the source to the destination, small payload size results in high frequent transmissions but small service time at links along the routing path, and large payload size results in less frequent transmission but long service time at each link. Both of these scenarios may cause long end-to-end delays for a given routing path.

On this basis, given the amount of information bits required to be transmitted along a routing path, the problem becomes how to efficiently manage the data packet transmissions for a full load routing path.

5.8.2 Network Model

In section 4.4, given the total M bits required to be transmitted, a loading balance solution (equation (4.17)) is developed for energy-aware disjoint multipath routing. Once the bits allocated to each routing path is known, the end-to-end delay along the related routing path is studied and a payload management solution is proposed

to minimize the end-to-end delay.

Here, the continuous packet transmission is assumed. When the known total m_J bits are allocated to a routing path J , the end-to-end delay, T_{EtE}^J , can be simply modelled as follows.

$$T_{EtE}^J = \sum_{i \in J} \bar{T}_i^J + \left(\frac{m_J}{L_J} - 1 \right) \bar{T}_{\max}^J \quad (5.29)$$

where $\bar{T}_i^J$ is the average service time for link i along the routing path J , L_J is the payload size and $\bar{T}_{\max}^J$ is the maximum average service time for the related link along the routing path J . $\sum_{i \in J} \bar{T}_i^J$ is the delay for transmitting the first packet from the source node to the destination node. $\frac{m_J}{L_J} - 1$ means the rest of the number of data packets except the first data packet. As the average service time modelling based on the MAC DCF mechanism, the transmission period difference between two adjacent data packets is given by the maximum average service time along a routing path J . $\left(\frac{m_J}{L_J} - 1 \right) \bar{T}_{\max}^J$ represents the delay that results as a consequence of the rest of the being data packets transmitted. The sum of these two delays is the end-to-end delay for continuously transmitting total m_J bits (or total $\frac{m_J}{L_J}$ data packets) along the routing path J .

5.8.3 End-to-End Delay Minimization Solution

In order to find a solution to minimize the end-to-end delay, the equation (5.29) is simplified by integrating the simplified average service time expression (equation (5.17)) as follows.

$$\begin{aligned} T_{EtE}^J &= \sum_{i \in J} (X_i^J \cdot L_J + Y_i^J) + \left(\frac{m_J}{L_J} - 1 \right) (X_{\max}^J \cdot L_J + Y_{\max}^J) \\ &= L_J \cdot \sum_{i \in J} X_i^J + \sum_{i \in J} Y_i^J + m_J \cdot X_{\max}^J + \frac{m_J \cdot Y_{\max}^J}{L_J} - X_{\max}^J \cdot L_J - Y_{\max}^J \\ &= L_J \cdot \sum_{i \in J, i \neq \max} X_i^J + \sum_{i \in J, i \neq \max} Y_i^J + m_J \cdot X_{\max}^J + \frac{m_J \cdot Y_{\max}^J}{L_J} \\ &= q_J \cdot L_J + n_J + \frac{p_J}{L_J} \end{aligned} \quad (5.30)$$

where $q_J = \sum_{i \in J, i \neq \max} X_i^J$, $n_J = \sum_{i \in J, i \neq \max} Y_i^J + m_J \cdot X_{\max}^J$, and $p_J = m_J \cdot Y_{\max}^J$. By double integrating the equation (5.30) and try to find the extreme value as

follows.

$$\begin{aligned}\frac{\partial T_{EtE}^J}{\partial L_J} &= 0 \\ q_J - \frac{p_J}{L_{J,0}^2} &= 0 \\ L_{J,0} &= \pm \sqrt{\frac{p_J}{q_J}}\end{aligned}\tag{5.31}$$

As the payload size $L_J > 0$, equation (5.30) achieves the extreme value when

$$L_{J,0} = \sqrt{\frac{p_J}{q_J}}\tag{5.32}$$

Next, do double integral for equation (5.30) as follows.

$$\frac{\partial^2 T_{EtE}^J}{\partial L_J} = \frac{2p_J}{L_J^3} > 0\tag{5.33}$$

Based on the equation (5.33), this working identifies that the end-to-end delay for routing path J , T_{EtE}^J , achieves the minimum value when the optimal payload size, $L_{J,0}$ is given as follows.

$$L_{J,0} = \sqrt{\frac{m_J \cdot Y_{\max}^J}{\sum_{i \in J, i \neq \max} X_i^J}}\tag{5.34}$$

$$\begin{aligned}T_{EtE, \min}^J &= 2\sqrt{p_J q_J} + n_J \\ &= 2\sqrt{m_J \cdot Y_{\max}^J \cdot \sum_{i \in J, i \neq \max} X_i^J} + \sum_{i \in J, i \neq \max} Y_i^J + m_J \cdot X_{\max}^J\end{aligned}\tag{5.35}$$

5.8.4 Performance Evaluation

In this section, the performance of proposed end-to-end delay minimization solution is examined. The network topology given in figure 2.2 and the related four disjoint routing paths discovered in figure 4.13 are used. Furthermore, the total amount of information bits required to be transmitted from the source to the destination node is given, $M = 3.55$ megabits. Based on the load balancing approach developed in the equation (4.17), the information bits allocated to four routing paths are as follows: $m_1 = 800kbits$, $m_2 = 858kbits$, $m_3 = 912kbits$, and $m_4 = 985kbits$. The given amount of bits, which is required to be transmitted along each routing path, is divided into a number of data packets for transmissions, e.g. $\frac{m_J}{L_J}$, where L_J is the payload size for routing path J and $J \in [1, 2, 3, 4]$.

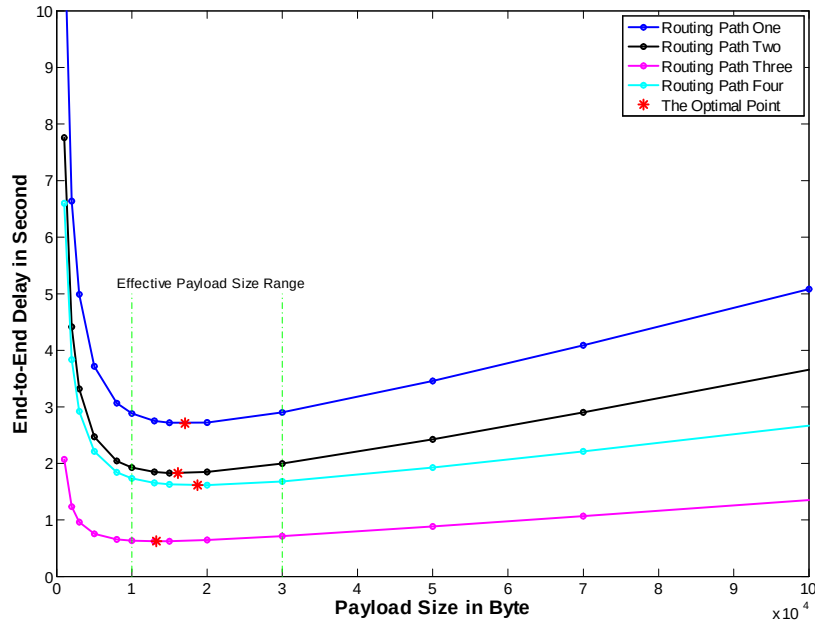


Figure 5.13: End-to-end Delay as a Function of Data Packet Size

The result in figure 5.13 demonstrates the end-to-end delay for four routing paths given in figure 4.13, and it identifies that short payload size results in long end-to-end delay due to high frequent transmissions. Although a large payload size achieves low frequent transmissions, it also results in long service time at each link along the routing path; hence it produces lengthy end-to-end delays. By using the proposed end-to-end delay minimization approach, the optimal payload size, which results in the minimum end-to-end delay for a given routing path, is easily found. Taking the first routing path as an example, the optimal payload size is about 17kbytes and about six data packets are transmitted to achieve a minimum end-to-end delay, $T_{EtE,min}^1 = 2.7 \text{ second}$.

An additional point to consider consists in the comparison of the end-to-end delay achieved by the two-group and the single-group HSDPA. The finding demonstrated in figure 5.14 clearly shows that the two-group allocation results in shorter end-to-end delays due to shorter service time. Finally, the result in figure 5.15 identifies that the optimal payload size increases with the increase of the total bits required to be transmitted.

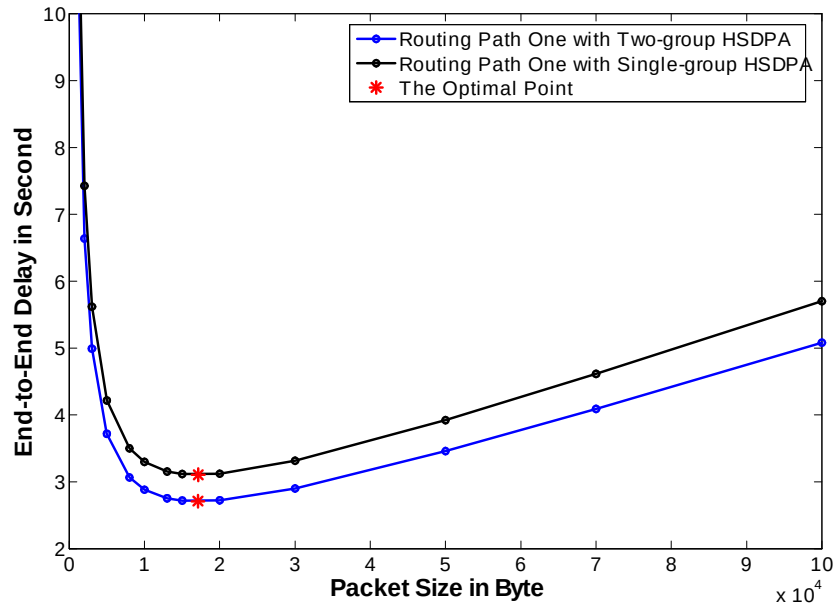


Figure 5.14: Comparison of End-to-end Delay for Two-group and Single-group HSDPA

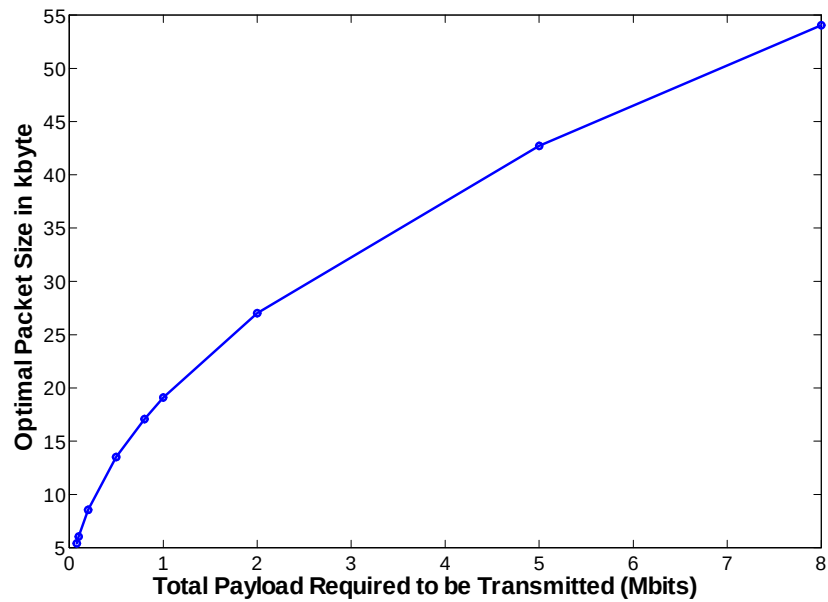


Figure 5.15: Optimal Data Packet Size as a Function of Total Payload

5.9 Conclusion

In this chapter, the MAC DCF-based service time in an ad hoc network is studied and the DCF model developed in [12] is simplified for further study. Here, two main problems are addressed comparison of the overhead effect on the network throughput and the end-to-end delay minimization.

The data rate enhancement developed in chapter 3: two-group allocation, is examined in an ad hoc network to identify its related network throughput enhancement. Next, the overhead issue, which results in the throughput reduction at the MAC layer, is analyzed and addressed. In order to achieve an efficient transmission and to improve the network throughput in a multi-rate ad hoc network, two adaptive payload allocation approaches are developed.

Given a network topology and the total amount of information bits that need to be transmitted, a Lagrange optimization solution is proposed to efficiently allocate payload at nodes to overcome the overhead effect, and therefore significantly enhance the overall network throughput. Another equal payload transmission period approach not only provides a simple payload allocation solution, but also performs just as well as the Lagrange optimization approach. From the results, both the two-group allocation (physical enhancement) and the adaptive payload allocation (MAC enhancement) improve the network throughput. Unlike two-group allocation, adaptive payload allocation results in throughput reduction at some nodes, but still significantly enhances the network throughput.

Once the information bits are allocated at each routing path based on the load balancing approach developed in chapter 4, the end-to-end delay consumed for the routing path is studied. A payload transmission management problem is addressed and discussed. When a number of data packets are continuously transmitted from the source to the destination, small payload size results in high frequent transmissions and the large payload size results in long service times at each link. Both of these scenarios may cause long end-to-end delays for a given routing path.

On this basis, an optimal payload size allocation is developed to provide an end-to-end delay minimization solution. Given the total amount of information bits required to be transmitted along a routing path, the end-to-end delay is examined as a function of payload size, and the results identify that the proposed solution can easily and accurately find the optimal payload size which results in the least end-to-end delay for a given routing path.

Chapter 6

Summary and Future Work

In this chapter, this thesis is summarized and some open questions and future directions are considered for the purpose of ongoing research.

6.1 Thesis Summary

This thesis studies the resource allocation in ad hoc networks, and it focuses on the physical, MAC and network layers. Unlike the centralized network, the ad hoc network does not have any central administration and the energy is constrained, e.g. the ad hoc sensor network, so the resource allocation plays very important role in efficiently managing the limited energy in ad hoc networks.

At the physical layer, the multi-code CDMA is a novel technology used to provide high data rates to satisfy the multimedia communication requirements. HSDPA, which is the application of multi-code CDMA, has been applied as the enhancement version of WCDMA in 3G mobile communication systems. However, there are only three options of the number of parallel channels for transmission, e.g. $K = 5, 10$ or 15 . In addition, the same modulation scheme is used for all parallel channels during one transmission. Both of these limitations result in residual energy and data rate reduction in HSDPA systems.

In order to solve this inefficient energy utilization problem, two approaches are developed: channel adaptive HSPDA and two-group HSDPA. In channel adaptive HSDPA systems, the constraint of the number of channels used for transmissions is eased. The number of channels used for transmissions is adjusted based on the channel condition (or received SNR). From the results, this approach significantly reduces the residual energy and enhances the data rate within low SNR range in

HSDPA systems.

However, the residual energy is still large within the high SNR range in HSDPA systems as the channel adaptive approach still applies the same modulation scheme for all parallel channels. Based on the total measured SNR at the receiver, two-group HSDPA allocates bits into two groups of channels with different modulation schemes (e.g. QPSK, 16QAM or 64QAM). This approach has been examined in [9] and the results identify that two-group allocation significantly reduces the residual energy upper bound within both low and high SNR ranges in HSDPA systems. Furthermore, the data rate achieved by two-group allocation is much closer to the system capacity in HSDPA systems.

After studying resource allocation techniques to efficiently utilize energy and enhance data rates in HSDPA systems at the physical layer, this thesis then focuses on the network layer and related energy-aware routing scheme [10] [11]. By considering the physical layer parameters: the transmission power, P_T , and the data rate, r_{data} , the energy consumed per bit is modelled as the cost function to select the related routing paths. A trellis-hop diagram approach is developed to find all possible routing paths for a given node pair in an ad hoc network. However, this trellis-hop approach has large computational complexity and it is not available for routing paths discovery in large networks. In order to reduce the computational complexity of the routing path discovery process, a modified Viterbi algorithm is developed. During the routing path discovery process, the modified Viterbi algorithm groups the uncompleted routing paths which pass through the same link (known as link-joint routing path), and it selects the link-joint routing path with the minimum cost and deletes other link-joint routing paths from the group. A "hop level checking" approach is studied to implement the modified Viterbi algorithm in routing path discovery process. On this basis, an energy-aware single-path routing (EASR) scheme and an energy-aware disjoint multipath routing (EADMR) scheme are proposed in ad hoc networks.

Furthermore, given the amount of information bits required to be transmitted through multipath routing in a network, a load balancing approach is developed to further reduce the energy consumption. Through comparing EASR with AODV and EADMR with AOMDV, the results identify that the proposed energy-aware routing scheme, EASR and EADMR, consume much less energy when transmitting the same amount of information bits for a node pair in an ad hoc network. Furthermore, the results also show the further energy consumption reduction achieved by load

balancing for multipath routing schemes, such as EADMR and AOMDV.

Finally, this thesis studies the DCF service time at the MAC layer. By investigating the MAC DCF model developed in [12], the throughput enhancement achieved by two-group allocation in HSDPA systems is examined. The result shows that two-group allocation significantly improves the network throughput due to its data rate enhancement at the physical layer.

However, although the data rate (which is the upper bound of network throughput) is improved at the physical layer, the overhead at MAC layer results in inefficient transmissions and throughput reduction. In order to overcome the overhead effect, two adaptive payload allocation approaches are developed. One payload allocation solution is proposed to maximize the overall network throughput by introducing Lagrange optimization. Another payload allocation solution is developed through allocating payload at individual node to achieve equal data transmission periods for all nodes.

The results show that equal data transmission period approach not only achieves similar performance to Lagrange optimization approach, but it also provides a simple payload allocation solution. The results compare the throughput achieved by both adaptive payload allocation and equal payload allocation, and identify that adaptive payload allocation approach significantly improves the network throughput. As both two-group allocation and adaptive payload allocation provide throughput enhancement, the combination of them shows greater and further throughput enhancement. Finally, after allocating amount of information bits to one of multiple routing paths based on the load balancing approach, the end-to-end delay along this routing path is studied by considering the MAC DCF service time. An end-to-end delay minimization approach is developed through finding the optimal payload size for transmissions.

6.2 Future Work

The resource allocation studies presented in this thesis fall into three main areas: (1) energy utilization and data rate enhancement in HSDPA systems at the physical layer; (2) energy-aware routing and load balancing studies at the network layer; and (3) DCF based payload allocation at the MAC layer. Regarding future work, there are some potential open questions relating to each of these areas.

Regarding the physical layer work, to build upon results of this thesis, the two-

group allocation approach has been identified to efficiently utilize the energy in HSDPA systems and improve the data rate. One potential future avenue of work will be the examination of data rate enhancement in a real two-group HSDPA system, which could be implemented in a hardware testbed, such as FPGA. One FPGA board is required to play the roles of transmitter and receiver in a wireless communication system. The two-group HSDPA system will be designed and coded in FPGA boards by using high-level (C/C++) or low-level (VHDL) programming languages. Regarding the wireless propagation transmission, either radio or acoustic communications will be available to the testbed. The radio communication is closed to the real wireless transmissions, but it is different in term of design and implementation. Acoustic communication is cheap/cost-efficient and simple to implement, but it only provides a very narrow bandwidth due to slow sound speed.

To date, the multipath effect for two-group HSDPA systems have been studied in [119] [120] [121], and the coding schemes in two-group HSDPA systems have also been investigated in [122] [123]. However, many aspects have not been considered and examined in existing two-group HSDPA systems. Currently, MIMO technology is a hot topic as regards improving the upper bound capacity of wireless communication systems by using multiple antennas at both the transmitter and the receiver. In addition, MIMO technology is considered as one option in the advanced version of HSDPA standard for future 3G mobile communications. Taking this into account, the investigation for two-group MIMO HSDPA systems will be further studied based on existing finding.

Another area to consider is that, as two-group allocation is a mathematical concept used to implement adaptive modulation in multi-tone transmissions, further effort will be made to apply and examine two-group allocation in OFDM, which divides data stream into multiple sub-carriers. Furthermore, as MIMO OFDM is defined in Long Term Evolution (LTE), which is a future 4G communication system candidate, two-group allocation in MIMO OFDM systems will be of further interest as far as the direction of future work is concerned.

In this thesis, the proposed routing scheme at the network layer is only examined in the network environment designed in MATLAB. More accurate simulation results are expected to be obtained from the network simulator (e.g. NS2). However, the energy consumption is difficult to be measured at nodes in the network simulator. Another solution is to implement a proposed energy-aware routing scheme in a network testbed, which consists of a number of laptops or sensors. As the energy

resource for laptops and sensors are batteries, the consumed and remaining energy in batteries are straight forward to be measured.

Another direction for future research could be the computational complexity reduction of the routing discovery process. In this thesis, a "hop level checking" method is developed to implement the trellis-hop diagram and modified Viterbi algorithm for the energy-aware routing path discovery process. Although some initial computational complexity work has been done in [124] for proposed energy-aware routing scheme, further computational complexity reduction work will focus on simplified routing tables or cross hop level checking. Furthermore, the network lifetime will be another research topic for studying to extend the network lifetime by using proposed energy-aware routing. Some initial work and results can be consulted in [10].

Regarding the MAC DCF service time study, further simulation work will be done in a network simulator environment, e.g. NS2, to examine the throughput enhancement achieved by both two-group allocation and adaptive payload allocation. In order to generate a multi-rate network in NS2, the payload size could be adjusted. By studying the "bottleneck" along the routing path and related maximum MAC service time, the MAC-aware routing scheme will also be investigated in order to find the routing path with the maximum throughput. Delay-aware routing schemes could be another topic on the routing paths discovery to achieve small end-to-end delay. A more complex and accurate end-to-end delay model for routing paths is also proposed to be developed.

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