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Efficient Traffic Control Mechanisms for Multi-service Wireless Networks

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*To my wife, Qi Wang,
and my parents, Zhongli Long and Shumei Li*

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Table of Contents

Acknowledgements iv

Table of Contents iv

Summary viii

List of Figures xi

List of Tables xiii

List of Abbreviations xiv

List of Mathematical Symbols xvii

Chapter 1 Introduction 1

 1.1 Applications and Service Requirements of Wireless Networks 1

 1.2 Motivations of the Research 4

 1.3 Research Objectives 7

 1.3.1 New Packet Scheduling Schemes for Wireless Networks 8

 1.3.2 Joint Admission Control/Packet Scheduling Framework for Wireless Networks 8

 1.3.3 Optimal Congestion Control for Multi-hop Wireless Networks 8

 1.4 Major Contributions of the Thesis 9

 1.5 Thesis Organization 11

Chapter 2 Background 12

 2.1 Quality of Service (QoS) in Wireless Networks 12

 2.2 Wireless Packet Scheduling 14

 2.3 Admission Control in Wireless Networks 21

 2.4 Congestion Control in Multi-hop Wireless Networks 23

**Chapter 3 A Channel-State-Aware BER Scheduling Scheme for Wireless Mul-
timedia Access Control Protocol 25**

 3.1 Introduction 25

 3.2 Channel-State-Aware WISPER (CSA-WISPER) 26

 3.2.1 Channel State Model 27

 3.2.2 CSA-WISPER mechanisms 27

 3.2.3 Operation Example of CSA-WISPER 29

 3.3 Simulation and Numerical Results 32

3.3.1	Simulation Scenario 1	32
3.3.2	Simulation Scenario 2	34
3.4	Conclusion	38
Chapter 4 Channel States Dependent Fair Service: A New Packet Scheduling		
	Algorithm for CDMA	39
4.1	Introduction	40
4.2	System Model and Theoretical Analysis	42
4.2.1	Proposed CDMA Scheduling Model	42
4.2.2	Multiplexing Schemes in High-Data Rate CDMA: TDM versus CDM	43
4.3	Channel States Dependent Fair Service for CDMA (CSDFSC)	48
4.4	Simulation Results	52
4.5	Conclusion	60
Chapter 5 Efficient Rate Guaranteed Opportunistic Scheduling for Wireless		
	Networks	62
5.1	Introduction	62
5.2	System Model	66
5.2.1	System Performance Value	66
5.2.2	GR Node with Loss	67
5.2.3	Problem Formulation	69
5.3	Rate-guaranteed Opportunistic Scheduling	70
5.3.1	Rate-guaranteed Opportunistic Scheduling	70
5.3.2	Two Users Case	71
5.3.3	Multiple Users Case	73
5.3.4	Implementation Consideration	76
5.3.5	Computational Complexity	77
5.4	Optimality and Delay Bound of ROS	78
5.4.1	Proof of Optimality	78
5.4.2	Delay Bound	79
5.4.3	Properties of $D(k)$	80
5.4.4	Packet Loss Probability in ROS	81
5.5	Behaviors and Performance of ROS	81
5.5.1	Fairness	82
5.5.2	Delay Bound	84
5.5.3	Optimality of Throughput	85
5.5.4	Packet Loss Rate	86
5.6	Conclusion	86
Chapter 6 A New Joint Framework of Packet Scheduling/Admission Control		
	for Multi-service Wireless Networks	88
6.1	Introduction	88
6.2	System Model	91
6.2.1	Channel Capacity	92
6.2.2	Traffic Model	92
6.2.3	Channel Model	93
6.3	Rate Guaranteed Scheduling (ROS)	93
6.4	New Admission Control Policy— ROS-CAC	96

6.4.1	Analysis of PLR	96
6.4.2	Admission Control Policy	99
6.4.3	Example of Admission Control Operation	100
6.4.4	Discussions	101
6.5	Performance Evaluation	103
6.5.1	Video Traffic Scenario	103
6.5.2	Mixed Traffic Scenario	104
6.6	Conclusion	106
Chapter 7 Optimizing Congestion Control for Multi-hop Wireless Networks . .		109
7.1	Introduction	110
7.2	Congestion Control for WMN	112
7.3	System Model	114
7.3.1	Multi-channel WMN	115
7.3.2	Time Constraint	116
7.3.3	Utilization Function	118
7.4	Proposed Congestion Control Algorithm	119
7.4.1	Problem Formulation	120
7.4.2	End-to-end Algorithm	121
7.4.3	Price Function	124
7.4.4	Hop-by-hop Congestion Control Algorithm	125
7.5	Performance Evaluation	127
7.5.1	Simulation Scenario	127
7.6	Conclusion	131
Chapter 8 Conclusions and Future Work		134
8.1	Conclusions	134
8.2	Future Work	137
Author's Publications		139
Bibliography		140

Summary

Broadband wireless communication networks will be an important part of the global communication infrastructure in the future. With the growing deployment of wireless data services and the increasing demand for multimedia applications, it is highly expected that future wireless networks will provide services for heterogeneous classes of traffic, and satisfy their different quality of service (QoS) requirements. Thus the importance of traffic control mechanisms to support QoS for future multi-service wireless network becomes apparent. Because of the unique characteristics of wireless channel, such as location dependent channel error and time varying channel states, the problems faced by future wireless traffic control mechanisms are greatly different from traditional wireline networks and much more challenging. In this Ph.D research project, we address some key issues in traffic control including packet scheduling, admission control and congestion control mechanisms for multi-service wireless networks.

As the first step of the research work on packet scheduling in wireless networks, we propose and evaluate a new scheme, called Channel-State-Aware wireless multimedia access control protocol with BER(bit error rate) scheduling (CSA-WISPER), for Code-Division Multiple Access (CDMA) networks. The basic idea behind CSA-WISPER is to exploit wireless channel states to improve channel utilization by combining WISPER with error-deal mechanisms. We design a number of error-deal mechanisms with different complexity and effectiveness. We use simulations based on ns-2 to evaluate the performance of the proposed mechanisms. The numerical results show that combining WISPER and error-deal mechanisms can achieve both higher throughput and lower delay, compared with the original WISPER.

Next, we focus on packet scheduling for high speed CDMA (CDMA/HDR) networks. First, we investigate the appropriate multiplexing method for CDMA/HDR networks. Based on the proposed CDMA scheduling model, we compare Time-Division Multiplexing (TDM) with Code-

Division Multiplexing (CDM). According to our mathematical analysis, TDM instead of CDM should be used in most situations. In other words, one user transmitting in a timeslot at full rate is more efficient than multiple users transmitting at low rates in a timeslot. Based on this finding, we propose a new packet scheduling scheme, namely Channel States Dependent Fair Service for CDMA (CSDFSC), which is designed specially for high data rate CDMA network. CSDFSC tries to emulate general processing sharing (GPS) to achieve fairness between best-effort traffic flows. Flow weights are used to differentiate guaranteed and best-effort traffic flows. The numerical results show that CSDFSC outperforms WISPER in terms of link resource utilization, packet loss ratio, fairness between best-effort traffic flows, etc. Another notable feature of CSDFSC is the implementation complexity, which is much lower than that of WISPER.

Furthermore, we develop an efficient Rate-guaranteed Opportunistic Scheduling (ROS) scheme for wireless networks. By exploiting time-varying channel states, ROS can optimize system throughput while satisfying the “fairness” constraint of resource allocation. We prove the optimality of the proposed scheduler by mathematical analysis and the computational complexity of ROS is lower than that of existing opportunistic scheduling algorithms. Moreover, we define a new concept of Guaranteed-Rate (GR) node with loss, which is extended from the “service curve with loss” model, to analyze the performance bounds provided by ROS, such as the statistical end-to-end packet delay bound. The numerical results validate our mathematical analysis and demonstrate the improvement of throughput performance.

Besides ROS, we further develop a new admission control algorithm which works in conjunction with ROS. By sharing traffic information and working coordinately, these two schemes can provide statistical QoS guarantee in multi-service wireless networks. In the proposed admission control policy, packet loss rate (PLR) is used as the admission criteria. Based on our proposed mathematical model, we derive the probability distribution function (PDF) of queue length under ROS and deduce the PLR for individual flows. When a new incoming flow arrives, the PLRs of existing flows are re-calculated. If the PLR requirements of all flows (including the new flow) are satisfied, this new incoming flow is admitted; otherwise it is rejected. Our simulation results validate the effectiveness of the proposed admission control policy.

Finally, we study the congestion control for multi-hop wireless networks. As a kind of multi-hop

wireless networks, wireless mesh network (WMN) provides a new network access solution for both mesh and conventional mobile clients. As an access network solution, WMN will be integrated with various networks (e.g. Internet, cellular, IEEE 802.11, sensor networks, etc.). The integration of WMNs with other networks requires an effective and robust congestion control mechanism to maintain network stability and achieve high network resource utilization. In this thesis, we consider congestion control for WMNs with multi-channel MAC, where multiple non-overlapped channels are used by each node. We formulate the design of congestion control mechanism for such a system as a utilization maximization problem subject to time constraint. As the solution of this constrained maximization problem, an end-to-end congestion control algorithm is first developed. Based on that, a distributed adaptive algorithm with hop-by-hop congestion information feedback is developed. We prove the convergence and stability of this new algorithm by both theoretical analysis and simulation results. We find that the proposed hop-by-hop algorithm has almost the same stability as the end-to-end algorithm, but significantly outperforms the latter in terms of queue length and packet delay.

The findings and achievements of this Ph.D research can be used to design traffic control mechanisms for future wireless networks to efficiently provide multi-service for emerging applications. The analytical models we establish in this thesis will also provide very useful reference for further research in this area.

List of Figures

2.1	Unified wireless fair queuing framework	16
3.1	GE channel model	27
3.2	Flow chart of CSA-WISPER scheme	29
3.3	Packet allocation of CSA-WISPER	31
3.4	Error-deal scheme in CSA-WISPER	32
3.5	Simulation network topology	33
3.6	Number of lost packets of flows (transmission rate= 500Kbps)	33
3.7	Number of lost packets of flows (transmission rate= 520Kbps)	34
3.8	Max packet delay of flows 1 and 2	34
3.9	Average packet delay of flows 1 and 2	35
3.10	Simulation network topology 2	35
3.11	Number of lost packets of flows(transmission rate= 100Kbps)	36
3.12	Number of lost packets of flows(transmission rate= 102Kbps)	36
3.13	Average packet delay of flows	37
3.14	Maximum packet delay of flows	37
4.1	CSDFSC scheduler framework	49
4.2	Frame allocation procedure in CSDFSC	50
4.3	Network topology for simulation experiments	53
4.4	Number of users CSDFSC and WISPER can support	55
4.5	Comparison of packet loss ratio of TCP traffic	55
4.6	Comparison of packet loss ratio of Video traffic	56
4.7	Link utilization under maximum traffic load	56
4.8	TCP transmission rate in CSDFSC	57
4.9	TCP transmission rate in WISPER	58
4.10	Cumulative distribution function (CDF) of packet delay	58
4.11	Channel utilization vs video users	59
4.12	Channel utilization vs TCP users	60
5.1	Transmission rate vs SNR	82
5.2	Variance of D(k)	84
5.3	Delay bound and the maximum end-to-end delay of flows	85
5.4	Throughput of four flows in ROS and WRR	86
5.5	Packet loss rate in ROS and WRR	87
6.1	PDF of queue length in ROS($\rho = 0.1, 0.12$ and 0.14 , respectively)	101

6.2 Packet loss rate in video scenario 104

6.3 Average end-to-end delay in video scenario 105

6.4 System throughput in video scenario 105

6.5 Packet loss rate in mixed scenario 106

6.6 Average end-to-end delay in mixed scenario 107

6.7 System throughput in mixed scenario 107

7.1 Multi-channel WMN with two NICs 116

7.2 Example of link constraint 117

7.3 Flow notations 122

7.4 Simulation topology 128

7.5 Aggregated transmission rates of all flows(hop-delay=4ms) 129

7.6 Queue length at the bottleneck(hop-delay=4ms) 130

7.7 Average packet end-to-end delay(hop-delay=4ms) 131

7.8 Aggregated transmission rates of all flows(hop-delay=8ms) 131

7.9 Queue length at the bottleneck(hop-delay=8ms) 132

7.10 Average packet end-to-end delay(hop-delay=8ms) 132

List of Tables

4.1	Transmission rate vs SIR value	47
4.2	Parameters of flows	50
4.3	Maximum transmission rates of flows	51
4.4	Priority of flows	51
4.5	Throughput of flows	51
4.6	$Z_i(t)$ of flows	51
4.7	Parameters of channel model	53
4.8	CSDFSC parameters of traffic	54
4.9	WISPER parameters of traffic	54
5.1	Rate ratio of users	83
5.2	Video flow parameters	83
6.1	Simulation parameters	103
7.1	Simulation parameters	128

List of Abbreviations

3G	Third Generation
3GPP	Third Generation wireless Partnership Project
4G	Fourth Generation
AC	Admission Control
AD	Aggregation Device
AMPS	Advanced Mobile Phone Service
AP	Access Point
BER	Bit Error Rate
CC	Congestion Control
CDF	Cumulative Distribution Function
CDM	Code-Division Multiplexing
CDMA	Code-Division Multiple Access
CDMA/HDR	High Data Rate CDMA
CIF-Q	Channel-condition Independent Fair Queuing
CSA-WISPER	Channel-State Aware WISPER
CSDFSC	Channel States Dependent Fair Service for CDMA
CSDPS	Channel State Dependent Packet Scheduling
CTR	Capsule Transmission Request
DiffServ	Differentiated Service Model
DSR	Dynamic Resource Scheduling
FCFS	First Come First Serve
FFQ	fluid fair queuing

FQMM	Flexible QoS Model for MANET
GPS	General Processing Sharing
GR	Guaranteed-Rate
GSM	Global System for Mobile communications
HOL	Head of Line
IETF	Internet Engineering Task Force
IntServ	Integrated Service Model
IP	Internet Protocol
LAN	Local Area Network
MAC	Media Access Control
MANET	Mobile Ad hoc Networks
MLWDF	Modified Largest Weighted Delay First
NIC	Network Interface Card
PAN	Personal Area Network
PER	Packet Error Rate
PDC	Personal Digital Cellular
PDF	Probability Distribution Function
PLR	Packet Loss Rate
QoS	Quality of Service
RED	Random Dearly Detection
ROS	Rate-guaranteed Opportunistic Scheduling
ROS-CAC	Rate-guaranteed Opportunistic Scheduling Coordinated Admission Control
SAC	traffic aided Smooth Admission Control
SCDMA	Scheduled CDMA
SIR	Signal-to-Interference Ratio
SNR	Signal-to-Noise Ratio
STFQ	Start Time Fair Queuing
TCP	Transport Control Protocol
TDM	Time-Division Multiplexing

TDMA	Time-Division Multiple Access
TFRC	TCP-Friendly Rate Control protocol
UMTS	Universal Mobile Telecommunications System
VBR	Variable Bit Rate
WCDMA	Wideband CDMA
WFQ	Weighted Fair Queueing
WISPER	Multimedia Access Control Protocol with BER Scheduling
WLAN	Wireless LAN
WMN	Wireless mesh network
WRR	Weighted Round Robin
WSN	Wireless Sensor Network
xDSL	Digital Subscriber Line

List of Mathematical Symbols

ϕ_i	throughput proportion of user i
c_{ik}	channel state of user i at time slot k
p_{ik}	normalized power allocation of user i at time slot k
r_{ik}	transmission rate of user i at time slot k
$Y_i(t)$	aggregated throughput of user i at time t
γ_i	SIR of user i
P_i	transmission power of user i
W_i	predefined flow weight of flow i
$R_i(t)$	maximum transmission rate of user i at time t
$y_i(k)$	“relative throughput” of user i at time slot k
$D(k)$	maximum difference of “relative throughput” at time slot k
a_n	arrive time of packet n
d_n	departure time of packet n
l_n	packet length of packet n
$\beta(t)$	service curve
d_{bound}	delay bound of ROS
L_{max}	maximum packet length
P_l	packet loss rate of ROS
L_{on}	average length of “on” period
L_{off}	average length of “off” period
b	transit probability from “on” state to “off” state
g	transit probability from “off” state to “on” state

p_i^d	PLR requirement of user i
p_i^o	probability of buffer overflow of user i
A_i	probability of i arrivals in a time slot
u_i	probability of i packets waiting in the queue
r_f	transmission rate of flow f
F_l	set of flows traversing link l
η	efficiency of MAC protocol
$N(V)$	number of nodes in the network
$N(L)$	number of links in the network
$N(F)$	number of flows in the network
$U()$	utilization function
W_f	weight of flow f
$p_j()$	price function

Chapter 1

Introduction

In wireless networks, new distributed applications are growing rapidly. Many of these applications, such as multimedia applications, are bandwidth-intensive, and may also require a highly reliable network infrastructure. The protocols' lack of QoS control has slowed the deployment of these applications in wireless networks. While the technology for multimedia application is now available and the service is much in demand, its deployment in the wireless networks has been severely limited because of the bandwidth and congestion problem. Obviously, we need to either greatly increase existing network capacity, or develop more sophisticated network service schemes which utilize the existing capacity more efficiently and are able to provide QoS assurances under high load, or combine both of these approaches. This is the motivation behind developing more sophisticated traffic control mechanisms to support the emerging applications in wireless networks. In this chapter, we first give a brief overview of the applications and service requirements of wireless networks, and then present the motivations, objectives and our contributions in this Ph.D research.

1.1 Applications and Service Requirements of Wireless Networks

Since the occurrence of the first generation (1G) wireless networks, wireless networks have been steadily improved with better quality, lower cost, higher transmission rate and capacity, broader coverage, and more impressive services [42].

The first wireless cellular system in the world was not implemented until 1979. In Japan,

Nippon Telephone and Telegraphy company (NTT) deployed a commercial advanced mobile phone service (AMPS) system. The first North American cellular system, which had only one cell, was started at Mexico city, USA, in august 1981. Meanwhile, Nodic Mobile Telephone (NMT) system started in Europe in 1981. The rapid development of mobile telecommunications is one of the most noticeable successes of the 1990s. In particular, the second generation (2G) wireless systems, including global system for mobile communications (GSM), personal digital cellular (PDC) and digital advanced mobile phone service (D-AMPS), have brought mobile telephony capabilities to an impressive market. Compared with 1G wireless networks, the 2G wireless networks solve lots of important problems, e.g. voice quality, handoff and network capacity, etc., to a great extent by applying digital techniques. Thus, the 2G wireless networks, which are based on digital techniques, have dominated the market. Some of them are still being widely used now. 2G wireless network can only support voice and low-rate data applications (such as email), which need loose QoS guarantees.

The initial vision of the 3G wireless networks is to have a single worldwide standard called International Mobile Communications-2000 (IMT-2000) [26, 29, 72] for all wireless networks such that inter-operability and global roaming are supported. However, in reality, multiple 3G wireless networks have evolved from 2G and 2.5G wireless networks. Typical examples include wideband CDMA networks [11], CDMA2000 3x [51], and some wireless local area networks/personal area networks (LANs/PANs) such as IEEE 802.11 [28, 33, 83], 802.15 [2], 802.16 [3], HiperLAN/2 [33], and Bluetooth [25, 76]. Since 1999 there have been much efforts in harmonizing the standards. The new standard body, the 3rd Generation Wireless Partnership Project (3GPP [9]), is developing the 3G standards for GSM-based and wideband code division multiple access (WCDMA) air interface. Also another partnership, the 3rd Generation Partnership Project 2 (3GPP2 [10]), is developing 3G standards for CDMA2000-based systems, which is aimed to enhance the IP architecture for multimedia services. The Universal Mobile Telecommunications System (UMTS)/WCDMA version of the 3G systems plays a key role in the developing of 3G systems since the GPRS has already been well defined and deployed.

Compared with 2G systems, the 3G systems offer global roaming, higher capacity, and increased flexibility for multimedia services, such as audio, video, teleconferencing, and so on. The 3G systems will provide higher data rates of 144 kbps for vehicular, 384 kbps for pedestrian, and 2 Mbps for

indoor environment. For example, the world's first 3G wireless network service called Freedom of Multimedia Access (FOMA), which has been deployed in selected areas of Japan since 2001, has made videophone service a reality with face-to-face communications using real-time sound and video. Besides videophone, various applications (e.g. email, web-browsing etc.) are supported in FOMA. These multimedia and high-rate data services need much more strict QoS guarantees, such as low delay, high throughput, than that of 2G networks. It is expected that the multimedia services combining the convenience of Internet and the 3G wireless systems will bring an even more intensive opportunities and market.

The fourth generation (4G) [1] systems, which are Internet Protocol (IP) based, are proposed to be an extension of the 3G wireless systems by at least one order of magnitude regarding the capacity. The 4G systems will support interactive multimedia services like teleconferencing, multimedia mail, Java-enabled applications and wireless Internet, and they will provide wider bandwidth, even much higher data rates (up to 100 Mbps) than the 3G, global (worldwide) mobility, service portability, and the same quality as the current wireline networks. Compared with the 3G, the services in 4G systems require more stringent QoS guarantees. Moreover, the terminal battery capacity/weight/volume ratios will be increased by one order of magnitude compared with mobile terminals used for 3G systems. In addition, tight security features will protect the services from all sorts of security threats such as identity masquerade, unauthorized access, denial of service attacks, and so forth. In Korea, three 4G systems- High-speed Mobile Multimedia (HMM), High-speed Mobile Internet (HMI) and High-speed Portable Internet (HPI) have been under active development since 2002 [45]. The other Asian countries, China, Japan and Singapore, are investing huge amounts of funds to the research and development of 4G wireless core technology [64].

Along with the rapid development of wireless cellular networks, Ad-Hoc wireless networks attract much attention and become a new research hot spot in recent years. Ad-Hoc wireless networks are defined as the category of wireless networks that utilize multi-hop radio relaying and are capable of operating without the support of any fixed infrastructure. As an infrastructure-less network, Ad-Hoc network was formerly mainly used in military applications. But now more non-military applications such as group sessions, crowd management and disaster recovery, etc., are emerging. Internet Engineering Task Force (IETF) formed a working group, termed the mobile ad hoc net-

works (MANET) working group [5], to standardize the protocols and functional specifications of Ad-Hoc wireless networks. Wireless mesh networks (WMN) and wireless sensor networks (WSN) are specific examples of Ad-Hoc wireless networks.

1.2 Motivations of the Research

It is not a easy task to provide required services for supporting various applications, as wireless networks have some unique characteristics, which do not exist in wireline networks. The key characteristics of the wireless channel include: (a) bursty and location-dependent errors; (b) varying channel capacities due to the neighborhood interference and user mobility; (c) contention in the channel among multiple mobile hosts; (d) lack of global channel state (in terms of which other hosts contending for the same channel have packets to transmit, etc.) at mobile hosts; (e) limited link capacity compared with wireline networks. The bandwidth is the most valuable resource in wireless networks; (f) constraints in terms of processing capacity and battery power. These characteristics make it a challenging problem to provide QoS for applications in wireless networks.

Furthermore, the user mobility makes the wireless channel status more dynamic with handover from a cell to another. The traffic control mechanisms are influenced by the mobility and handover protocol. For example, admission control scheme should consider the movement of users and reserve some resources for handover users.

Unlike traditional one-hop cellular networks, multi-hop Ad-Hoc networks are stand-alone wireless networks which lack the services of a backbone infrastructure. They consist only of a group of mobile stations, which can act as repeaters or routers for other mobile stations in the network. Such networks were initially designed for use in military and emergency-relief applications. Lately, the Ad-Hoc network model has also been proposed and used in other applications such as sensor networks, wireless mesh networks, personal area networks, and regular wireless network applications by virtue of their better spatial reuse characteristics in comparison to the conventional cellular wireless network model. Besides those in cellular wireless networks, Ad-Hoc networks have some particular characteristics: (a) Multi-hop wireless links. Unlike the single-hop cellular networks, Ad-Hoc network nodes connect remote nodes (which they cannot connect directly) via intermediate nodes (which can connect both of them directly) instead of via base-station. (b) Self-forming,

self-healing and self-organization. Without a backbone infrastructure, Ad-Hoc networks have a flexible architecture, which will adapt the topology changes fast and effectively. These features also suggest that traffic control mechanisms designed for conventional wireless networks are not directly applicable in this environment. Dedicated traffic control mechanisms are necessary for supporting various applications in Ad-Hoc networks.

In the past decade, there has been a tremendous growing demand for the integrated services supporting broadband multimedia such as audio and video as well as the conventional data traffic. A large number of researchers have been looking into communication services with guaranteed Quality of Service (QoS) such as delivery delay and link bandwidth in both wired and wireless networks. In wireless networks mobile users would expect the similar services with those in wired network through wireless devices while they are moving around, or without the limitation of wiring. Part of this expectation will come true in the near future thanks to remarkable research and development efforts currently underway. Based on the work for the 3G wireless networks, there is still no well-defined architecture for the next generation wireless networks. Nevertheless, it is widely accepted that the next generation wireless networks will have the following features [88]:

- The wireless networks will be fully integrated with Internet Protocol (IP). Not only the protocol stack at mobile terminals and base stations is based on IP, the backbone networks for the next generation wireless systems will be IP networks as well.
- There is no single wireless network that can provide global coverage of wireless communication. Instead, various wireless networks, based on technologies that are already deployed or still under development, are vertically and horizontally organized in a hierarchy to constitute the next generation wireless networks.
- Multimedia traffic with diverse quality of service (QoS) requirements will be transmitted in the next generation wireless networks.
- The transmission of multimedia traffic in the next generation wireless networks makes new traffic control mechanisms an essential feature. These traffic control mechanisms should be robust, adaptive, efficient, etc.

Considering such features of the next generation wireless networks, many challenging research

problems of traffic control need to be carefully addressed, especially packet scheduling, admission control and congestion control, in order to effectively and efficiently support a wide range of applications.

Due to the wireless channel characteristics, existing packet scheduling algorithms which have been proven efficient in wireline network are not directly applicable to wireless networks. As wireless links are subject to bursty and location-dependent errors, packet scheduling in wireless networks is link state dependent. Moreover, the wireless channel states and capacity vary randomly in time on both slow and fast time scales [15]. Other factors of the wireless transmission influencing packet scheduling include: (a) contention in the channel among multiple mobile hosts; (b) limited bandwidth in wireless networks; and (c) limited processing power and battery power of mobile hosts. Besides considering these wireless channel characteristics, the design of wireless scheduling algorithms should address some special issues as below.

- Fairness among flows

In wireline networks, scheduling fairness is usually achieved by guaranteeing a certain service rate to a flow, and then the scheduling algorithm tries to make different flows independent with each other. Because wireline channel can be considered error-free, a flow's allocated service rate is actually the percentage of service share received by this flow. However, the fairness issue in wireless networks is much more complicated. For better channel utilization, it is a reasonable choice to defer transmission of the user experiencing error channel state till the channel recovers from the error state. Then the flow with error channel state temporarily loses its share of the transmission bandwidth and another flow with good channel state "steal" more share of bandwidth. To guarantee fairness, the flow loss share of bandwidth should be compensated later when the link recovers. However, determining how to compensate for it is not an easy task. There are some fairness definitions for wireline scheduling algorithms available in [37, 46]. In a wireless communication network, the definition and objectives of fairness guarantees become more ambiguous. The appropriate interpretation of fairness for wireless scheduling should depend on the service model, traffic type, and channel characteristics.

- QoS support

In future broad-band wireless networks, heterogeneous classes of traffic with different QoS requirements should be supported. Thus, QoS differentiation and guarantees become crucial. To achieve this goal, future wireless scheduling algorithm must include the corresponding mechanisms for QoS support. The variability of wireless channel state will make it hard to guarantee QoS in wireless network. Nevertheless, QoS should be guaranteed for the flows, either deterministically or statistically, on those links where the physical channel variation does not exceed certain thresholds.

Admission control is another essential building block of the traffic control framework for QoS provisioning besides packet scheduling. The main function of admission control scheme is to decide the admission of a new incoming flow by judging whether the QoS of existing flows is violated and QoS of the new incoming flow can be satisfied or not. Recently, various admission control schemes have been proposed for wireless networks [30] [43] [66]. By using different admission criteria, these schemes deduce different admission regions. Typical admission criteria for wireless networks include the signal interference ratio (SIR) [30], throughput [43] and packet loss rate [66] etc.

In order to achieve high end-to-end throughput and maintain system stability in multi-hop wireless networks, congestion control mechanism is essential and should be designed carefully. The main function of congestion control is to regulate source transmission rates so that the total transmission rates on any link of the network do not exceed the available link capacity, or the sum of data rates do not exceed the forwarding capacity for every node. Congestion control is important to maintain network stability and avoid network fatal collapse. While this topic has been intensively studied in the Internet context in the past few years, but little relevant work is known in the Ad-Hoc network context [92], especially in WMN context.

1.3 Research Objectives

In this thesis, we focus on the traffic control to provide QoS for multi-service wireless networks. More specifically, we are to address some key issues, including packet scheduling, admission control and congestion control, for future wireless networks. Our research objective is to provide new efficient solutions of traffic control mechanisms to support a wide range of applications with diverse

QoS requirements in multi-service wireless networks.

1.3.1 New Packet Scheduling Schemes for Wireless Networks

By exploiting wireless channel states, which can be derived (from wireless channel model) or measured (from actual wireless networks), packet scheduling schemes usually should achieve optimization of one or more QoS metrics in wireless networks. Thus the packet scheduling problems can be formulated as different optimization problems. The optimization objectives could be throughput, link utilization, etc. These optimization problems usually are constrained by other QoS metrics, such as delay requirement and link capacity. In this thesis, we are going to devise new efficient packet scheduling schemes as the solution to these optimization problems. Another issue should be addressed by packet scheduling is the computational complexity. We aim at developing scheduling schemes with low computational complexity, since mobile devices have limited power supply (battery) and low computational capability.

1.3.2 Joint Admission Control/Packet Scheduling Framework for Wireless Networks

In order to provide QoS guarantee for wireless networks, admission control and packet scheduling schemes should work coordinately. Admission control mechanism is employed to assure that total bandwidth occupied by transmitting flows will not exceed the system capacity. In the framework, packet scheduling and admission control schemes can exchange system operation state information between them. Furthermore, there exist more interactions between scheduling and admission control. For example, the effective bandwidth at the same link may change with different scheduling schemes. While the admission region of admission control varies with the change of effective bandwidth. This is so-called “joint” operation. In this thesis, we aim at designing a new admission control mechanism which can work in conjunction with the packet scheduling scheme.

1.3.3 Optimal Congestion Control for Multi-hop Wireless Networks

In this thesis, we are going to develop new congestion control mechanisms for multi-hop wireless networks. Current Internet congestion control schemes (e.g. TCP and its variants) can be in-

terpreted as seeking the global optimization of a certain aggregated utilization function subject to network capacity constraints [62]. If routing information and channel capacity constraints are given, end-to-end control schemes (e.g. TCP) can be developed to solve this problem. On the other hand, for multi-hop wireless networks, this problem can be solved by approximate distributed algorithms, which can converge to a stable operation point [27].

1.4 Major Contributions of the Thesis

The contributions of this thesis fall into the following three subjects of traffic control in wireless networks:

1. Packet scheduling. We propose and study a channel-state-aware WISPER (CSA-WISPER) scheme for Code-Division Multiple Access (CDMA) networks. Compared with original WISPER, CSA-WISPER has an advantage that it has mechanisms to deal with channel errors, which can make use of channel states to improve channel utilization. We have conducted simulations based on NS-2 to compare the performance of proposed mechanisms. The numerical results show that CSA-WISPER can achieve both higher throughput and lower delay than original WISPER. This work has been published in ICCS'2004 [59].

Next, we propose a new packet scheduling scheme, namely Channel States Dependent Fair Service for CDMA (CSDFSC), which is designed specially for high data rate CDMA (CDMA/HDR) [18] networks. We start our investigation from examining appropriate multiple-access method for CDMA/HDR networks. We compare Time-Division Multiplexing (TDM) with Code-Division Multiplexing (CDM) in the CDMA/HDR system. According to our mathematical analysis, TDM instead of CDM should be used in most situations. In other words, one user transmitting in a timeslot at full rate is more efficient than multiple users transmitting with low rates in a timeslot. CSDFSC uses multi-state Markov model, which is more accurate in describing CDMA channel than the two-state model. CSDFSC can provide fair services for best-effort traffic flows. We compare CSDFSC with the known WISPER via ns-2 simulations and the numerical results show that CSDFSC performs much better in link resource utilization, packet loss ratio, fairness between best-effort traffic, etc. Another notable feature of

CSDFSC is its low implementation complexity. CSDFSC has been published in Computer Networks [60].

As CSDFSC mainly considers the resource utilization and fairness, and cannot provide performance bounds for traffic flows, we develop an efficient Rate-guaranteed Opportunistic Scheduling (ROS) scheme which is able to provide statistic delay bound for traffic flows in wireless networks. By exploiting time-varying channel states, ROS solves the optimization problem of system throughput performance value while satisfying the “fairness” constraint of resource allocation. We prove the optimality of the proposed scheduler through mathematical analysis. Furthermore we prove that ROS has lower computational complexity compared with existing opportunistic scheduling algorithms. In order to analyze the performance bounds provided by ROS, we define a new concept of Guaranteed-Rate (GR) node with loss, which is extended from the service curve with loss model. We validate the mathematical analysis and demonstrate the throughput performance improvement and statistical delay bound by ns-2 simulations. This work has been submitted to IEEE Trans. on Vehicular Technology [58].

2. Admission control. We develop a new admission control algorithm (ROS-CAC) which works with ROS coordinately, to support statistic QoS guarantee in multi-service wireless networks. Based on our proposed mathematical model, we derive the probability distribution function (PDF) of queue length at a ROS scheduler and deduce the packet loss rate (PLR) for all flows. The new admission control policy will admit a new incoming flow if the PLR requirements of all flows (including the new flow and existing flows) are satisfied. We have conducted comprehensive simulation experiments to demonstrate the effectiveness of the proposed admission control scheme. This work has been published in Journal of Communications and Networks [61].
3. Congestion control. We develop a new hop-by-hop congestion control mechanism for multi-channel WMNs. So far, there is no similar work in the literature to address the problem in this environment. We formulate the design of congestion control mechanism in such a system as a utilization maximization problem subject to time constraint. A distributed adaptive algorithm with hop-by-hop congestion information feedback is designed to solve

the maximization problem with constraint. We prove the convergence of this new hop-by-hop algorithm to the stable system operation point. This work has been accepted by ICCS 2006 [56] and been submitted to IEEE Trans. on Wireless Communications [57].

1.5 Thesis Organization

This thesis is organized as follows. Chapter 1 introduces the motivations and objectives of this thesis. We describe some related work in Chapter 2. In Chapter 3, we present a so-called Channel-State-Aware WISPER (CSA-WISPER), which improves the system throughput by combining existing WISPER with channel error-deal mechanisms. Chapter 4 develops a new Channel-State Dependent Fair Service for CDMA (CSDFSC), which computes the packet priority based on the channel state and “relative throughput”. We propose Rate-guaranteed Opportunistic Scheduling (ROS) in Chapter 5. Besides providing statistic delay bound for flows, ROS can achieve throughput optimization compared with other Guarantee Rate (GR) nodes. In Chapter 6, we present a new admission control scheme which works coordinately with ROS. This so-called “ROS-CAC” scheme uses packet loss rate as the admission criteria. We develop a new hop-by-hop congestion control scheme for multi-hop wireless networks in Chapter 7. This new scheme comes from the solution to an optimization problem with time constraint in Wireless Mesh Network (WMN) with multi-channel MAC. Finally, we summarize this thesis and point out some directions for future work in Chapter 8.

Chapter 2

Background

2.1 Quality of Service (QoS) in Wireless Networks

One of the main objectives of traffic control is to provide QoS. QoS models specify the architecture in which some kinds of services could be provided. It is the system goal to be achieved. All QoS components such as QoS signaling, QoS routing and QoS MAC must cooperate together to achieve this goal. In this section, we will review some known QoS models including the classic integrated service model (IntServ) and differentiated service model (DiffServ) and a recent one called flexible QoS model for MANET (FQMM).

Integrated Service Model) [65] in Wireless Networks

This per-flow based approach reserves the bandwidth and guarantees controlled delay required by application in advance. Before the application starts transmitting data/voice, the resources along the path are reserved using a control protocol and not to be released until the communication is completed. No other application is allowed to use these resources during this period of time, even though these resources are not always fully utilized. Requirement of an additional signaling protocol and low utilization of bandwidth are the two major reasons why this model is not popular in wireless networks like multi-hop wireless networks. Additionally this model is not scalable since all the devices in the network have to run the same signaling protocol, which is a big challenge in

multi-vendor environments, especially over the Internet.

In [65], Mahadevan and Sivalingam extend the IntServ and DiffServ to wireless networks, by considering some special characteristics (e.g. high losses, mobility) in wireless environments. In this extended IntServ model for wireless networks, some new QoS parameters, such as “loss profile”, “power level” and “probability of seamless communications” are added to deal with the high loss, power constraint and mobility problems in wireless networks. In order to reduce the bandwidth occupancy and power consumption, some modifications are made in the resource reservation mechanism in IntServ model. For example, RSVP refresh messages are not sent so often as that in original IntServ. However, this extended model is still immature and has not been widely deployed.

Differentiated Service Model (DiffServ) in Wireless Networks [65]

This class-based approach [36] is scalable as it does not require any signaling protocol and prior reservation of resources. This model divides the traffic flows into multiple classes, and prioritizes these classes by application requirements. By using certain bits in the protocol header, this model can distinguish different traffic classes and service them appropriately. Furthermore, this model provides almost the same kind of QoS implementation as Integrated Service Model.

Although classic QoS models are not considered suitable for modern networks like MANETs and WMNs due to mobility of nodes, Nokia’s wireless mesh network for residential broadband [19] considers a static topology of wireless antennae on the roofs of various homes which interconnect to form a static mesh. This WMN implements the DiffServ architecture by providing packet classification, regulation at ingress points, per-flow assurance levels consistent with capacity and efficient MAC with assured per-hop delays.

Flexible QoS Model for MANET (FQMM) [90]

This model is proposed based on both the existing IntServ and Diffserv models. FQMM considers the characteristics (e.g. highly variable link and node mobility) of MANET and combines the high quality of IntServ and service differentiation of DiffServ. Furthermore, FQMM uses a hybrid QoS provision scheme which combines the per-flow granularity of IntServ and per-class granularity of DiffServ. Specifically, for applications with high priorities, per-flow QoS guarantees of IntServ

are provided. On the other hand, applications with lower priorities obtain DiffServ per class differentiation. FQMM uses a traffic conditioner to policy the incoming traffic at the ingress node. As the key component of this traffic conditioner, adaptive traffic profile is used to differentiate per-flow and per-class sessions and to adapt network fluctuations. As FQMM separately applies both IntServ and DiffServ for different applications, the drawbacks related to IntServ and DiffServ still remain. The upcoming 802.11e standard implements QoS enhancements that are based on both the DiffServ and IntServ approaches and can be considered as an FQMM approach.

2.2 Wireless Packet Scheduling

Unified Wireless Fair Queuing Framework [69]

Since fluid fair queuing (FFQ) has been proven to be efficient in wireline case, many wireless fair queuing algorithms try to emulate FFQ when all flows perceive error-free channels, but swap channel allocation between flows that perceive channel error and flows that perceive a clean channel in order to make short location-dependent error bursts transparent to the end user at the expense of providing coarser properties for delay, instantaneous fairness, and throughput. Since many wireless fair queuing algorithms only differ in terms of how the swapping takes place, between which flows the swapping takes place, and how the compensation model is structured. Therefore, Nandagopal [69] concludes that all these algorithms can be considered as instances of a unified wireless fair queuing architecture, which consists of the following five components:

- The error-free service, which defines an ideal fair service model assuming no channel errors.
- The lead and lag model in wireless service, which determines which flows are leading or lagging their error free service, and by how much.
- The compensation model, which compensates lagging flows that perceive an error-free channel at the expense of leading flows, and thus addresses the key issues of bursty and location-dependent channel error in wireless channel access.
- Slot queues and packet queues, which allow for the support of both delay sensitive and error sensitive flows in a single framework and also decouples connection level packet management

policies from link-level packet scheduling policies.

- Channel monitoring and prediction, which provide a reliable and accurate measurement and estimation of the channel state at any time instant for each backlogged flow.

Figure 2.1 depicts the interactions between these components in the unified architecture. Within the context of this architecture, a wireless fair queuing algorithm has the ability to “plug-in” different algorithms for each component.

General Wireless Scheduling Schemes

Recently, wireless scheduling becomes one of the research hot spots and many wireless scheduling schemes have been developed, some of them can be mapped into the unified architecture mentioned in this section. This section will introduce four wireless fair queuing algorithms, and analyze their compensation model and implementation complexity.

One of the first pieces of research work that is focused on the problem of location-dependent and bursty errors in packet scheduling for wireless networks is Channel State Dependent Packet Scheduling (CSDPS) [94]. CSDPS can use any error-free scheduling discipline, such as standard weighted round robin (WRR) algorithm. When a flow is allocated a timeslot according to the error-free service, and when the flow perceives a channel error, its transmission will be deferred and the timeslot will be allocated to the next flow according to the error-free service. WRR is used in CSDPS among flows that perceive clean channels. Since CSDPS does not measure lag or lead values for flows, there is no compensation mechanism in CSDPS. As a consequence, fairness between flows is not guaranteed.

Idealized Wireless Fair Queuing (IWFQ) model is defined with reference to WFQ [63] as error-free service. Each arriving packet is tagged as in WFQ, and the service tag for a flow is set to the finish tag of its head-of-line packet. Among all the backlogged flows with a clean channel, the flow with the least service tag is selected. Then the head of line (HOL) packet of the selected flow will be transmitted.

The service received by a flow in the IWFQ system (with link errors) is compared with the error-free system. A flow is said to be *leading*, *lagging*, or *in-sync* at any time instant if its queue size is smaller than, larger than, or the same as the queue size in the error-free system. The lag

Figure 2.1: Unified wireless fair queuing framework

of a lagging flow is the difference between the service tag of the flow in the error-free simulation and the service tag of the flow in the real system, upper bounded by $B * r_i$, where B is a scheduler parameter and r_i is the normalized weight of the flow. Since precedence of tags is maintained, a lagging flow has a low service tag and captures the channel whenever it perceives a clean channel. Among lagging flows with clean channels, the flow with the lowest tag gets to transmit until it either perceives a dirty channel or its finish tag is greater than that of some other flow with a clean

channel. This compensation model guarantees that lagging flows will catch up their lag, but may starve out leading flows in the short term.

IWFQ is the first algorithm which has a structured adaptation of fair queuing to the wireless domain. However, it has several drawbacks. First, since absolute priority is given to packet with the smallest finish time, when a flow is compensated for its previous lagged service all other error-free flows will not be served at all. For the same reason, a lagging flow will receive compensation at a rate independent of its allocated service rate, violating the semantics that a larger guaranteed rate that implies the better QoS. Second, the choice of the parameter reflects a conflict between the delay and fairness properties. A large B means that the lagging flows can receive more compensation after recovering from errors, but it also causes other error-free flows to be denied service for longer periods. In addition, the delay and throughput guarantees are closely coupled, which may not be desirable.

The major difference between Wireless Packet Scheduling (WPS) and IWFQ is that WPS uses Weighted Round Robin (WRR) with spreading of slots, not fair queuing, as its error-free service [63]. Here is an example to illustrate the WPS operation. When a backlogged flow f_1 is allocated a time-slot but perceives a channel error, it tries to swap the slot with a future slot allocation within the same frame. If this is not possible (i.e. there is no backlogged flow perceiving a clean channel with a slot allocation later in the frame), and if there is at least one backlogged flow f_2 that can transmit at the current time (f_2 does not have any slots during the remainder of the frame), f_1 's credit is incremented and f_2 's credit is decremented (both within bounds). At the start of a frame, WPS sets the effective weight of a flow equal to the sum of its default weight and its credit, and resets the lag to 0. The frame is then generated based on the effective weights of flows. WPS uses one-step prediction model to determine channel state.

The compensation mechanism has two steps in WPS. At first, intra-frame swapping attempts to compensate flows that encounter channel error by locally trading slot allocations. If this fails, the credit accounting mechanism described above keeps the difference between the real service and the error-free service across frames. By changing the effective weight in each frame depending on the result of the previous frame, WPS tries to compensate lagging flows at the expense of leading flows. In the ideal case, in-sync flows are unaffected at the granularity of frames, though their slot

allocations may change within the frame.

WPS has performance characteristics similar to IWFQ. It also has coarse short-term fairness and throughput bounds, but provides bounded delay channel access and long-term fairness. It may disturb in-sync flows when intra-frame swapping fails to find a compensation flow. It is susceptible to a lagging flow accumulating a large lag. However, it prevents complete channel capture because each flow receives the effective weight worth of slots in each frame.

As in IWFQ, Channel-condition Independent Fair Queuing (CIF-Q) simulates an error free service. Start Time Fair Queuing (STFQ) [39] is used as the error-free service. STFQ is an approximation of WFQ that eliminates the dV/dt computation complexity by setting $V(t)$ to the start tag of the transmitting packet. The lag of a flow is the difference in service between the error-free service and the real service (i.e. lead is negative lag). A flow is considered to be “active” if it is either leading or backlogged. The error-free service is applied among all active flows. If a backlogged leading flow is allocated a slot, it relinquishes the slot with a probability of α , a system parameter. If a non-backlogged leading flow is allocated a slot, it relinquishes the slot. A relinquished slot is called additional service. Instead of allocating all the additional service to the flow that has the largest lag, the compensation is distributed among the lagging flows in proportion to their allocated service rates. If no lagging flow can receive additional service because of link errors, the additional service is distributed to no lagging flows in proportion to their allocated service rate. As a result of this compensation policy, in-sync flows are not disturbed if lagging flows can receive the additional service, and leading flows degrade their service gracefully. However, in pathological cases, a lagging flow may capture the channel, as in IWFQ, and starve out other flows.

The most significant advantage of CIF-Q is its nice performance on short-term and long-term fairness, bounded delay channel access and flow throughput. However, “in-sync” flows may be disturbed during redistribution of channel allocations that cannot be used by lagging flows or the selected flow. In the general case, CIF-Q achieves the properties of wireless fair service except that it disturbs in-sync flows, and in some pathological cases, a lagging flow may capture the channel as in IWFQ.

Existing CDMA Scheduling Schemes

CDMA is the most promising multiple-access technique for the future wireless networks, and is chosen to be the standard of 3G wireless networks. In future wireless networks, there will be a mixture of different traffic classes that have different QoS requirements, such as BER, transfer rate and delay bound. For example, voice service can tolerate BER's of up to 10^{-3} , but data packets need BER's below 10^{-9} . It will be wasteful to transfer voice and data packets in same channel, since the channel must satisfy the most stringent BER specification of data packets. Thus an efficient packet scheduling scheme is essential for future CDMA networks.

Akyildiz *et.al.* [13] proposed a packet scheduling algorithm, named WISPER, for slotted CDMA network based on BER requirement. WISPER is the short term for wireless multimedia access control protocol with BER scheduling. This algorithm satisfies all flows' BER requirements while maintaining high channel bandwidth utilization. But WISPER does not consider fairness, delay bound and error-prone channel issues. WISPER incorporates a novel packet scheduler, which performs the selection and efficient accommodation of the packets to be transferred in frames. For each frame, the scheduler computes the packets' priorities, accommodates packets with highest priority to the frame, and tries to arrange packets with same or similar BER requirements into the same slot. Thus the BER requirements are satisfied and channel throughput is maximized in the same time.

WISPER [13] assumes that packets are generated in batches, and all packets in the same batch have same timeout value. The timeout value is represented by the number of frames before packet timeout. After computing each batch's priority in the next frame, the scheduler computes the number of packets that each user should transmit in the next frame. Then the scheduler accommodates packets into the next frame. The accommodation chooses packets from that with the highest priority, and follows three criteria (and in the following order) listed below. The most serious drawback of WISPER is that it does not consider wireless channel states.

- Accommodation in empty slots or in slots that have packets with same traffic class.
- Accommodation in slots that have packets with higher traffic class.
- Accommodation in slots that have packets with lower traffic class.

Andrews *et.al.* [15] described an “online” scheduling algorithm: Modified Largest Weighted Delay First (MLWDF). The weight of queue j in base-station is calculated as below:

$$\Phi_j(t) = \gamma_j W_j(t) r_j(t) \quad (2.1)$$

where $W_j(t)$ is the delay of head-of-the-line packet in queue j , $r_j(t)$ is the channel capacity for queue j , and γ_j is a positive constant that is defined arbitrarily and is different for different queue. Base-station chooses the queue with the largest weight to serve. The advanced feature of MLWDF is that it chooses queue not only based on current channel state but also current queues’ states. Furthermore it’s very easy to implement. Another advanced feature of MLWDF is its ability to adjust delay bound for flows of different class. With a larger γ_j than the other flows, flow j will obtain a smaller delay bound than the other flows. Thus the delay distributions can be shaped by choosing suitable parameters. However, MLWDF only considers delay requirement and other QoS requirements are not considered.

M.A. Arad *et.al.* proposed a hybrid CDMA/TDMA scheduler- Scheduled CDMA [16], which mainly focuses on power-constraint and uplink scheduling. In SCDMA, data is exchanged between the Base-Station and Mobile-Stations in a fixed-size unit called *capsule*. When an MS has new packets to transmit, it sends a capsule transmission request (CTR) to the BS. The scheduler in the BS places CTR in a global queue. When there is a time slot needing to be filled, scheduler chooses CTRs according to their priority level, delay requirement and time spent in queue. The choice continues while following conditions are satisfied. After the choice procedure is finished, the BS sends transmission permission capsules to MSs to inform them the transmission time and power level, which satisfies

$$\sum_{i=1}^N g_i < 1 \quad (2.2)$$

where g_i is the power index of session i . The index is given by

$$g_i = \frac{\gamma_i}{\gamma_i + G_i} \quad (2.3)$$

where G_i is the fixed spreading gain of session i , and γ_i is the signal-to-noise ratio (SNR) require-

ment, which is equivalent with BER requirement, of session i .

Based on SCDMA, O. Gurbuz *et.al.* proposed Dynamic Resource Scheduling (DSR) [41], which is a modified SCDMA without the TDMA aspect. DSR is a centralized and adaptive framework for providing resource scheduling through optimal power assignment and code hopping in wideband CDMA (WCDMA). The advanced feature of DSR is that it classifies flows into two queues (guaranteed and best effort) according to their required services. The traffic flows requesting predefined rate requirement, such as CBR, VBR and minimum-rate ABR, are classified as guaranteed flows, and will be put into the guaranteed queue; UBR and excess ABR flows are classified as best effort flows, and will be put into the best effort queue. The CTRs in the guaranteed queue receive service before those in the best effort queue, thus the predefined rates of sessions are guaranteed. However, delay bounds are not guaranteed in both SCDMA and DSR.

2.3 Admission Control in Wireless Networks

As an essential building block of the traffic control framework for QoS provisioning besides packet scheduling, the admission control scheme decides the admission of a new incoming flow by judging whether the QoS of existing flows is violated and QoS of the new incoming flow can be satisfied or not. However, the admission decision making is not easy in wireless networks, since the available resources (e.g. bandwidth) and services received by users are time-varying and location-dependent in wireless environment. Besides wireless channel characteristics, various QoS requirements (e.g. packet loss, delay) should be considered in the design of admission control for wireless networks.

Recently, a number of admission control schemes have been proposed for wireless networks [30] [43] [66]. These schemes deduce different admission regions as they use different admission criteria. Typical admission criteria for wireless networks include the signal interference ratio (SIR) [30], throughput [43] and packet loss rate [66], etc. In the following, we present some existing admission control mechanisms for wireless networks.

A joint scheduling/admission control framework is proposed in [30] to optimize resource management for uplink in WCDMA. The main motivation is that there is residual channel capacity due to the burst of voice traffic. “Relative load” is defined to evaluate the actual traffic load of base

station j as follows.

$$L_j = \sum_{i=1}^M \frac{\beta_i^t g_{ij}}{\sum_{k \in K_i} g_{ik}} \quad (2.4)$$

where β_i^t is the target carrier to total interference ratio (C/I) for the mobile user i , g_{ij} is the processing gain from base station j to user i , M is the set of all mobile users with an active link to base station j , K_i is the set of all base stations that have positive processing gain with user i . We use a simple admission policy that L_j should be less than a threshold. However, in order to support VBR traffic, WCDMA can provide multiple transmission rates for users. The transmission rate of a flow in WCDMA is proportional with its C/I. A variable bit rate profile is defined for such VBR traffic. By assigning different credit limits a_v to different rates (less credit to higher rate), the target C/I ratio requirement is recomputed. The transmission rates higher than r are not guaranteed and a_v controls the maximum time period transmitting at rate v . The spare capacity of all VBR users is accumulated in pool, and can be used by any user. A new VBR user is admitted only if there is spare capacity after the minimum requirement of the new user is satisfied.

In [53], Kwok and Lau investigate the issue of burst data admission control and scheduling in CDMA networks. In order to provide guaranteed service for bursty multimedia traffic in CDMA networks, the existing simple policies (FCFS and equal sharing) are suggested. However, these simple policies can not optimize resource utilization while providing performance guarantees. The main contribution of Kwok and Lau's work is a novel integer programming formulation of burst data admission control and scheduling. This formulation deduces an optimal algorithm to generate admission control policy, by considering other important factors such as mobility, power control and soft handoff.

A traffic aided smooth admission control (SAC) for opportunistic communication systems, which can provide throughput guarantee, is proposed in [43]. The existing trunk reservation policy of making "hard" admission decisions has two limitations: (1) it is difficult to calculate explicitly the link capacity for opportunistic multi-access systems; and (2) the admission of incoming new user may cause the overload of system and degrade the services received by existing users. To overcome these limitations, SAC spreads the decision over a trial period. In the trial period, the amount of time resources allocated to new incoming users increase gradually. Furthermore, the modified weighted proportional fair (WPF) scheduling is used to devise a QoS driven weight

adaptation algorithm, and the weights assigned to new users are also increased gradually. The measured throughput in this trial period is used to make admission decisions. SAC performs quite well except that the computation complexity is high.

2.4 Congestion Control in Multi-hop Wireless Networks

Although congestion control is essential for multi-hop wireless networks to maintain network efficiency and fairness [44], it is not deeply studied. When the wireless channel capacity is varying or the traffic load is increasing, wireless nodes should adapt their data rates based on congestion feedback. Otherwise, the network may collapse due to congestion. Congestion control in multi-hop wireless networks should also consider the unique characteristics of multi-hop wireless networks, such as multi-hop link, power constraint and mobility.

In the following, we review some known work on the congestion control for multi-hop wireless networks. A host-by-host congestion control mechanism for wireless sensor networks is presented in [44]. At first, each node detects possible congestion. The simplest way is to monitor the queue size at the node. If the queue size is greater than a threshold, the sensor is considered in congestion and then the sensor in congestion sets the congestion bit in the header of its outgoing packets. After overhearing the congestion bit from its child node, a sensor will reduce data forwarding rate to allow its child to drain the queues. If the hop-by-hop feedback eventually arrives at the source node, the application level flow control mechanism will adapt the source rate. CODA [86] uses another method to detect congestion. Before the sensor sends a packet, it samples the channel states in a fixed time interval. A utilization factor is computed based on the number of times that the channel is busy. If the utilization factor is greater than a threshold, the sensor is considered in congestion. When congestion is detected, the node sets the congestion bit in the header of every outgoing packets. Otherwise, the congestion bit will be cleared. In [92], Yi and Shakkottai propose a fair hop-by-hop congestion control algorithm with the time constraints at the MAC layer for multi-hop wireless networks. The stability of this algorithm is proved by a Lyapunov function method. It is also proved that the algorithm has the property of spatial spreading, i.e. the congestion at bottleneck node can be “smoothed” over nearby hops. Besides rate control, power control can also be integrated into a congestion control scheme. Some work focuses on the congestion control

implemented by controlling the transmit power level of mobile hosts, since the transmission rate can be adjusted easily and quickly by adjusting the transmitter power [71]. In order to achieve the highest throughput, an iterative algorithm-CBW [71] is designed to seek an optimal value for a set of transmitters' power level. One advantage of CBW is the capability to adapt to changes of channel conditions, network topology and routing in wireless networks.

As a new solution for last-mile broadband Internet access, wireless mesh networks (WMNs), which is a special kind of multi-hop wireless networks, become a hot research topic recently [14, 47, 77]. WMNs are anticipated to solve the limitations and to significantly improve the performance of Ad-Hoc networks, wireless local area networks (WLANs), wireless personal area networks (WPANs), and wireless metropolitan area networks (WMANs), which are undergoing rapid progress and inspiring numerous deployments. Furthermore, WMNs are developed to deliver wireless services for various applications from personal, local, campus, to metropolitan areas scale. More importantly, WMNs will be integrated with other networks, such as the Internet, cellular network, IEEE 802.11, IEEE 802.15, IEEE 802.16, sensor network, etc. This integration poses an urgent need for the effective congestion control in this environment to maintain network stability and achieve a high network resource utilization.

Chapter 3

A Channel-State-Aware BER Scheduling Scheme for Wireless Multimedia Access Control Protocol

As a key component of traffic control mechanisms, packet scheduling for wireless networks is the first issue we have addressed. In future wireless networks, the packet scheduler plays a key role to support heterogeneous classes of traffic with different bit-error rate (BER) and other quality of service (QoS) requirements. In this chapter, we propose a channel-state-aware BER scheduling scheme for CDMA networks (CSA-WISPER) by combining the existing WISPER scheme with a number of mechanisms that we design to deal with channel errors and guarantee fairness. Simulation experiments based on NS-2 illustrate that our scheme can achieve both lower packet loss ratio and lower packet delay compared with the original WISPER scheme.

3.1 Introduction

With the growing use of wireless data services and the increasing demand for multimedia applications, it is highly expected that future wireless networks will provide services for heterogeneous classes of traffic, and satisfy their different QoS requirements. In order to provide QoS differentiation and guarantee in wireless networks, various new technologies have been and being proposed. Wireless packet scheduling is one of the most important relevant technical issues that need to be addressed. It is essential to design a flexible, scalable scheduling scheme that can accommodate different kinds of multimedia traffic, and satisfy various QoS requirements.

In wireline networks, many scheduling algorithms, such as fair queuing scheduling [31], virtual

clock [94], EDD [48], etc., are capable of providing certain guaranteed QoS. However, these existing service disciplines are not directly applicable in wireless networks context because they have not taken into account some unique problems in wireless networks, such as varying wireless link capacity, location-dependent and burst errors, as well as joint scheduling of uplink and downlink flows in a wireless cell. CDMA is the multiple access standard of 3G wireless networks and is expected to be widely used in future communication infrastructure [13]. One of the biggest differences between CDMA and other multiple access modes is that in CDMA, a mobile terminal can use multiple channels at the same time, i.e. base station can transmit several packets to one terminal in one slot. This corresponds to “multi-server” issue, which makes scheduling problem in CDMA networks more complicated.

In this chapter, we first summarize the problems faced by fair queuing scheduling in wireless networks and review several known scheduling algorithms for CDMA networks, including the known wireless multimedia access control protocol with BER scheduling (WISPER) [13]. We realize that WISPER does not take into account the varying channel states, which cause poor performance in error-prone wireless networks. This motivates us to propose a modified scheme called Channel-State-Aware WISPER (CSA-WISPER) by combining WISPER with mechanisms that deal with error-prone channel states. CSA-WISPER can achieve higher throughput as well as lower delay bound for different classes of traffic, compared with the original WISPER.

The rest of this chapter is organized as follows. We elaborate the proposed CSA-WISPER scheme in Section 3.2 and present our simulation results in Section 3.3. Section 3.4 concludes this chapter.

3.2 Channel-State-Aware WISPER (CSA-WISPER)

The WISPER scheme tries to achieve as high as possible throughput without violating BER requirements of all flows. However, it does not use channel state prediction and consider channel state in the scheme. Thus combining mechanisms dealing with channel errors and WISPER should be a reasonable choice and is expected to achieve better performance.

3.2.1 Channel State Model

The proposed CSA-WISPER makes use of channel state information and fairness-guaranteed algorithms. In our proposal, we assume that the states of wireless channels follow the two-state Markov chain model and are independent with each other. The base station has the knowledge of the channel states of all links [60].

Two-state Markov chain model is widely used in communication systems due to its simplicity and effective to present the properties of communication modules, such as channel, queue, source etc [89]. For a channel with memory, the Gilbert-Elliott (GE) model [89] is one of the simplest models. The GE channel is a first-order, discrete-time, stationary, Markov chain with two states: good and bad, which are denoted by G and B respectively as shown in Figure 3.1. The probabilities that channel state changes from G to B and from B to G are denoted by b and g respectively. Wilhelmsson [89] shows GE model is accurate for a broad class of parameters, such as delay constraint, SNR and mobile speed, of a land mobile channel.

Figure 3.1: GE channel model

3.2.2 CSA-WISPER mechanisms

Channel error handling is important for wireless networks due to the error-prone characteristic of wireless channels. Deterring the transmission of mobile user which is experiencing channel error will be a natural choice to improve system throughput. However, this choice may violate the fairness between users, since different mobile users experience various channel states. The transmission opportunities of mobile users with bad channel states will be reduced; while the mobile users with good channel states will obtain extra transmission opportunities. Our new CSA-WISPER exploits the channel state information to implement packet scheduling mechanism

while not violating fairness bound. Thus the channel error handling algorithm is one of the main components in CSA-WISPER.

Since WISPER does not consider error-prone wireless channel state, it has no mechanism to handle channel errors. CSA-WISPER has several mechanisms to handle channel errors in scheduling the packets: “drop”, “give-up”, “swap” and “compensation”. The simplest policy is “drop”, which corresponds to the original WISPER. In this policy, base station transmits packets regardless the channel state. When the channel is in a bad state, the packets will be dropped and cannot reach the mobile terminal.

A possible improvement of the “drop” mechanism is to utilize the information on the channel state at a BS. Since a base station knows the states of all wireless links connected to it, when it finds out that a specific mobile terminal in slot t of the next frame will experience a bad channel state, that terminal will “give up” this slot, and base station will allocate packets, which have the highest priority in the queue and perceive a clean channel, to occupy this slot.

In the CSA-WISPER scheme, every flow has an attribute “lead”. When a flow has to give up a slot because of channel errors, it will decrease its “lead” value by the number of packets that it will give up. The flow that obtains this slot will increase its “lead” value by the number of packets it will send in this slot. A flow is said to be “leading”, “lagging” and “in-sync”, when its “lead” value is positive, negative and zero, respectively. To achieve better delay performance, a more complicated policy, “swap”, has been proposed. In this scheme, a flow (flow n) with a bad channel state tries to find another flow m , which has packets in another slot s in the same frame but experiences good channel state in slot t . Then the two flows “swap” their packets in slots s and t . If flow n cannot find another flow to swap, it should give up this slot as in the “give up” policy.

In order to guarantee “fairness” among all flows, the “compensation” scheme is proposed to allocate more service to lagging flows in future frames. A flow is considered “lagging” when this flow gave up service opportunity before and its “lead” attribute is negative. When a frame is scheduled and ready to send, the base-station will check if there are packets belonging to the leading flow. If yes, the base-station further checks if there are packets in the queue belonging to the lagging flow. If lagging packets are found, the leading flow will relinquish the time-slot it uses in current frame to the lagging flow which has the highest priority packets in queue and perceiving good

channel state. After the relinquishment, the leading flow will decrease its “lead” value by the number of packets it relinquished and the lagging flow will increase its “lead” value by the number of packets it transmitted in this relinquishment. Flows with error channel states are handled as that in the “swap” scheme. The algorithm flow chart for the “compensation” mechanism is shown in Figure 3.2.

Figure 3.2: Flow chart of CSA-WISPER scheme

3.2.3 Operation Example of CSA-WISPER

We use an example to illustrate the operation of CSA-WISPER as shown in Figure 3.3. CSA-WISPER assumes that packets arrive in batches. The packets in the same batch have the same timeout value. The timeout value is the number of frames before packet timeout. The priority of packets in a batch is calculated as follows:

$$\Phi_{\beta}(t) = \begin{cases} \frac{|P_{\beta}(t)/M_n|}{F_{\beta}(t)}, & \text{if } \frac{|P_{\beta}(t)/M_n|}{F_{\beta}(t)} \leq N_p \\ N_p, & \text{otherwise} \end{cases} \quad (3.1)$$

where M_n represents the maximum number of packets can be simultaneously transmitted in a slot for end-user n , $F_{\beta}(t)$ represents the timeout value at time t of batch β , $P_{\beta}(t)$ represents the number of packets left in batch β , N_p is the number of slots in a frame.

Let there be 21 packets in six batches in the queue at the BS, and these packets belong to three traffic classes: A, B and C. Class A has the most stringent BER requirement so that it can only transmit $S=4$ packets in the same slot to maintain the BER requirement. The mobile terminal transmitting packets of class A has a maximum transmission capability $M=2$, i.e. it can only transmit 2 packets in the same slot. The corresponding parameters for class B and C are $S=5$, $M=3$ and $S=7$, $M=3$ respectively. In Figure 3.3, each box with a priority record represents a batch of packets. On the upper left corner of each box is the timeout value - TO, which is the frame number before the packets expiry. If $TO=1$, the packets in this batch will expire if they are not transmitted in this frame. The number on the upper right corner indicates the number of packets in this batch. The rows below "Priority" and "Packs" are filled with the priority value corresponding to the total number of packets in the batch, and the number of packets that can be transmitted according to a given priority value. The priority value and number of packets which can be transmitted are calculated according to equations in [13]. In this example we assume that a frame has 4 slots.

The packet allocation is performed as follows. At first, A-1 and C-1 have the highest priority equal to 2 in all batches, so they are allocated in the frame as shown in the table a) in Figure 3.3. Next, B-1 and C-2 with the priority equal to 1 are chosen. Since there is no empty slot now and no slot has packet of class B, B-1 first tries to be accommodated with higher class A. Thus B-1 packets are accommodated with A-1 in slots 1 and 2. C-2 can be accommodated with C-1 as shown in table b) in Figure 3.3. Then, C-3 with priority equal to 0.66 is chosen, and it is accommodated with C-1 and C-2 as shown in table c). At last, B-2 batch with priority equal to 0.5 is chosen. One packet of B-2 is accommodated with B-1 at slot 2. While the other two packets of B-2 cannot find packets with the same or higher class to be accommodated, they try to stay with lower class

packets. Thus these two packets are accommodated with C-1 at slot 3 as shown in table d). The maximum number of packets can be accommodated in slot 3 is changed to $S=5$. Now all slots are full and the BER requirements of all packets in the frame are satisfied. As a conclusion, scheduler accommodates 20 packets in this frame, which is 20% more than the worst case. This part of operations is similar to that in [13].

Figure 3.3: Packet allocation of CSA-WISPER

Figure 3.4 shows an example of our proposed schemes to handle channel errors. The upper left table is the frame after packet allocation process shown in Figure 3.3. Assume that the BS finds that C-1 at slot 3 encounters channel errors. At first, the BS tries to find the same class packets to swap. C-2 at slot 4 is suitable to swap with C-1. But there is already a C-1 packet in slot 4, thus only 2 of C-1 packets in slot 3 can be swapped with 2 of C-2 packets in slot 4, as shown in the upper right table. The last packet of C-1 at slot 3 cannot swap with any other packet, so it has to relinquish its space to the C-3 packet in queue, as shown in low left table in Figure 3.4. From this example, we can see CSA-WISPER can reduce packet loss ratio.

Figure 3.4: Error-deal scheme in CSA-WISPER

3.3 Simulation and Numerical Results

LBNL's network simulator NS-2 is employed in our simulation experiments. We use Pareto traffic as the traffic model in our simulations. Pareto traffic is an ON/OFF packet generator with Pareto distributed on and off times. When it is in "on" time, the traffic is generated in a constant rate. While there is no traffic generated during "off" time. Compared with traditional Poisson distribution [32] traffic, Pareto traffic is more suitable for describing broadband multimedia traffic. All data sources last for 50 seconds during the simulation period. The packet length is fixed to 500 bytes.

3.3.1 Simulation Scenario 1

In our first simulation experiment, we use a double star topology with 2 flows as shown in Figure 3.5. In this topology, there are two BSs and four mobile hosts. Flow 1 is transmitting from S1 to D1; while flow 2 transmits from S2 to D2. Flow 1 traverses an error-prone link; while flow 2 experiences an error-free link. The link between BS1 and BS2 is the bottleneck link with 1Mbps bandwidth and 10ms transmission delay. All other links have the same bandwidth and delay. Both flows have the same type of source and transmission rates, and they start and stop transmission at the same time. We use packet loss, average and maximum packet delay as the performance metrics. Four schemes: WISPER ("drop"), "give up", "swap" and "compensation" are compared in terms of the above performance metrics.

Figure 3.5: Simulation network topology

At first, the transmission rate for both flows is set to 500Kbps. The number of packet losses is shown in Figure 3.6. Since the total transmission rate is smaller than the link capacity, all schemes have no loss except the WISPER scheme. However, when the transmission rate increases to 520Kbps, all schemes have packet losses as shown in Figure 3.7. We can also find that “give up” scheme has the largest number of packet losses; while “swap” and “compensation” scheme have the least number of losses. In the “give up” policy, flow 1 has more packet losses than flow 2, but in “swap” and “compensation” policies, both flows have the same number of packet losses. The “swap” and “compensation” schemes provide better fairness for flows.

Figure 3.6: Number of lost packets of flows (transmission rate= 500Kbps)

Figures 3.8 and 3.9 show the maximum packet delay and the average packet delay respectively. We can find that the WISPER scheme has the highest packet delay; while the other schemes have almost the same packet delay.

Figure 3.7: Number of lost packets of flows (transmission rate= 520Kbps)

Figure 3.8: Max packet delay of flows 1 and 2

3.3.2 Simulation Scenario 2

Next, we change the simulation scenario by increasing the number of flows to ten, as shown in Figure 3.10. Suppose there are ten flows transmitting from ten source nodes to ten destination nodes, and there are still two wireless base-stations. All links between two base-stations and between nodes and routers have bandwidth of 1Mbps, and delay of 10ms. The CSA-WISPER scheme is deployed at BS1 and BS2. Between each pair of source-destination nodes, there is a traffic flow. All ten flows have the same type of data sources, transmission rates and start to

Figure 3.9: Average packet delay of flows 1 and 2

transmit at the same time. Let the odd number flows belong to flow class 1, whose maximum number of packets per slot is 3 and maximum transmission rate per terminal per slot is 1; and even number flows belong to flow class 2, whose maximum number of packets per slot is 4 and maximum transmission rate per terminal per slot is 2. Without loss of generality, we assume that flow 2 from node S2 to node D2 traverses an error-prone link.

Figure 3.10: Simulation network topology 2

The performance metrics we use include packet loss, average and maximum packet delay. We compare four scheduling schemes: “drop” (original WISPER), “give up”, “swap” and “compensation” in terms of the above performance metrics.

At first, the transmission rate for all the flows is set to 100Kbps. The number of lost packets for different schemes is shown in Figure 3.11. In this figure and hereafter, the x-axis is the flow number, which is from 1 to 10. Since the total transmission rate is smaller than the link capacity, all schemes have no loss except the original WISPER scheme, since only WISPER has no error-deal

mechanism. However, when the transmission rate increases to 102Kbps, all schemes have packet losses as shown in Figure 3.12. From this figure we can find that original WISPER scheme has the largest loss number while “swap” and “compensation” scheme have the least number of lost packet. Since the scheduler choose packet from flow with lower index to flow with higher index, the packet losses are accumulated in flow 10.

Figure 3.11: Number of lost packets of flows(transmission rate= 100Kbps)

Figure 3.12: Number of lost packets of flows(transmission rate= 102Kbps)

Figures 3.13 and 3.14 show the average and maximum end-to-end packet delay of all flows, we can see flows belonging to same class have similar delay values. For almost all flows, “swap” and

“compensation” scheme have lower average and maximum packet delay, while the original WISPER (“drop”) scheme has the highest average and maximum packet delay.

Figure 3.13: Average packet delay of flows

Figure 3.14: Maximum packet delay of flows

The simulation results illustrate that “swap” and “compensation” schemes have lower packet loss ratio and smaller average delay than other two schemes. Since the total number of packets generated in simulation is fixed, lower packet loss means higher throughput and link utilization. The original WISPER algorithm has the largest average delay and lowest throughput. The simulation results also show that in most situations, “swap” performs as good as “compensation” algorithm. Since in

our simulation, there always exists same class traffic in a specific frame, thus a flow that encounters channel error always can find another flow to swap. Therefore, there is no need to compensate. But in realistic networks, this situation cannot always exist. Thus we can expect that “compensation” scheme will have better performance than “swap”. Another important improvement of “swap” and “compensation” is that fairness among flows is guaranteed. The flows of the same class share channel more fairly in the “swap” and “compensation” algorithms than in other two schemes.

3.4 Conclusion

In order to guarantee QoS for multiple class traffic in CDMA networks, an efficient scheduling scheme must consider not only wireless channel variability but also special properties of CDMA networks, such as multi-class packets in the same slot. In this chapter, we have proposed and studied a so-called CSA-WISPER scheme for CDMA networks by taking into account the error-prone channel states. We have conducted simulations based on NS-2 to compare the performance of several schemes. The numerical results reveal that combining WISPER and some general error-deal schemes can achieve both higher throughput and lower average delay.

Chapter 4

Channel States Dependent Fair Service: A New Packet Scheduling Algorithm for CDMA

In previous chapter, we have presented CSA-WISPER- a packet scheduling scheme for CDMA networks. We further our work on packet scheduling in order to propose a more effective and QoS-concern scheduling scheme. In this chapter, we address the problem of packet scheduling for high speed CDMA, which can provide high speed data service for mobile users. We first investigate the appropriate multiplexing schemes in high speed CDMA. We find through mathematical analysis that TDM is more efficient than CDM for downlink in high speed CDMA networks in general situations. Based on this finding, we propose and evaluate an efficient scheduling scheme, namely Channel States Dependent Fair Service for CDMA (CSDFSC), for packet scheduling in high speed CDMA networks. CSDFSC tries to maximize channel throughput under fairness and transmission power constraints. This scheme differentiates best-effort and guaranteed services by assigning different queue weights to various traffic classes. By considering the difference between the pre-defined and actually occupied bandwidth proportion in the priority computation, CSDFSC exhibits good fairness properties for best-effort flows. We compare the performance of CSDFSC with that of WISPER and WRR by using ns-2 simulations, where VBR video and TCP traffic flows are considered. The numerical results show that CSDFSC outperforms WISPER and WRR in terms of throughput, link utilization and average end-to-end packet delay, while retaining a low implementation complexity.

4.1 Introduction

CDMA is the multiple access standard of 3G wireless networks and will be widely used in future communication infrastructure [85]. CDMA systems have been successful in providing high-quality voice service with high system capacity. In recent years, on the other hand, there has been rapid growth of demand for wireless data services, in conjunction with the explosive growth of the Internet. As a response to this trend, high speed CDMA networks which can provide high speed data service for mobile users have been proposed. For example, high data rate CDMA (CDMA/HDR) [18] as a overlay to CDMA has been recently proposed for providing packet data service to mobile users. With the growing deployment of wireless data services and the increasing demand for multimedia applications, it is highly expected that future wireless networks will provide services for heterogeneous classes of traffic, and satisfy their different quality of service (QoS) requirements. In order to provide service differentiation and QoS guarantee in wireless networks, many new technologies have been and being proposed. Packet scheduling in wireless networks is one of the most important technical issues that need to be addressed. A good wireless packet scheduling algorithm should try to achieve following objectives: accommodate different kinds of multimedia traffic with various QoS requirements; provide fairness and throughput guarantee to error-free flows; achieve high channel utilization and minimize packet loss; achieve low implementation complexity.

In wireline networks, many existing scheduling algorithms, such as fair queuing scheduling [31], virtual clock [94], and EDD [48], are capable of providing certain guaranteed QoS. However, these existing service disciplines are not directly applicable in wireless networks because they do not consider the unique problems in wireless networks, such as varying wireless link capacity, location-dependent and burst errors and joint scheduling of uplink and downlink flows in a wireless cell. The packet scheduling in CDMA networks is a more challenging problem. One of the greatest differences between CDMA and other multiple access methods is that in CDMA, a mobile terminal may use multiple channels at the same time, i.e. base station can transmit several packets to one terminal in one slot. This corresponds to “multi-server” issue, which makes scheduling problem in CDMA networks more complicated [23].

In this chapter we address the packet scheduling problem in high speed CDMA networks. We first summarize the problems faced by fair queuing scheduling in wireless networks and review

several known scheduling algorithms for CDMA networks, including the wireless multimedia access control protocol with BER scheduling (WISPER) [13]. Based on the CDMA downlink scheduling model we formulated, we theoretically prove that TDM is more efficient than CDM multiplexing scheme for downlink in high data rate CDMA networks under certain conditions. In other words, a base station should schedule only one user to transmit in a time slot. Our finding coincidentally is the same as CDMA/HDR and HSDPA(High Speed Downlink Packet Access).

Based on this finding, we propose a new scheduling algorithm especially for high speed CDMA networks, namely Channel States Dependent Fair Service for CDMA (CSDFSC), which makes decision in each time slot and chooses the user with highest priority to transmit data in one time slot. Since bandwidth is the most scarce resource in wireless networks, the objective of our proposal is to maximize channel throughput and minimize packet loss while supporting multiple classes of traffic and providing fair service for best-effort flows. In order to achieve these objectives, CSDFSC computes the priority of packets according to their queue weights, fairness among flows and current channel states. By using different queue weights for best-effort and guaranteed traffic flows, CSDFSC differentiates the services and provides higher priority and lower packet delay for the latter. Another advantage of CSDFSC is that it tries to provide fair service for best-effort traffic by measuring the difference between actual occupied bandwidth and predefined proportion of bandwidth, and computing the priority of flows based on this difference. In the design of CSDFSC, multiple states Markov model is used to describe the CDMA link states and the maximum transmission rate in CSDFSC. The flow perceiving channel error will have lower priority and may give up their transmission opportunity to those without channel error. Furthermore, the scheduling process is much simpler in CSDFSC than that in WISPER, because of the lack of packet allocation procedure in CSDFSC. As a result, its implementation complexity is significantly lower than that of WISPER. For a more comprehensive evaluation for the performance of our proposed algorithm, we have conducted extensive simulation experiments to compare the scheduling algorithm of CSDFSC with both WISPER and Weighted Round Robin (WRR) schedulers. The numerical results demonstrate that CSDFSC can achieve not only higher channel capacity, but also higher link bandwidth utilization and lower packet loss ratio as compared with WISPER and WRR. Moreover, CSDFSC can achieve lower average packet delay for the guaranteed traffic and “fairer” service for best-effort

traffic as compared to WISPER.

The rest of this chapter is organized as follows. We present the system model and elaborate the theoretical analysis on the multiplexing scheme for high speed CDMA in Section 4.2. We present the proposed CSDFSC in Section 4.3. The simulation results and discussions are presented in Section 4.4. We conclude this chapter by highlighting our contributions in Section 4.5.

4.2 System Model and Theoretical Analysis

4.2.1 Proposed CDMA Scheduling Model

In this section, we present the proposed packet scheduling model for the downlink in CDMA networks. We consider a single cell in a CDMA system, where there are N users and each user has one flow that needs to be served. We denote the set of users by $\mathbf{N} = \{1, \dots, N\}$. Each flow has its own queue in the base station. The channel transmitting time is divided into some intervals with a fixed duration. A time interval $[t, t + \delta)$, where $t = 0, \delta, 2\delta, \dots$, is called the time slot. For simplicity, the channel state is represented in terms of the noise level, including fading and interference, experienced by the users. $\mathcal{C} = \{s_1, s_2, \dots, s_C\}$ is the set of possible channel states, where C denotes the number of the channel states, and the channel state is invariable within one time slot. The random channel state process c is assumed to be a discrete time Markov chain with the finite state space $\mathcal{C}$, an efficient model to describe CDMA link [69]. Associated with each state $c \in \mathcal{C}$ is a maximum feasible data rate $r_c \in \mathcal{R}$. This model is adopted from [78]. $\mathcal{R}$ is the finite set of feasible rates of CDMA users, $\mathcal{R} = \{r_1, r_2, \dots, r_C\}$. Let c_{ik} be the channel state of user i at slot k , which is assumed to be known by the base station or predictable by some algorithms such as one-step prediction [63]. Let p_{ik} be the normalized power allocation of user i at time slot k . In CDMA, the base station can assign different signal power level by choosing a spreading code for a mobile user, the actual transmission rate is determined by SIR (Signal Interference Ratio) experienced by the mobile user. Then the transmission rate of user i at time slot k , represented by r_{ik} , is given by

$$g(c_{ik}, p_{ik}) = r_{ik} \in \mathcal{R},$$

where $g()$ expresses the transmission rate as a function of channel state and transmission power level. Moreover, let $Y_i(t)$ be the throughput of user i from time 0 to t . $t = M\delta$, where M is an integer. Thus we have

$$Y_i(t) = \sum_{k=1}^M r_{ik} * \delta$$

The scheduling objective is to maximize the total throughput

$$\sum_{i \in \mathbf{N}} Y_i(t) = \sum_{i=1}^N \sum_{k=1}^M g(c_{ik}, p_{ik}) * \delta \quad (4.1)$$

subject to two constraints:

1. Fairness constraint

$$\frac{Y_i(t)}{\phi_i} = \frac{Y_j(t)}{\phi_j} \quad \text{for each } i, j \in \mathbf{N}, \quad (4.2)$$

where ϕ_i represents the assigned proportion of bandwidth of user i , which is usually determined by service provider [13];

2. Power constraint

$$\sum_{i \in \mathbf{N}} p_{ik} \leq 1 \quad \text{and} \quad p_{ik} \geq 0, \quad (4.3)$$

where p_{ik} is normalized by total transmission power, since total transmission power of the base station is limited [82] [21].

In the following, we analyze the multiplexing method for high speed CDMA by using the proposed system model.

4.2.2 Multiplexing Schemes in High-Data Rate CDMA: TDM versus CDM

As the CDMA packet scheduling issue is related with the multiplexing scheme, we first compare the existing multiplexing methods to find the appropriate one for downlink in high speed CDMA. TDM (Time Division Multiplexing) and CDM (Code Division Multiplexing) are two types of widely deployed multiplexing schemes for communication networks. In TDM, only one user can transmit at a time and it only transmits in a short time scale. In CDM, multiple users transmit data simultaneously in a long time scale but with a lower transmission rate. Both schemes have their advantages and disadvantages. Interference between users exists in CDM, while TDM needs precise

and complicated control of transmission time. In the following, we employ a mathematical model to examine TDM and CDM for high speed CDMA networks to choose an appropriate one.

First, we examine the transmission rate in one time slot. Our objective is to maximize the sum of all users' transmission rates in this time slot. If the transmission rate is maximized in every time slot, the total throughput can be also maximized. In a time slot, user i receives transmission power P_i , with SIR γ_i and transmission rate R_i . Hence the optimal problem becomes *Maximize* $\sum_{\forall i} R_i$. First, let us consider a special case where $\frac{R_i}{r_0} = \frac{\gamma_i}{\gamma_0}$, in which r_0 is the minimum feasible transmission rate, and γ_0 is the minimum SIR which can provide r_0 for user i . We have the following theorem.

Theorem 4.1 *If the maximum transmission rate is in proportion to SIR, $\sum R_i = \sum \gamma_i \frac{r_0}{\gamma_0}$ reaches maximum value when only one user transmits at full power.*

Proof: For simplicity, we express P_i in unit of total interference $I + \sigma$, where σ is the thermal noise and I is the total interference from other sources, including the other cells in networks [52]. Here we focus on downlink scheduling instead of uplink scheduling in [52]. To meet SIR requirement, the following condition should be satisfied:

$$\frac{P_i}{\sum_{j \neq i} P_j + 1} \geq \gamma_i \quad (4.4)$$

Obviously, for the optimal situation the solution will be $P_i / (\sum_{j \neq i} P_j + 1) = \gamma_i$, or $\frac{P_i}{\gamma_i} = \sum_{j \neq i} P_j + 1$. Thus we have

$$P_i \frac{\gamma_i + 1}{\gamma_i} = \sum_{j \in \mathbf{N}} P_j + 1 = \text{Const}, \quad (4.5)$$

where Const is a constant. Const can be resolved as $\text{Const} = 1 + \text{Const} \sum_{i \in \mathbf{N}} \frac{\gamma_i}{\gamma_i + 1}$, or $\text{Const} = 1 / (1 - \sum_{i \in \mathbf{N}} \frac{\gamma_i}{\gamma_i + 1})$. Defining $\alpha_i = \frac{\gamma_i}{\gamma_i + 1}$, from Equation (4.5) we have

$$P_i = \frac{\alpha_i}{1 - \sum_{i \in \mathbf{N}} \alpha_i} \quad (4.6)$$

Considering the power constraint $\sum P_i \leq 1$, in an optimal situation,

$$\frac{\sum_{i \in \mathbf{N}} \alpha_i}{1 - \sum_{i \in \mathbf{N}} \alpha_i} = 1 \Rightarrow \sum_{i \in \mathbf{N}} \alpha_i = \frac{1}{2} \Rightarrow \sum_{i \in \mathbf{N}} \frac{\gamma_i}{\gamma_i + 1} = \frac{1}{2} \quad (4.7)$$

Our objective is to maximize $\sum R_i = \sum \gamma_i \frac{r_i}{\gamma_i}$, while satisfying Equation (4.7). Using $\alpha_i = \frac{\gamma_i}{\gamma_i + 1}$, the problem becomes:

$$\text{Maximize } f(\alpha) = \sum_{i \in \mathbf{N}} \frac{\alpha_i}{1 - \alpha_i} = \sum_{i \in \mathbf{N}} \left(\frac{1}{1 - \alpha_i} - 1 \right),$$

where $\alpha = \{\alpha_i, i = 1, 2, \dots, N\}$, subject to constraint $\sum_{i \in \mathbf{N}} \alpha_i = \frac{1}{2}$.

According to Lagrange Multiplier, let

$$F(\alpha) = f(\alpha) - m * \left(\sum_{i \in \mathbf{N}} \alpha_i - \frac{1}{2} \right) \quad (4.8)$$

The extremum of $f(\alpha)$ subject to the constraint given by Equation (4.7) should satisfy:

$$\begin{cases} \frac{\partial F}{\partial \alpha_i} = \frac{\partial f}{\partial \alpha_i} - m = 0 \\ \sum_{i \in \mathbf{N}} \alpha_i - \frac{1}{2} = 0 \end{cases}, \quad \forall i \quad (4.9)$$

Since $\frac{\partial f}{\partial \alpha_i} = \frac{1}{(1 - \alpha_i)^2}$, we obtain the extremum of $f(\alpha)$, which is $f(M_0) = \sum_{i \in \mathbf{N}} \left(\frac{2N}{2N-1} - 1 \right) = \sum_{i \in \mathbf{N}} \frac{1}{2N-1} = \frac{N}{2N-1}$, where $M_0 = \{\alpha_i, \alpha_i = \frac{1}{2N}, i = 1, 2, \dots, N\}$.

Next we determine whether this extremum is the maximum or minimum extremum. First, let matrix $B = (b_{ij})_{N \times N}$, where $b_{ij} = F''_{\alpha_i \alpha_j}(M_0)$, $i, j = 1, 2, \dots, N$. We have another condition $\gamma_i \geq 0$. Thus

$$0 \leq \alpha_i = \frac{\gamma_i}{\gamma_i + 1} < 1 \quad (4.10)$$

From Equation (4.9), it is easy to find that

$$F''_{\alpha_i \alpha_j} = f''_{\alpha_i \alpha_j} = \begin{cases} \frac{2}{(1 - \alpha_i)^3} > 0, & i = j \\ 0, & i \neq j \end{cases} \quad (4.11)$$

We can find that the objection function is a convex function. We also have

$$b_{ij} = \begin{cases} \frac{2(2N)^3}{(2N-1)^3} > 0, & i = j \\ 0, & i \neq j \end{cases} \quad (4.12)$$

Obviously B is positive definite ($|B| > 0$) and hence M_0 is a minimum extremum point. Then the maximum value of $f(\alpha)$ should exist at the boundary. The boundary of $f(\alpha)$ is given by

$$\alpha_i = \begin{cases} \frac{1}{2} & i = j \\ 0, & i \neq j \end{cases}, \quad i = 1, 2, \dots, N; j \in \mathbf{N} \quad (4.13)$$

and the value of $f(\alpha)$ at boundary is

$$f(\alpha_i = \frac{1}{2}) = 1 \quad (4.14)$$

In conclusion, we should choose only one user in a time slot and exploit total available power to transmit data as fast as possible, when the maximum transmission rate is proportional to SIR. ■

Next, we consider a looser condition. Suppose $R_i = g(\gamma_i)$ and $g(\cdot)$ satisfies $r_0 = g(\gamma_0)$. We have the following theorem.

Theorem 4.2 Suppose $g(\cdot)$ is an increasing function and the increasing rate of $g(\cdot)$ is high enough, such that $g(1) > N * g(\frac{1}{2N-1})$. $\sum R_i$ reaches the maximum value with TDM deployed.

Proof: In this case, since $g(\cdot)$ is an increasing function with $g'(\gamma_i) > 0$, we have $\frac{\partial f}{\partial \alpha_i} = \frac{g'(\gamma_i)}{(1-\alpha_i)^2} > 0$. Similar to Equation (4.8), $M_0 = \{\alpha_i\}, \alpha_i = \frac{1}{2N}, i = 1, 2, \dots, N$ is also the extremum point. Then the extremum is

$$\begin{aligned} f(M_0) &= \sum_{i \in \mathbf{N}} g\left(\frac{2N}{2N-1} - 1\right) \\ &= \sum_{i \in \mathbf{N}} g\left(\frac{1}{2N-1}\right) = N * g\left(\frac{1}{2N-1}\right) \end{aligned} \quad (4.15)$$

The value of $f(\alpha)$ at the boundary is given by

$$f(\alpha_i = \frac{1}{2}) = g(1) \quad (4.16)$$

Regardless that $f(M_0)$ is the maximum or minimum extremum, if $g(1) > N * g(\frac{1}{2N-1})$, the maximum value of $f(\alpha)$ exists at the boundary shown in Equation (4.16). Obviously we have the same conclusion as Theorem 4.1. Unlike in Theorem 4.1, we do not consider convexity of the object function here. ■

Table 4.1: Transmission rate vs SIR value

SIR $\geq$	Rates(Kb/s)
-12.5	38.4
-9.5	76.8
-8.5	102.6
-6.5	153.6
-5.7	204.8
-4.0	307.2
-1.0	614.4
1.3	921.6
3.0	1228.8
7.2	1843.2
9.6	2457.6

Let us use an example to illustrate Theorem 4.2. Table 4.1 shows the relationship between transmission rates and SIR value (in dB) in CDMA 1xEV-DO system [18]. For instance, $N = 5$, according to Equations (4.15) and (4.16), we have

$$\gamma_{M0} = \frac{\gamma_{M1}}{2N-1}, \quad (4.17)$$

where γ_{M0} is the SIR of a user at M_0 point, and γ_{M1} is the SIR value at boundary $\alpha_i = \frac{1}{2}$. If we measure SIR value in dB, Equation (4.17) becomes $\gamma_{M0} = \gamma_{M1} - 10\log 9 = \gamma_{M1} - 9.5$. For example, with $\gamma_{M1} = 3 \text{ dB}$, $\gamma_{M0} = -6.5 \text{ dB}$. According to Equations (4.15) and (4.16),

$$\begin{cases} f(M_0) = 5 * g(-6.5 \text{ dB}) = 5 * 153.6 = 768 \text{ Kb/s} \\ f(\alpha_i = \frac{1}{2}) = g(3 \text{ dB}) = 1228.8 \text{ Kb/s} \end{cases} \quad (4.18)$$

Obviously, TDM for high speed CDMA has higher throughput in this case and hence Theorem 4.2 holds.

4.3 Channel States Dependent Fair Service for CDMA (CSDFSC)

Motivated by the above finding, we propose a new packet scheduling scheme, namely Channel States Dependent Fair Service for CDMA (CSDFSC). CSDFSC is designed for a high data rate CDMA wireless network to achieve service differentiation, fairness among best-effort traffic flows, and high channel throughput with low implementation complexity. In CSDFSC, we assume that system operates in the hybrid TDMA/CDMA mode, in which TDM is deployed in the downlink and CDM is deployed in the uplink. The downlink scheduling decision is made at each time slot and only one user can transmit data in each time slot.

We also assume that the channel state follows the multi-state Markov channel model, and the base station is aware of the state information from the channel feedback. The two-state Markov channel model is widely used in previous research for its simplicity and efficiency to describe wireless channel characteristics. However, in CDMA the situation is more complicated since the user's SIR is related to its transmission rate in CDMA networks. With different code and higher power allocation, a mobile user can transmit in lower but non zero rate under a "Bad" channel state. Thus, the multiple states Markov chain model [35] is more accurate for describing CDMA channel characteristics. In CSDFSC, we employ a three-state Markov chain for modeling channel states.

As shown in Figure 4.1, CSDFSC consists of two main components: Flow Classifier and Scheduler (FCS), and Frame Structurer (FS). The arriving data are classified at the base station and put into different queues. FCS chooses the transmission flow according to the channel states, bandwidth allocations and flow weights. FS organizes the head of line (HOL) packets of selected flows into a data frame, which may have different size in different systems, and sends them out.

With the channel states, weights of flows and bandwidth allocation, the priorities of flows are calculated as:

$$prio = W_i * (Z_{\max}(t) - Z_i(t)) * R_i(t), \quad (4.19)$$

Figure 4.1: CSDFSC scheduler framework

where W_i is the predefined weight of flow i , $Z_i(t) = \frac{Y_i(t)}{\phi_i}$ and $Z_{\max}(t) = \max\{Z_i(t)\}$, and $R_i(t)$ is the maximum feasible transmission rate of flow i at time t . The priority calculation in Equation (4.19) includes three parts: flow weight, relative throughput difference and current transmission rate. Therefore, the priority is determined by user pre-definition [13], fairness constraint and channel states. In our implementation, $Y_i(t)$ is calculated at the end of each time slot. Suppose flow j is selected to transmit data in this time slot, then only $Y_j(t)$ will be updated after the transmission completed. It's necessary and efficient since the scheduling procedure makes decision each time slot. Another way is to update $Y_i(t)$ packet by packet, which needs more computation.

In this formula, the term $Z_{\max}(t) - Z_i(t)$ is the difference between the service received by a specific flow and the maximum service, which reflects the fairness among flows. The current channel states are reflected by $R_i(t)$. The predefined weight is used to differentiate guaranteed flow from best-effort flow.

When the next frame needs to be scheduled, CSDFSC assigns the flow with the highest priority in the first time slot and updates this flow's priority after allocation. The above procedure is repeated until all time slots in this frame are fully occupied, as shown in Figure 4.2.

In order to achieve low implementation complexity, CSDFSC uses a simple priority equation to determine the sequence of transmitting packets. As a result, the implementation complexity is much less than that of WISPER. In WISPER, the number of packets transmitted and the number of mobile users in a time slot are not fixed; scheduler should compute them according to the BER requirements of flows. Furthermore, the procedure of allocating packets into a frame is also very complicated. To avoid the complex virtual service time computation which is essential in GPS

and similar scheduler in wireline networks, CSDFSC tries to emulate GPS by calculating relative difference, represented by $Z_{\max}(t) - Z_i(t)$, of received service between two flows in computing the flows' priorities. The flow that has obtained more transmission opportunity will have lower priority than a flow that has shared less transmission opportunity. Then the fairness between flows will be achieved to a certain extent.

Figure 4.2: Frame allocation procedure in CSDFSC

Now we use an example to illustrate the scheduling procedure in CSDFSC. Let there be three flows f1, f2 and f3. The queue weights and assigned proportion of each flow are listed in Table 4.2. At time 0, there are 14, 6 and 6 packets belonging to f1, f2 and f3, respectively. The maximum transmission rates, in unit of packets, of each flow from time slot 1 to 8 are listed in Table 4.3.

Table 4.2: Parameters of flows

	f1	f2	f3
Queue weight(W_i)	2	1	1
Proportion(ϕ_i)	0.5	0.25	0.25

The process of making scheduling decisions in CSDFSC is interpreted as follows. At time 0, all flows' priorities are equal to zero, as shown in Table 4.4. The scheduler chooses the flow with the largest number of packets, f1, and the maximum transmission rate of f1 at time slot 1, $R_1(1)$, equal to 4 packets, and thus at time slot 1 flow f1 transmits 4 packets, as shown in Table 4.5. Table 4.6 shows the $Z_i(t)$ of each flow during the scheduling process. After time slot 1, the throughput of flow

f1 is changed to 4. According to Equation (4.19) the priority of flow f2 is changed to 16 because $R_2(2) = 4$, $Z_{\max}(2) - Z_2(2) = 4$ and $W_2 = 1$. The priority of f3 remains 0 because $R_3(2) = 0$. Hence flow f2 is chosen to transmit at time slot 2. Repeating this procedure, all packets of flows are transmitted after time slot 8 and no packet is lost during transmission.

Table 4.3: Maximum transmission rates of flows

	Slot 1	Slot 2	Slot 3	Slot 4	Slot 5	Slot 6	Slot 7	Slot 8
f1	4	4	4	2	2	0	4	4
f2	2	2	0	0	4	4	4	4
f3	2	0	4	4	4	2	2	0

Table 4.4: Priority of flows

	Slot 1	Slot 2	Slot 3	Slot 4	Slot 5	Slot 6	Slot 7	Slot 8
f1	0	0	0	32	16	0	96	32
f2	0	16	0	0	32	0	0	0
f3	0	0	32	0	0	16	0	0

Table 4.5: Throughput of flows

	Slot 1	Slot 2	Slot 3	Slot 4	Slot 5	Slot 6	Slot 7	Slot 8
f1	4	0	0	2	0	0	4	4
f2	0	2	0	0	4	0	0	0
f3	0	0	4	0	0	2	0	0

Table 4.6: $Z_i(t)$ of flows

	Slot 1	Slot 2	Slot 3	Slot 4	Slot 5	Slot 6	Slot 7	Slot 8
f1	0	8	8	8	12	12	12	20
f2	0	0	8	8	8	24	24	24
f3	0	0	0	16	16	16	24	24

4.4 Simulation Results

In this section, we compare the performance of CSDFSC with that of WISPER and WRR by using ns-2 simulations on a typical network topology. WISPER is a known CDMA scheduling algorithm, the comparison results will be used for verifying Theorems 4.1 and 4.2. Moreover, WISPER is proven to be efficient in terms of throughput. WRR assigns different queue weight values for different flows, which is similar to CSDFSC. In wireline networks, WRR is proven to be efficient in throughput and fairness, and is widely used as comparison base in wireless scheduling research [93] [79]. The performance metrics we evaluate include the number of users the system can support, packet loss, link utilization, fairness, *etc.* The traffic in the simulations experiments include VBR video and TCP flows.

Figure 4.3 depicts the network topology used in our simulation experiments. There are two cells in the network and each cell has a basestation and multiple video and data users. One cell is “destination cell” which contains a base station and some destination mobile stations while the other is used as “source cell” which contains a base station and some source nodes. Traffic flows are transmitted from source nodes to destination nodes, thus in destination cell there is only downlink traffic. Since our theoretical analysis is concerned with downlink traffic in a single cell, we focus on the analysis of the traffic transmission within the destination cell. The link bandwidth between two basestations is 4Mb/s, and all link bandwidth between mobile users and basestation is 10Mb/s. Multi-state Markov model is used to define wireless channel in our simulation. The parameters of the link model are listed in Table 4.7. The same traffic weight values in this table are also used in WRR.

Two types of traffic are used in our simulation. One is VBR video traffic, which is modeled as three-state on-off source. At different state, packets are generated at different data rate. Another is TCP data traffic, which simulates the transmission of a very large file via FTP. The VBR video traffic has higher traffic weight than TCP traffic. The parameters of two types of traffic are listed in Table 4.8. We use the same WISPER parameters (Table 4.9) of these two types of traffic as those in [13].

In the first experiment, we compare the maximum number of video and TCP data users which the forward link is able to support under CSDFSC with that under WISPER and Weighted Round

Figure 4.3: Network topology for simulation experiments

Table 4.7: Parameters of channel model

Parameters	Value
Number of states	3
Transition matrix	$\begin{bmatrix} 0.9995 & 0.0005 & 0 \\ 0 & 0 & 1 \\ 1 & 0 & 0 \end{bmatrix}$
State 1 average duration time	0.1 second
State 2 average duration time	0.01 second
State 3 average duration time	0.01 second
Transmission rate in state 1	4 packet per slot
Transmission rate in state 2	2 packet per slot
Transmission rate in state 3	0 packet per slot

Robin (WRR) schemes. In this experiment, the link with packet loss ratio of video users larger than 1% is said to be “out of capacity”. Otherwise the link is said to be “in capacity”. The maximum number of video and data users when the link remains “in capacity” is called “maximum traffic load”. The curves in Figure 4.4 show the relationship between the number of TCP users and the number of video users the system is able to support simultaneously under “maximum traffic load”. For example, CSDFSC can support nine TCP users in addition to two video users. In comparison, WISPER and WRR can support only additional five TCP users as shown in Figure 4.4. With seven video users, CSDFSC can support other two TCP users while WISPER and WRR cannot support any more TCP user. Obviously, the link can support more users with CSDFSC than with WISPER and WRR. This is because that CSDFSC addresses the location-dependent and burst error, while WISPER and WRR do not consider the channel state.

Table 4.8: CSDFSC parameters of traffic

Parameters		Value
Video	State1 transmission rate	500 Kb/s
	State2 transmission rate	400 Kb/s
	State3 transmission rate	200 Kb/s
	State1 probability	0.5
	State2 probability	0.3
	State3 probability	0.2
	Average On-time	1 second
	Average Off-time	0.1 second
	Packet size	256 bytes
	Traffic weight	2
TCP	Type	Reno
	Window	40
	Packet size	256 bytes
	Traffic weight	1

Table 4.9: WISPER parameters of traffic

Parameters		Value
Video	Maximum Tolerable BER	10^{-6}
	Maximum Number of Packets per Slot S	6
	Packet timeout value	3 frames
	Maximum Transmission Capacity	4 packets/slot
TCP	Maximum Tolerable BER	10^{-9}
	Maximum Number of Packets per Slot S	4
	Packet timeout value	$(2 * total_packets)$ frames
	Maximum Transmission Capacity	4 packets/slot

Next, we compare the packet loss ratio of flows in two schemes when the numbers of the video and TCP users reach the “maximum traffic load” values. As shown in Figure 4.5, the packet loss ratio of TCP users remains a low level in CSDFSC. In comparison, in WISPER and WRR TCP users experience significantly higher packet loss ratio, which will impede the transmission of TCP traffic. This is mainly because that WISPER and WRR do not use the channel state information. When the channel state is not “good”, all transmitting packets will be lost in WISPER and WRR. In CSDFSC, on the other hand, the scheduler will lower current user’s transmission rate or schedule another user with good channel state to transmit. Another observation is that the packet loss ratio of video users is below the minimum requirement 1%, in all schemes, as shown in Figure 4.6. In

Figure 4.4: Number of users CSDFSC and WISPER can support

CSDFSC, the packet loss ratio increases with the number of video users, while it is relatively stable in WISPER and WRR.

Figure 4.5: Comparison of packet loss ratio of TCP traffic

Next, we compare the link utilization between WISPER, WRR and CSDFSC schemes when the system is offered with “maximum traffic load”. As shown in Figure 4.7, CSDFSC achieves higher bandwidth utilization than WISPER, but slightly lower than WRR. Although WRR can achieve the highest link utilization, it can only support the smallest number of users. The difference of link utilization between WISPER and CSDFSC becomes more significant when the number of

Figure 4.6: Comparison of packet loss ratio of Video traffic

video users is greater than four. Higher packet loss ratio implies not only the waste of transmission bandwidth, but also the impairments of transmission rate of TCP traffic.

Figure 4.7: Link utilization under maximum traffic load

Fairness among best-effort traffic (defined in Equation (4.2)) is also an important performance metric of scheduling scheme. We evaluate the fairness performance of CSDFSC in the scenario of two video users and two TCP users. In Figures 4.8 and 4.9, the transmission rates of two TCP users under two different schemes are described. Although the transmission rates of TCP users in WISPER are almost twice that in CSDFSC due to the lack of control on TCP traffic in WISPER,

we only consider the relative rate difference since we focus on fairness among TCP flows under the same scheme in this chapter. The two straight lines, which represent the average transmission rates of two TCP traffic flows, are very close in CSDFSC scheme. However, these two lines are separated far away in WISPER scheme. This means that the CSDFSC scheme can provide better fairness performance for data users than WISPER.

Another difference between two schemes is that the TCP transmission rate is more stable in CSDFSC than that in WISPER. Moreover, TCP traffic flows are under quite good control in CSDFSC scheme. They do not try to occupy all available bandwidth as they do in WISPER.

Figure 4.8: TCP transmission rate in CSDFSC

Figure 4.10 depicts the cumulative distribution function (CDF) of end-to-end packet delay for both video and TCP traffic under CSDFSC and WISPER, respectively. We can see that video users experience shorter delay in CSDFSC, at the expense of longer delay for TCP data users. In WISPER, the service received by guaranteed(video) traffic has no much difference from the best-effort(TCP) traffic. However, using different queue weights in CSDFSC, guaranteed and best-effort traffic flows are well differentiated. These two kinds of traffic flows will be served at different levels. We can observe that the end-to-end delay of guaranteed traffic flow in CSDFSC is less than that in WISPER.

Next we compare channel utilization of the two schemes when there are only video users in the system. Since the video model used in our simulation has rate limitation-500Kb/s, the channel

Figure 4.9: TCP transmission rate in WISPER

Figure 4.10: Cumulative distribution function (CDF) of packet delay

utilization increases with the number of video users. As shown in Figure 4.11, the channel utilization ratios are very close between two schemes, while CSDFSC achieves higher utilization as WISPER experiences more packet loss. The channel utilization increases with the number of video users since each user needs limited bandwidth. However, when number of video users exceeds the x-axis value of the right end point, the packet loss rate of video users will be larger than 1%, while implies that the system cannot support them.

Finally we compare the channel utilization of the two schemes when there are TCP users only.

Figure 4.11: Channel utilization vs video users

TCP can adjust transmission rate according to the available channel bandwidth. Thus small number of TCP users can reach high channel utilization, as shown in Figure 4.12. From the same figure, we can also find that channel utilization is quite stable while the number of TCP users increases in CSDFSC, although the utilization drops slightly when number of TCP users is more than 6. On the other hand, the channel utilization varies largely when number of TCP users increases in WISPER. The main cause is also the higher packet loss in WISPER. When the number of TCP users is small, each TCP user should have high transmission rate. Unfortunately, packet loss has more dominating influence on TCP users than on video users, especially on high transmission rate TCP users. Thus small number of TCP users cannot achieve high transmission rate and high channel utilization in WISPER. Furthermore, because of the same reason, the difference of channel utilization between two schemes with TCP users is larger than that with video users, as shown in Figures 4.11 and 4.12. Because TCP users can adjust transmission rate according to available bandwidth, the right end points of curves in Figure 4.12 are not the maximum number of TCP users which can be supported.

Figure 4.12: Channel utilization vs TCP users

4.5 Conclusion

In this chapter, we have investigated the appropriate multiplexing method for downlink in high speed CDMA network. According to our mathematical analysis, TDM instead of CDM should be used in most situations. In other words, one user transmitting in a time slot at full rate is more efficient than multiple users transmitting simultaneously in a time slot with lower transmission rates.

Based on the above conclusion, we propose a new packet scheduling scheme designed particularly for high speed CDMA network, namely Channel States Dependent Fair Service for CDMA (CSDFSC). CSDFSC uses multi-state Markov model to define a CDMA wireless link. We compare the performance of CSDFSC with that of WISPER and WRR via ns-2 simulations. The numerical results show that CSDFSC is more efficient in terms of channel capacity, link resource utilization, packet loss, fairness among best-effort traffic flows, end-to-end delay of guaranteed traffic. The good properties in providing fair service to TCP flows shows that CSDFSC emulates GPS quite well. In WISPER, guaranteed traffic experiences a lower end-to-end delay than best-effort traffic but the difference is not significant. However, in CSDFSC guaranteed traffic experiences a lower end-to-end delay than in WISPER and the average delay is much lower than that of best-effort traffic. From these results, we can draw the conclusion that the queue weight setting in CSDFSC

is efficient and guaranteed traffic will obtain better service.

Chapter 5

Efficient Rate Guaranteed Opportunistic Scheduling for Wireless Networks

We have introduced two different wireless packet scheduling schemes in the previous two chapters. However, these two schemes have a common weakness—without theoretical performance analysis. In order to analyze the performance bounds of wireless packet scheduling scheme, we try to apply service curve model to wireless scenario. In this chapter, we present an efficient Rate-guaranteed Opportunistic Scheduling(ROS) scheme which optimizes the system throughput performance while satisfying the “fairness” constraint for resource allocation by exploiting time-varying channel states. We prove the optimality of the proposed scheduler via mathematical analysis and show that it has a lower computational complexity compared with existing opportunistic scheduling algorithms. We define a new concept of Guaranteed-Rate(GR) node with loss, which is extended from the service curve with loss model, to analyze the performance bounds of ROS, such as the statistic end-to-end packet delay bound. We validate the mathematical analysis and demonstrate the throughput performance improvement and delay bound based on ns-2 simulations.

5.1 Introduction

It is expected that future wireless networks will support multi-class of traffic, including video, voice and data. A scheduler is essential to support Quality of Service (QoS) which is required by multimedia applications in wireless networks. It should be designed to dynamically allocate network resources to mobile hosts according to the varying traffic load and channel states, to be efficient

in utilization of network resources, and to be fair in scheduling service with a low implementation complexity.

In wireline networks, many existing scheduling algorithms, such as fair queuing scheduling [31, 46, 73, 91], virtual clock [94], etc. are capable of providing certain guaranteed QoS. However, these service disciplines are not directly applicable to wireless networks because their design has not considered some particular problems in wireless networks, such as varying wireless link capacities, location-dependent and burst errors, joint scheduling of uplink and downlink flows, power constraint in wireless cells, etc.

In recent development of cellular communication systems, such as CDMA/HDR(High Data Rate) proposed by Qualcomm [18], multiple users share the common transmission media on which data transmissions are scheduled. In CDMA/HDR, TDMA is deployed on downlink, and the transmission time is divided into fixed time-slots with duration of 1.67ms [18]. Though all users share the common downlink channel, only one user is scheduled to transmit data at a time in a time-slot. The possible transmission rate in a time-slot for a user is determined by the SNR experienced by this user. The decision of choosing which user to transmit in next time-slot is made on the basis of possible transmission rates of users as well as past transmission history. The time-scheduling approach is suitable in this hybrid TD/CDMA system. In CDMA/HDR system, scheduler can measure and estimate SNR of mobile users by sending pilot signal [18]. Since the length of time-slot is rather short, the scheduler can take into account Rayleigh fading with a frequency of a few tens of Hertz. Motivation of our work in this chapter partially comes from the QoS requirements in these systems.

In this chapter, we address two key issues for wireless scheduling in recent cellular communication systems such as CDMA/HDR: fairness among mobile users and end-to-end delay bound for flows. The fairness issue in wireless networks is much more complicated than that in wireline networks due to the error-prone wireless channel conditions. Delay bound is a critical QoS requirement for realtime applications. In the following we briefly discuss these two issues.

There is a tradeoff space between individual user satisfaction level and wireless resource efficiency. By employing measurement and estimation, scheduler may be aware of channel states fairly well [18]. When the channel of one mobile terminal is in bad state, and a packet is transmitted

to this terminal, this packet is corrupted and the transmission will waste transmission resources. Deferring transmission of this packet until the channel recovers from the error state is obviously a reasonable choice to achieve higher utilization of wireless resources such as bandwidth. This improvement of resource efficiency is very significant, especially for high data rate applications such as video, since bandwidth is a scarce resource in wireless networks. However, the improvement can not be achieved by all users simultaneously. Some of them may be satisfied while the others not. A user with error channel state temporarily loses its share of the transmission bandwidth and another user with good channel state “steals” more share of bandwidth. Furthermore, the users with a long distance to the base-station or high disturbance from other signal sources will be sacrificed while high transmission opportunity is allocated to those users near the base-station. For ensuring fairness, a user that loses share of bandwidth should be compensated later. However, how to compensate for it is not a trivial problem. In order to resolve this dilemma, we design a new scheduler which can optimize system throughput by exploiting channel states, while considering the fairness constraint in resource allocation among users. As one building block of the proposed conjuncted scheduling/admission control framework for wireless networks, our scheduling scheme is motivated by the work of [55], in which the authors present a solution to maximize system performance value by exploiting channel states while satisfying time-fraction assignment. However, that scheme has a high complexity in evaluating parameters. The authors also do not provide analysis of performance bound in the paper. Besides, there are some studies [60, 63, 70] which extend scheduling algorithms for wireline networks to wireless networks. These algorithms provide short-term or long-term throughput bound for incoming flows. At the same time, they are able to guarantee short-term or long-term fairness among flows.

In wireline networks, for input flows which are regulated by certain arrival curves (such as leaky bucket), we can derive some performance bounds such as delay bound using the service curve provided by the scheduler [22]. However, it is impossible to provide deterministic performance bound for a flow without any traffic regulation, since the flows being serviced may have unlimited burstiness of traffic. In the service curve model, the incoming flow of a network element is regulated by arrival curve $\alpha(t)$, which is an increasing function. The service received by a flow is defined by service curve $\beta(t)$. Three most important bounds, backlog bound, delay bound and output flow

bound, are analyzed in [22]. The Guaranteed-Rate (GR) node defined in service curve model [22], is proved to be able to provide guaranteed resource allocation (transmission rate) for users. The corresponding service curve for the GR node is a rate latency function: $\beta_{r,v} = rt - rv$, where r represents the service rate and v represents the service latency and t is the time. The variability of wireless channel state makes it difficult to guarantee QoS in wireless network. Nevertheless, QoS should be guaranteed for the flows, either deterministically or statistically, on those links that have bounded errors. In [17], the authors proposed a service curve model with loss, which can be used to describe wireless links and evaluate performance bounds for flows. In order to analyze the statistical performance bounds, especially delay bound, for our new packet scheduling scheme, we bring forward the definition of GR node with loss, which is based on the concept of service curve with loss model.

In this chapter, we present an efficient rate guaranteed opportunistic scheduling (ROS) scheme for wireless networks. The design goal is to achieve maximum system throughput while satisfying the resource allocation constraint, with a low implementation complexity. The fairness among users has various definitions [84] [67]. In this chapter, we use transmission rate to judge fairness (as defined in Equation (5.6)). If all users are provided with guaranteed transmission rates, the resource allocation fairness is supposed to be achieved. Our scheme employs a new algorithm to satisfy resource allocation constraint and achieves the same optimality in system throughput performance as [55] while using a simpler parameter evaluation approach. The optimality of our scheme is proved by mathematical analysis and also validated by simulation results.

To analyze the service curve provided by ROS, we first prove that a network node performing ROS is a GR node with loss. We employ the service curve with loss model to derive the statistical end-to-end packet delay bound in ROS. In the system, a path can be deemed as a concatenation of GR nodes with loss when all network nodes along this path employ ROS scheme. Supposing incoming flows are regulated by leaky bucket (σ, ρ) , we solve the statistical delay bound problem for concatenation of GR nodes with loss. Our simulation results also illustrate the existence of the upper delay bound in ROS. In order to satisfy the “fairness” constraint, we employ the measure “relative throughput”, which is defined as the throughput divided by the predefined weight. In more details, the difference of “relative throughput” among users is used as a core parameter in

the operation of ROS. The greater this parameter, the more unfairness service received by users. We also analyze the variation range of this parameter which is useful for computing the packet loss rate. The theoretical analysis and simulation results show that there is a upper bound for this parameter. Furthermore, we compare the computational complexity of ROS with that of existing schemes and prove that ROS has a lower scheduling complexity.

The rest of chapter is organized as follows. We describe the system model in Section 5.2. Section 5.3 presents our new scheme - Rate-guaranteed Opportunistic Scheduling(ROS). In Section 5.4, we prove the optimality of ROS and analyze the delay bound provided by ROS. In additions, we derive the distribution of maximum difference of “relative throughput” among users ($D(k)$) and the packet loss probability in this section. We examine the behaviors and performance of ROS via simulation in Section 5.5 and finally conclude the chapter in Section 5.6.

5.2 System Model

We consider a time-division multiple access (TDMA) wireless system or the downlink of hybrid TD/CDMA systems, such as the IS856 system or HDR [18], in which time is divided into fixed length (T_s) called time-slot. In TDMA systems, each time slot is allocated to only one user for transmitting data, although there could be more than one channels (frequency bands) in a cell. We assume that the channel is quasi-static in time, i.e., the channel states keep stable within a given time-slot. Mobile stations can feedback the received signal-to-noise ratio to the base-station without loss and delay, thus the base-station has the knowledge of current channel states for all mobile users. These assumptions are also used in relevant researches [87]. In this chapter, we focus on packet scheduling problem for a single channel. The scheduler chooses one user at the beginning of each time-slot, and this user transmits its Head of Line (HOL) packets in buffer.

5.2.1 System Performance Value

As mentioned before, wireless networks have a particular characteristic - time varying channel as the channel is error-prone and suffers from interference, fading and shadowing. When a user transmits data successfully, the system gains some performance value. The system performance value is thus also time-varying and dependent on channel states. Due to different objectives of

system design, the system performance value may have various definitions, such as throughput, power consumption etc. In this chapter, we use the throughput as the system performance value, since it is one of the most important performance metrics in wireless networks.

We define a random process $\{G_i^k\}$ as the performance value of user i , where G_i^k represents the “gain” of system after the transmission of user i in time-slot k . In general $\{G_i^k\}$ is a widely increasing function of channel SNR perceived by user i . In this chapter, we focus on the problem of throughput optimization, thus the “gain” of system in a time-slot is represented by the amount of transmitted data in this time-slot. Defining R_i^k to be the maximum available transmission rate for user i in time-slot k , then

$$G_i^k = R_i^k, \quad i = 1, 2, \dots, N; k = 1, 2, \dots$$

where N is the total number of users in the system. Wireless channel can be described fairly well by the multiple-state Markov model [35]. Since R_i^k is determined by channel state of user i at time-slot k , $\vec{G}$ is assumed to be a Markov stationary stochastic process in this chapter.

5.2.2 GR Node with Loss

It is known that General Processing Sharing (GPS) and its deviation PGPS [73] or weighted fair queueing (WFQ) [31] are efficient fair scheduling schemes which can dynamically allocate bandwidth (or service rate). By facilitating statistical multiplexing, GPS has two important characteristics in service for end users: (1) “isolation”- service received by one user will not be affected by the failed transmission of another user; (2) “fairness”- service received by each user is in proportion to the assigned weight. Assume flow i is assigned a positive constant (acting as a queueing weight) ϕ_i and the amount of service received by flow i during the interval $(\tau, t]$ is denoted by $O_i(\tau, t)$. If flow i is continuous backlogged during interval $(\tau, t]$, it holds for GPS [73] that,

$$\frac{O_i(\tau, t)}{O_j(\tau, t)} \geq \frac{\phi_i}{\phi_j}$$

GPS and PGPS have been proven to be able to provide rate guarantee service and a node performing GPS or PGPS is a GR node [22]. In the service curve model [22], a network node with

a scheduler is defined to be GR (Guaranteed Rate) node as follows. Consider a node that services a flow. Packets are numbered in the order of arrival. Let $a_n \geq 0, d_n \geq 0$ be the arrival and departure times of packet n of a flow respectively and l_n is its length. A node is said to be a guaranteed rate (GR) node for this flow, with rate r and delay e , if it guarantees that $d_n \leq f_n + e$, where f_n is defined as

$$\begin{cases} f_0=0 \\ f_n=\max\{a_n, f_{n-1}\} + \frac{l_n}{r} \quad \text{for all } n \geq 1 \end{cases} \quad (5.1)$$

The GR node is a kind of basic but important scheduler. In addition to providing guaranteed service rate for flows, it can maintain resource allocation and fairness among flows as well. Moreover, it can provide some performance bounds such as delay bound, backlog bound and output bound, for regulated flows. Providing these performance bounds is an important requirement for a scheduler to provide QoS guarantee.

However, the GR node definition cannot be directly applied to wireless networks since the channel capacity is not fixed and packets may be lost during transmission in wireless scenario. A service curve model with loss is proposed in [17]. “The service curve S with loss parameter $1 - \alpha$ is said to be delivered to an input flow, if and only if at least α fraction of all packets meet the deadlines assigned via S .” We extend the definition of service curve with loss to define GR node with loss as follows.

Definition 1 Consider a node that serves a flow. Packets are numbered in order of arrival. Let $a_n \geq 0, d_n \geq 0$ be the arrival and departure times. A node is called a guaranteed rate (GR) node with loss parameter $1 - \alpha$ for this flow, with rate r and delay e , if at least an α fraction of all the packets of this flow satisfy that

$$d_n \leq f_n + e \quad (5.2)$$

where f_n is defined as Equation (5.1).

In the service curve with loss model, when a flow is served through a network node, at least an α fraction of packets will meet their deadline, which is the same as in the original service curve model, assigned via a service curve. This model is also proved to be composable [17], which means that the service delivered by a tandem of two nodes can be described by the combination of the

service delivered by each one of the nodes. Thus, we can say there is a statistic delay bound, which has the same value as in the original service curve model, for a flow served by a network of GR nodes with loss.

5.2.3 Problem Formulation

ROS aims to achieve the maximum throughput while all users can share bandwidth fairly even they may experience different channel states. The design objective is to find a solution to the problem

$$\text{maximize } \sum_i E(O_i(t)), \quad t \rightarrow \infty \quad (5.3)$$

subject to the fairness constraint,

$$\frac{E(O_i(t))}{\sum_j E(O_j(t))} = \frac{\phi_i}{\sum_j \phi_j}, \quad i = 1, 2, \dots, N \quad t \rightarrow \infty \quad (5.4)$$

where $O_i(t)$ represents the throughput of user i from time zero to t , which is given by

$$O_i(t) = \sum_{k=0}^K R_i^k \delta(i, k), \quad t = KT_s$$

where K is the number of time-slots between $(0, t]$, T_s is the length of a time-slot and $\delta(i, k)$ is defined as

$$\delta(i, k) = \begin{cases} 1 & ; \text{ if user } i \text{ transmits in time-slot } k \\ 0 & ; \text{ otherwise} \end{cases}$$

We assume $t = KT_s$ as transmission time is divided to time-slots.

This maximization problem has a fairness constraint. The fairness constraints have various definitions. In this chapter, we consider time-fraction as the fairness criteria, i.e. on the average, user i should be allocated a fraction $\phi_i / \sum_j \phi_j$ of the whole transmission time. Our design goal is achieve system throughput maximization with the aid of time-varying channel condition knowledge, subject to the time-fraction fairness constraint.

Our scheduler is designed to work for both uplink or downlink of wireless networks. In a general situation, the scheduler is deployed at the base station. At the beginning of each time-slot, the

scheduler chooses a user to transmit according to its maximum possible transmission rate, R_i^k , which is determined by user's channel state(SNR) and SNR can be detected by the base station. After the selected user receives or transmits data in this time-slot, the system throughput is increased by the amount of data transmitted in this time-slot. The design goal is *an efficient scheduler which can maximize system throughput while performing as a GR node by exploiting known channel states, and provide some performance bound with a low computation complexity for wireless networks.*

5.3 Rate-guaranteed Opportunistic Scheduling

5.3.1 Rate-guaranteed Opportunistic Scheduling

In the following discussion, we consider a TDMA system described in Section 4.2. For simplicity, in this chapter we also model the channel states as a two-state Markov chain, and each user's channel state is independent with others, without loss of generality. In such a system, at "good" and "bad" states, the corresponding data rates are R and zero respectively. By using $\vec{R} = \{R_i\}$, which is a random variable, to denote the maximum number of packets of flow i which can be transmitted in a time-slot, we have

$$R_i = \begin{cases} R; & \text{if channel state is "good"} \\ 0; & \text{if channel state is "bad"} \end{cases}, \quad \forall i$$

Suppose scheduling scheme S is deployed at basestation, and $S(\vec{R})$ represents the scheduling result of S . If flow i is chosen by S , $S(\vec{R}) = i$. After flow i transmits data successfully, the system will obtain throughput gain $R_{S(\vec{R})} = R_i$. Then the expectation of system throughput in a time slot is

$$E(R_{S(\vec{R})}) = \sum_i R_i E(S(\vec{R}) = i)$$

The design objective of our scheduler can then be presented as

$$\text{maximize } E(R_{S(\vec{R})}) \quad (5.5)$$

Besides throughput optimization, we also consider resource allocation fairness guarantee in ROS design. It is well known that GPS can provide deterministic guaranteed fairness for users by using

the fluid model. In the service curve model, GPS and its deviation PGPS(or WFQ) are proved to be GR node and GR node can guarantee allocated transmission rate for all flows [22]. Our new scheduling scheme should also be a GR node with loss. That is, the following condition should be satisfied.

$$E(S(\vec{R}) = i) = \frac{\phi_i}{\sum_j \phi_j} \quad (5.6)$$

We denote by S^* our new scheduler, Rate-guaranteed Opportunistic Scheduling (ROS) scheme. It is defined as

$$S^*(\vec{G}) = \arg \max_i (G_i + x_i) = \arg \max_i (R_i + x_i) \quad (5.7)$$

where $G_i = R_i$, since throughput is the performance value we consider in this chapter, and

$$x_i = \max_j \{y_j(t)\} - y_i, \quad i, j = 1, 2, \dots, N$$

where $y_i(t)$ is the “relative throughput” which is defined as

$$y_i(t) = O_i(t)/\phi_i \quad (5.8)$$

where $O_i(t)$ represents the amount of data of flow i which has been serviced, ϕ_i is the normalized rate-fraction of flow i , i.e., $\phi_i = r_i/r_0$, $r_0 = \min\{r_i\}$.

Next we will prove the new scheme S^* can guarantee allocated bandwidth r_i to flow i and is thus a GR node. We will analyze the variance of x_i before the proof, as the result is necessary for the proof of GR node. We will first investigate the two-user case and then extend it to general multiple-user case.

5.3.2 Two Users Case

Let us start our discussion from a simple case where there are only two users in the system and both flows start to transmit at time zero, i.e. $O_1(0) = O_2(0) = 0$. Since time is divided into time-slots in the system, we will use index of time-slot k instead of t to represent time in the following analysis.

We define $D(k) = |y_1(k) - y_2(k)|$ to be the difference of “relative throughput” between two users. After time-slot k , we assume $y_1(k) > y_2(k)$ without loss of generality. If both flows 1 and 2

perceive “good” channel state, obviously $G_2 + x_2 > G_1 + x_1$ and flow 2 will transmit in time-slot $k + 1$. Thus

$$D(k+1) = \begin{cases} D(k) - R/\phi_2 & \text{if } y_1(k) > y_2(k) + R/\phi_2 \\ 0 & \text{if } y_1(k) = y_2(k) + R/\phi_2 \\ R/\phi_2 - D(k) & \text{otherwise} \end{cases} \quad (5.9)$$

From Equation (5.9), we can see that $D(k+1)$ will either decrease or come back to zero or jump to a value less than R/ϕ_2 . Obviously we have

$$D(j) < R/\phi_2, \forall j. \quad (5.10)$$

If flow 1 perceives “good” state while flow 2 perceives “bad” state, then

$$\begin{cases} G_1 + x_1 &= R + y_1(k) - y_1(k) &= R \\ G_2 + x_2 &= 0 + y_1(k) - y_2(k) &= D(k) \end{cases}$$

Now there are two possible situations, $R \leq D(k)$ or $R > D(k)$. If $R \leq D(k)$, flow 2 will transmit in time-slot $k + 1$, but this transmission will be lost. Therefore, $D(k+1) = D(k)$ which means $D(k+1)$ remains the same as $D(k)$. On the other hand, if $R > D(k)$, flow 1 will transmit in time-slot $k + 1$ and $D(k+1) = D(k) + R/\phi_1$. It is undesirable since $D(k+1)$ is increased. Let us investigate this situation carefully. With condition $R > D(k)$, we have

$$D(k+1) = D(k) + R/\phi_1 < R + R/\phi_1$$

which means $D(k+1)$ will not exceed the limit $R + R/\phi_1$.

If after time-slot k , it is true that $y_2(k) > y_1(k)$, we can obtain similar results. In conclusion,

$$D(j) < \max\{R + R/\phi_i, i = 1, 2\} = D_{max} \quad j = 1, 2, \dots$$

$D(j)$ is constrained by an upper bound, which means that the maximum difference of “relative throughput” will not exceed a constant value - D_{max} . Since we make the assumption that channel state of one user is independent with that of the others, the throughput fairness is equivalent with

time-fraction fairness. As a result, the fairness constraint in Equation (5.6) is satisfied.

5.3.3 Multiple Users Case

Suppose there are N users in the system. Similar to the two users situation, we define

$$D(k) = \max_i y_i(k) - \min_i y_i(k)$$

After time-slot k , we assume

$$y_n(k) = \max_i y_i(k) \quad \text{and} \quad y_m(k) = \min_i y_i(k)$$

Thus

$$D(k) = y_n(k) - y_m(k)$$

If flow m will transmit data in time-slot $k+1$, we have $y_m(k+1) = y_m(k) + \frac{R}{\phi_m}$ and $y_i(k+1) = y_i(k), i \neq m$. The $D(k+1)$ will be,

$$D(k+1) \leq \begin{cases} \frac{R}{\phi_m}, & \text{if } y_m(k+1) > y_n(k+1) \\ D(k), & \text{otherwise.} \end{cases}$$

The former inequality holds since $y_m(k) = \min_i y_i(k)$. Then $D(k+1) = \frac{R}{\phi_m} + y_m(k) - \min_i y_i(k+1) \leq \frac{R}{\phi_m}$. While the latter inequality is obvious since $\min_i y_i(k+1) \geq y_m(k)$.

Otherwise, suppose flow l will have the priority to transmit data in time-slot $k+1$, which means that

$$R \geq y_l(k) - y_m(k) \geq 0$$

thus

$$y_l(k+1) - y_m(k+1) = y_l(k) + R/\phi_l - y_m(k) \leq R + R/\phi_l \quad (5.11)$$

Since only $y_l(k+1)$ increases after time-slot $k+1$,

$$D(k+1) = \begin{cases} D(k) & ; \text{ if } y_l(k+1) \leq y_n(k) \\ y_l(k+1) - y_m(k+1) & ; \text{ if } y_l(k+1) > y_n(k) \end{cases} \quad (5.12)$$

Finally, by combining Equations (5.11) and (5.12) together, we can draw the same conclusion as two users situation.

$$D(j) < \max\{R + R/\phi_i, i = 1, 2, \dots, N\} \quad j = 1, 2, \dots \quad (5.13)$$

After proving that the difference of “relative throughput” among users has an upper bound, we can obtain the result that $E(S(\vec{R}) = i) = \phi_i / \sum_j \phi_j$. Next, we will prove our new scheduler is a GR node with service curve β_{r_i, v_i} , which is called rate-latency function. It is defined as $\beta_{r_i, v_i} = r_i t - r_i v_i$ for flow i . Based on that, we will deduce the delay bound for flows which are regulated by leaky bucket (σ_i, ρ_i) , where $\rho_i \leq r_i$.

Whether a schedule node is a GR node or not can be determined by the following theorem [22].

Theorem 5.1 *Consider a system where packets are numbered $1, 2, \dots$ in order of arrival. Let a_n, d_n be the arrival and departure times for packet n respectively, and l_n be its size. Define by convention $d_0 = 0$. The system is a GR node with rate r and latency e if and only if for all n there is some $k \in \{1, \dots, n\}$ such that*

$$d_n \leq e + a_k + \frac{l_k + \dots + l_n}{r}$$

In the following, we examine whether the proposed ROS satisfies the above condition or not. Suppose time t_0 is the last time that the scheduler has no backlog before packet n arrives, packet k is the first packet that arrives after time t_0 . Let there be N flows transmitting in system. The input flows and outputs of the scheduler are represented by I_i and O_i respectively. Furthermore, flow i is allocated with rate r_i while C is the capacity of output link for the scheduler. The differences between “relative throughput” of any two flows are less than D_{max} . According to our analysis of

D_{max} in Equation (5.13), we have

$$\begin{aligned} |y_i(d_n) - y_j(d_n)| &\leq D_{max}, |y_i(a_k) - y_j(a_k)| \leq D_{max} \quad \forall i, j \\ \Rightarrow (O_i(d_n) - O_i(a_k))\phi_j &\geq (O_j(d_n) - O_j(a_k))\phi_i - 2D_{max}\phi_i\phi_j \end{aligned}$$

Summed with only j varying, we have

$$\begin{aligned} (O_i(d_n) - O_i(a_k)) \sum_j \phi_j &\geq \sum_j (O_j(d_n) - O_j(a_k))\phi_i \\ &\quad - 2D_{max}\phi_i \sum_j \phi_j \end{aligned}$$

Since $\sum_j (O_j(d_n) - O_j(a_k)) = \sum_{t=a_k}^{d_n} c(t) = C(a_k, d_n)$, we define $C'_k = \frac{C(a_k, d_n)}{d_n - a_k}$ and $r'_{ik} = C'_k\phi_i / \sum_j \phi_j$, we have

$$\begin{aligned} (O_i(d_n) - O_i(a_k)) &\geq \frac{C'_k(d_n - a_k)\phi_i}{\sum_j \phi_j} - 2D_{max}\phi_i \\ &= r'_{ik}(d_n - a_k) - F_i \end{aligned}$$

where $F_i = 2D_{max}\phi_i$.

Thus we have

$$d_n - a_k \leq \frac{F_i}{r'_{ik}} + \frac{O_i(d_n) - O_i(a_k)}{r'_{ik}} = \frac{F_i}{r'_{ik}} + \frac{l_k + \dots + l_n}{r'_{ik}} \quad (5.14)$$

We define $\bar{c} = E(c(n))$ to be the expectation of $c(n)$, and $r_i = \bar{c}\phi_i / \sum_j \phi_j$. Then clearly we have $E(C'_k) = \bar{c}$. The probability that $\{C'_k < \bar{c}, \text{ for all } k \leq n\}$ will be very small. In other words, it is very possible to find some $k \leq n$ that satisfy $C'_k \geq \bar{c}$. Then we have $r'_{ik} \geq r_i$, and following inequation is valid.

$$d_n - a_k \leq \frac{F_i}{r'_{ik}} + \frac{l_k + \dots + l_n}{r'_{ik}} \leq \frac{F_i}{r_i} + \frac{l_k + \dots + l_n}{r_i} \quad (5.15)$$

Suppose $1 - \alpha = P\{C'_k < \bar{c}, \text{ for all } k \leq n\}$. According to Theorem 5.1, our scheduler is obviously a GR node with loss parameter $1 - \alpha$ with rate r_i and latency $\frac{F_i}{r_i}$.

5.3.4 Implementation Consideration

For each coming time-slot k , the ROS scheduler chooses the flow with the highest priority to transmit. The priority of flow is defined as

$$Prio_i(k) = R_i(k) + \max_j \{y_j(k)\} - y_i(k) \quad (5.16)$$

where $R_i(k)$ represents the number of packets that flow i can transmit in time-slot k . The throughput $\{y_i(k)\}$ is also counted in number of packets.

We assume continuous backlog in the previous analysis. However, this assumption is not always valid in real communication networks. If one of the traffic flows is not continuous-backlogged, other flows will be hanged since the throughput of all flows is strictly bounded. In order to apply our scheme in non-continuous backlogged situation, the rate ratio is re-computed when the “active” users are changed in the implementation, where the “active” users are the users currently have backlog.

We have another assumption that the base-station is aware of the channel states for mobile users. It is difficult to fully fulfill this condition, but we can use some mechanisms to approximate it. The channel states are determined by received SNR of mobile users and basestation, which can be measured by mobile users in downlink and by base-station in uplink. In downlink case, the estimation may be done by end users and the estimated SNR will be feedback to base-station. The feedback may be done via different methods. It can be sent to base-station by piggybacked with acknowledge packets [63]. Alternatively, end users may maintain a small signal channel to base-station. While in uplink case, the estimation of SNR will be done at base-station. Based on SNR, the available transmission rate $\{R_i\}$ for each user can be estimated since transmission rate can be described as a function of SNR. The relationship between transmission rate and SNR may have various definitions, such as linear or the definition in HDR [18].

In our previous theoretical analysis, we assume the feedback of SNR has no loss and no delay. The loss of channel condition information will hardly happen by using error control scheme. The delay is inevitable in downlink case since it needs time to transmit the information from end users to base-station [34]. Fortunately, it is not a big problem if the channel states do not change too

rapidly.

In practical wireless networks, the tradeoff between fairness and throughput becomes more complicated due to the bursty and correlated channel errors. When a mobile user experiences “long” time bad channel states, it is obvious that its transmission should be terminated for the sake of throughput. In such a case, ROS should allow the temporary fairness violation. We can define a threshold for consecutive channel errors. When the number of consecutive channel errors experienced by a mobile user exceeds this threshold, the transmission of this user will be terminated until its channel state recovers.

In conclusion, there are three steps in the procedure of ROS: estimation of $\{R_i\}$, priority computation and update of parameter $\{x_i\}$. At the beginning of each time slot, $\{R_i\}$ is calculated according to estimated SNR of each user. Combining $\{R_i\}$ with $\{x_i\}$, ROS computes the priorities of all users and chooses the user with highest priority. After the transmission in this time slot, $\{x_i\}$ are updated.

5.3.5 Computational Complexity

In this subsection, we analyze the computational complexity of ROS. In ROS, the priority of flows is computed using Equation (5.16), where $\vec{x}(k+1) = \{x_i(k+1)\}$ is updated as

$$x_i(k+1) = \max\{y_i(k+1), i = 1, \dots, N\} - y_i(k+1) \quad (5.17)$$

Suppose user j transmits data in time-slot k , we have

$$\max\{y_i(k+1)\} = \begin{cases} \max\{y_i(k)\}, & \text{if } \max\{y_i(k)\} \geq y_j(k) + R/\phi_j \\ y_j(k+1), & \text{otherwise} \end{cases} \quad (5.18)$$

and

$$y_i(k+1) = \begin{cases} y_i(k), & \forall i \neq j \\ y_j(k) + R/\phi_j, & i = j \end{cases} \quad (5.19)$$

As shown in Equation (5.19), only $y_j(k+1)$ increases since only user j can send data in time-slot k . At the end of time-slot k , we have $y_j(k+1) = y_i(k) + R/\phi_j$. If $y_j(k+1)$ is greater than

$\max\{y_i(k)\}$, $\max\{y_i(k+1)\}$ will be equal to $y_j(k+1)$ since other users do not transmit data. Otherwise $\max\{y_i(k+1)\}$ remains the same as $\max\{y_i^k\}$, as shown in Equation (5.18). Therefore, the computation of $\vec{x}(k+1)$ needs $(N-1)P\{\max\{y_i(k)\} < y_j(k) + R/\phi_j\} + 1P\{\max\{y_i(k)\} \geq y_j(k) + R/\phi_j\}$ times of “add” computations. As a comparison, the scheme proposed in [55] needs $2N$ times of “add” and N times of “multiply” computations. Obviously our scheme has significantly lower complexity in computation of parameters.

5.4 Optimality and Delay Bound of ROS

In this section, we analyze the throughput performance of ROS and prove that ROS is a “throughput optimal” GR node. Furthermore, we will derive a delay bound for a system concatenated by a series of ROS nodes.

5.4.1 Proof of Optimality

In ROS scheme S^* defined in Equation (5.7), the parameter vector $\vec{x}$ is computed to guarantee the fairness among users and the scheduler to be a GR node. With continuous backlog, GR node with a scheduling scheme S provides guaranteed service rate for each user. For example, for the scheduler which randomly chooses a flow to transmit according to the predefined possibility of $\phi_i / \sum_j \phi_j$, we have $P\{S(\vec{G}) = i\} = \phi_i / \sum_j \phi_j$. Next, we will show the optimality of S^* . In other words, we will prove that $E(G(S(\vec{G}))) \leq E(G(S^*(\vec{G})))$ for any S satisfies $P\{S(\vec{G}) = i\} = \phi_i / \sum_j \phi_j$.

Suppose S is a scheduling scheme that satisfies $P\{S(\vec{G}) = i\} = \phi_i / \sum_j \phi_j$, we have

$$\begin{aligned} E(G(S(\vec{G}))) &= E(G(S(\vec{G}))) + \sum_{i=1}^N x_i (P\{S(\vec{G}) = i\} - \frac{\phi_i}{\sum_j \phi_j}) \\ &= E(\sum_{i=1}^N G_i \delta_{\{S(\vec{G})=i\}}) + \sum_{i=1}^N x_i (P\{S(\vec{G}) = i\} - \frac{\phi_i}{\sum_j \phi_j}) \\ &= E(\sum_{i=1}^N (G_i + x_i) \delta_{\{S(\vec{G})=i\}}) - \sum_{i=1}^N x_i \frac{\phi_i}{\sum_j \phi_j} \end{aligned}$$

where $\delta_{\{S(\vec{G})=i\}}$ is defined as

$$\delta_{\{S(\vec{G})=i\}} = \begin{cases} 1 & ; \text{ if } S(\vec{G}) = i \\ 0 & ; \text{ otherwise} \end{cases}$$

According to Equation (5.7), we have

$$E\left(\sum_{i=1}^N (G_i + x_i) \delta_{\{S(\vec{G})=i\}}\right) \leq E\left(\sum_{i=1}^N (G_i + x_i) \delta_{\{S^*(\vec{G})=i\}}\right) \quad (5.20)$$

Hence,

$$\begin{aligned} E(G(S(\vec{G}))) &\leq E\left(\sum_{i=1}^N (G_i + x_i) \delta_{\{S^*(\vec{G})=i\}}\right) - \sum_{i=1}^N x_i \frac{\phi_i}{\sum_j \phi_j} \\ &= E(G(S^*(\vec{G}))) + \sum_{i=1}^N x_i (P\{S^*(\vec{G}) = i\} - \frac{\phi_i}{\sum_j \phi_j}) \\ &= E(G(S^*(\vec{G}))) \end{aligned}$$

This complete the proof.

5.4.2 Delay Bound

In service curve model, packet arrivals from a real-time flow are constrained by arrives, which are described by general sub additive functions, each specifying the maximum amount of traffic that can arrive within any given time interval [22]. For example, the $\sigma - \rho$ model is most common to be used to describe the worst case traffic from a flow by a burst parameter σ and a long term parameter ρ . This model can be realized by a leaky bucket (σ_i, ρ_i) , where σ_i and ρ_i are the depth and rate of leaky bucket respectively.

Suppose the system is a concatenation of M nodes, and flow i is regulated by leaky bucket (σ_i, ρ_i) . Since ROS scheme S^* can provide guaranteed service rate for each flow (see our analysis in subsection 5.3.3), the service curve can be described as $\beta_{r_i, v_i} = r_i t - r_i v_i$, where $v_i = \frac{F_i}{r_i}$. The service curve provided by an ROS node is given by

$$\beta(t) = s_m t - \sum F_i = s_m t - D_{max} N$$

where s_m is the bandwidth of output link of node m . According to the delay bound analysis for concatenation of GR nodes in [22], the delay bound of ROS is

$$d_{bound} = \sum_{m=1}^M \frac{D_{max}N}{s_m} + L_{max} \sum_{m=1}^{M-1} \frac{1}{s_m} + \frac{\sum_i \sigma_i}{\min_m s_m} \quad (5.21)$$

where L_{max} is the maximum packet length.

5.4.3 Properties of $D(k)$

Let $d = \{D(k), k = 1, 2, \dots\}$ be a random variable, representing the maximum difference of “relative throughput” among flows. The system has only two states, $d < R$ and $d \geq R$, denoted by U and V respectively. Let p_1 and p_2 be the transit probability from state U to state V and from state V to state U respectively, then the transit probability from state U to U will be $1 - p_1$ and the transit probability from state V to V will be $1 - p_2$. We have

$$\begin{cases} p_1 = p\{y_1 > y_2\}p\{r_1 = R, r_2 = 0\} + p\{y_1 \leq y_2\}p\{r_1 = 0, r_2 = R\} \\ p_2 = p\{y_1 > y_2\}p\{r_2 = R\} + p\{y_1 \leq y_2\}p\{r_1 = R\} \end{cases} \quad (5.22)$$

The stationary probability distribution satisfies the balance equation,

$$P\{d < R\} * p_1 = P\{d \geq R\} * p_2 \quad (5.23)$$

We also have

$$\frac{p_1}{p_2} = \frac{p\{y_1 > y_2\}p\{r_1 = R, r_2 = 0\} + p\{y_1 \leq y_2\}p\{r_1 = 0, r_2 = R\}}{p\{y_1 > y_2\}p\{r_2 = R\} + p\{y_1 \leq y_2\}p\{r_1 = R\}} \quad (5.24)$$

In this chapter, we make the assumption that all flows’ channel states follow identical independent distribution. We denote by p_e the probability that a flow perceives bad state. We have

$$\begin{aligned} p\{r_1 = R, r_2 = 0\} &= p\{r_1 = 0, r_2 = R\} = p_e(1 - p_e) \\ p\{r_1 = R\} &= p\{r_2 = R\} = 1 - p_e \end{aligned}$$

Then we have

$$\begin{cases} P\{d < R\} = \frac{1}{1+p_e} \\ P\{d \geq R\} = \frac{p_e}{1+p_e} \end{cases} \quad (5.25)$$

These results are useful for computing the packet loss probability in the following subsection.

5.4.4 Packet Loss Probability in ROS

With the knowledge of channel states, the ROS scheduler tries to avoid transmitting data during bad state. As a result, the packet loss is reduced. We will compute packet loss when using ROS scheduler in following.

Again we first consider two users case. In this situation, the flow with bad channel state will be chosen in time-slot k if and only if $D(k) > R$. Thus the packet loss probability in ROS, denoted by P_l is given by

$$P_l = p_e * P\{D(k) \geq R\} = \frac{p_e^2}{1 + p_e} \quad (5.26)$$

The derivation of the packet loss problem for the multiple users case is much more complicated and the derivation is mathematically intractable. However, with the aid of channel states information, we can draw the similar conclusion as that in two users case: the packet loss probability will be much smaller than p_e . The simulation results in the next section will validate this result.

5.5 Behaviors and Performance of ROS

We have proved the optimality of ROS and it can provide rate guarantee with mathematical analysis. In this section, we employ simulation to validate the analysis and to demonstrate the performance of the proposed ROS scheme in terms of fairness, delay bound, throughput and packet loss. In the simulations, we suppose there is a center cell which are surrounded by six cells placed hexagonally, each cell has a base-station. There are some users moving in the cell. The movements of all users are independent with each other. The SNR(Signal Noise Ratio) of each user can be determined by its position, movement direction and speed, according to the path-loss model(Lee's model) and the slow log-normal shadowing model in [81]. The transmission rate of a user is determined by SNR value as shown in Figure 5.1. When SNR is greater than the threshold T_h , the transmission rate

will be R , otherwise the transmission rate is zero. Since there are only two possible transmission rates, we can use two-state Markov model to describe the channel states.

Figure 5.1: Transmission rate vs SNR

5.5.1 Fairness

In the first simulation experiment, we examine if ROS can provide fairness guarantee to all users. $D(k)$, which denotes the maximum difference of “relative throughput” among users, is one of the key parameters used in ROS. The value of $D(k)$ shows the degree of “unfairness” of service received by all users. The greater the $D(k)$ value, the more unfairness among users. In previous theoretical analysis, we have proven that $D(k)$ has an upper bound, i.e., the fairness among users is guaranteed in ROS. Here we validate this result via simulation. In this simulation experiment, there are four video flows carrying Variable-Bit-Rate (VBR) video traffic. Each flow is assigned a different rate ratios as shown in Table 5.1. We use three-state model to describe the VBR video traffic with parameters in Table 5.2. In “on” time, the video source generates packets at different rates in different states, while in “off” time it does not generate any packet. The average “on” and “off” time follow exponential distribution. The probabilities that video source stays in one state are also defined in Table 5.2. The length of time-slot is 2ms, the packet size is fixed to 256 bytes. Therefore, the number of packets can transmit in a time-slot at the bottle-neck link will be

$$R = \frac{3 \times 10^6 \times 2 \times 10^{-3}}{256 \times 8} = 3$$

According our analysis in section 5.3, Equation (5.13) becomes

$$D(k) < \max\{3 + 3/\phi_i, i = 1, 2, \dots, N\} \quad k = 1, 2, \dots \tag{5.27}$$

Figure 5.2 depicts the value of $D(k)$ from time 0 to 60 second. It shows that the variance of $D(k)$ follows the Equation (5.27). In fact, the maximum value of $D(k)$ equals to 3, which is the value of R . According to our analysis in Equation (5.25), the probability of $D(k) \geq R$ is significantly smaller than the probability of $D(k) < R$. Obviously, the case of $D(k) > R$ even does not occur in this experiment. Therefore, this figure demonstrates that there exists an upper bound for $D(k)$, i.e., the unfairness among received service of all users is limited by an upper bound. This result validates our previous theoretical analysis.

Table 5.1: Rate ratio of users

User	Rate ratio(ϕ)
1	1.0
2	1.2
3	1.4
4	1.6

Table 5.2: Video flow parameters

Parameters	Value
State1 transmission rate	$500 \times \phi$ Kb/s
State2 transmission rate	$400 \times \phi$ Kb/s
State3 transmission rate	$200 \times \phi$ Kb/s
State1 probability	0.5
State2 probability	0.3
State3 probability	0.2
Average On-time	1 second
Average Off-time	0.1 second
Packet size	256 bytes

Figure 5.2: Variance of $D(k)$

5.5.2 Delay Bound

In the second simulation experiment, we examine the end-to-end packet delay in ROS. The delay bound of the proposed ROS scheduler is calculated as Equation (5.21). In simulation, we use burst traffic regulated by leaky bucket (σ, ρ) , which generates σ packets at start time and generates packets in constant rate ρ after start time. We assume fixed packet size equals to 2Kbits, thus $L_j^{max} = 2Kb$. Four flows transmit from time zero, with σ equals to 10 packets and ρ equals to 500kbps. We also have $R = 3$ as the same as in previous experiment. Then we choose $D_{max} = 6$ since $\phi_i = 1$ for four flows. There are four hops(links) between source and destination nodes, each of them has the transmission delay equal to 5ms, and thus the network transmission delay from source to destination is 20ms. The bottle-neck bandwidth is 3Mbps and bandwidth of other links are 10Mbps. Then desired end-to-end delay bound is

$$d_i^e \leq 20 + 3 \times \frac{6 \times 4 \times 2K}{10M} + \frac{6 \times 4 \times 2K}{3M} + 3 \times 2K/10M + 4 \times 10 \times 2K/3M = 77.67ms \quad (5.28)$$

Figure 5.3 illustrate the maximum packet delays for four flows. We can see that the maximum end-to-end packet delay for all flows do not exceed the desired delay bound. The delay bound derived from Equation (5.28) is loose. One of the reasons is that we choose $D_{max} = R + R/\phi_i = 6$ in the computation. However, generally D_{max} will be smaller than R as we discuss in Section 5.4.3.

Figure 5.3: Delay bound and the maximum end-to-end delay of flows

If we choose $D_{max} = R = 3$, the delay bound will be $62.47ms$, which is much closer to the maximum packet delay of flows shown in Figure 5.3.

5.5.3 Optimality of Throughput

In the third simulation experiment, we examine the throughput optimality of ROS. It is impossible to compare the throughput of ROS to all other GR nodes. In this chapter, we compare the throughput performance of ROS with Weighted Round Robin (WRR), which is also a GR node since the rate-fraction of each backlogged flow is fixed. There are four users receiving video flows, each of them is assigned a rate ratio - ϕ as shown in Table 5.1. We also employ the same three-state video traffic model as in the first simulation experiment. The parameters of video model are listed in Table 5.2. Figure 5.4 shows the average throughput, in terms of the number of packets, for ROS and WRR respectively.

The x axis in Figure 5.4 indicates the index of flows. The first bar represents the average throughput of ROS scheme and the second bar shows the result of WRR. Obviously, the results in Figure 5.4 demonstrate that ROS achieves higher throughput compared with WRR for all four flows.

Figure 5.4: Throughput of four flows in ROS and WRR

5.5.4 Packet Loss Rate

In the last simulation experiment, we examine the packet losses of users in ROS. Packet loss is also an important performance metric of a scheduling scheme. Similar to above subsection, we also compare our scheme with WRR scheme. The simulation scenario and traffic parameters are the same as previous experiment. The loss rate of each flow is set to 1%. As illustrated in Figure 5.5, obviously ROS has much less packet loss than WRR for all flows, since the latter does not consider channel states. The x-axis of the figure is also the index of users.

5.6 Conclusion

In this chapter, we have presented an efficient Rate-guaranteed Opportunistic Scheduling scheme (ROS) for wireless networks. By measuring the “relative throughput” difference among users, ROS can provide rate-guaranteed service, which is the same as the definition of GR node in service curve with loss model, to mobile users. ROS chooses the user with maximum “opportunity” to transmit in the next time-slot. We prove that ROS can achieve maximum throughput compared with other GR nodes via theoretical analysis and validate the results via simulation. As a GR node, ROS can also provide delay bound for regulated incoming flows. Besides all these advantages, ROS has another important merit - low implementation complexity.

Figure 5.5: Packet loss rate in ROS and WRR

Chapter 6

A New Joint Framework of Packet Scheduling/Admission Control for Multi-service Wireless Networks

Quality of service(QoS) provision is an important and indispensable function for multi-service wireless networks. In order to provide better QoS performance, the packet scheduling scheme should work coordinately with admission control mechanism, another key component of traffic control mechanisms. In this chapter, we present a new scheduling/admission control framework, including an efficient Rate-guaranteed Opportunistic Scheduling (ROS) scheme (introduced in previous chapter) and a coordinated admission control (ROS-CAC) policy to support statistic QoS guarantee in multi-service wireless networks. Based on our proposed mathematical model, we derive the probability distribution function (PDF) of queue length under ROS and deduce the packet loss rate (PLR) for individual flows. The new admission control policy makes admission decision for a new incoming flow to ensure that the PLR requirements of all flows(including the new flow) are satisfied. The numerical results demonstrate the effectiveness of the new joint packet scheduling/admission control framework.

6.1 Introduction

The traffic control mechanisms designed for wireline networks, such as packet scheduling scheme and admission control policy, are not directly applicable to wireless networks since their design has not taken into account some unique problems in wireless networks, such as varying wireless link capacities, location-dependent and bursty bit errors, joint scheduling of uplink and downlink

flows, power constraint in wireless cells, etc. Specifically, three characteristics in wireless signal propagation determine the quality of the wireless channel: path loss variation, slow log-normal shadowing and fast multipath fading [55]. As a result, packet scheduling in wireless networks should also be link state dependent. All these make it a challenging issue to design a good traffic control scheme in wireless networks.

QoS provisioning for on-line traffic such as a conversational call or a video conferencing call is an indispensable function of multi-service wireless networks. These on-line traffic flows have multiple QoS requirements, such as packet delay bound and packet loss rate (PLR), etc. These QoS requirements are too multifold to be satisfied by a single scheme. A traffic control framework, including various mechanisms such as packet scheduling and admission control, is necessary for meeting multiple QoS requirements. In this chapter, we focus on a joint packet scheduling/admission control framework for providing statistic QoS guarantees in multi-service wireless networks. We first review a novel packet scheduling scheme we proposed for wireless networks, called Rate-guaranteed Opportunistic Scheduling(ROS) in Chapter 5 [58]. ROS can guarantee statistical delay bound for flows with arrival curve constraint. Furthermore, ROS can achieve optimal throughput and link utilization while providing fairness guarantee between flows. Both channel states and fairness are considered in the computation of priorities of flows, based on which flows are scheduled. The fairness degree of each flow is represented by the differences between “relative throughput” of each flow and the maximum “relative throughput” value among all flows. The proof of optimality of ROS and the study on its behaviors can be found in Chapter 5 [58]. In this chapter, we focus on a new admission control mechanism which works together with ROS to provide statistical QoS guarantees for multi-service wireless networks.

Admission control is another essential building block of the traffic control framework for QoS provisioning besides packet scheduling. The main function of admission control scheme is to decide the admission of a new incoming flow by judging whether the QoS of existing flows is violated and QoS of the new incoming flow can be satisfied or not. Recently, a lot of admission control schemes have been proposed for wireless networks [30] [43] [66]. These schemes deduce different admission regions as they use different admission criteria. Typical admission criteria for wireless networks include the signal interference ratio (SIR) [30], throughput [43] and packet loss rate [66] etc.

Packet loss rate and delays are two important QoS parameters for real-time communication services. However, voice and/or video users can usually tolerate the loss of a very small fraction of packets. Thus, it is possible to make a more efficient utilization of network resources by providing statistical QoS guarantees. In other words, the traffic control framework operates the network in such a way that congestion phenomena, which leads to packet losses or delays, only occurs with predefined small probabilities. The packet losses in modern wireless networks mainly consist of two parts: packet losses during transmission and packet losses due to buffer overflow in case of congestion at the MAC (medium access control) layer. In modern high-speed wireless networks, the packet losses due to erroneous transmission are well controlled by some mechanisms such as error control schemes and loss recovery and channel states prediction, and occur with a smaller probability than that due to congestion [66].

Traditionally, admission control policy for wireless networks is designed based on the criteria of SNR [30] or throughput [43]. A joint scheduling/admission control framework is proposed in [30] to optimize resource management for uplink in Wideband Code-Division Multiple-Access (WCDMA). New variable bit-rate (VBR) user is admitted only if there is spare capacity after the minimum carrier to total interference ratio (C/I) requirement of the new user is satisfied. A traffic aided smooth admission control (SAC) for opportunistic communication systems, which can provide throughput guarantee, has been proposed in [43]. SAC spreads the decision over a trial period, during which the amount of time resources allocated to new incoming users increase gradually. The throughput in this trial period is measured and used to make admission decisions.

There are also some proposed admission control policies using packet loss rate (PLR) as design criteria. An edge-to-edge measurement based admission control [66] is proposed to maintain PLRs for voice flows of both directions. Lack of mathematical modeling is its shortcoming. In a new data traffic control scheme proposed for integrated service Code-Division Multiple-Access (CDMA) systems [54], joint admission/congestion control is used to maintain the packet error rate (PER) of real-time traffic while non-real-time traffic can use the residual link resources. The admission control policy only admits the data call which will not experience higher delay than the required QoS constraint. The congestion control scheme ensures that the PER requirement of real-time voice traffic is satisfied. Markovian model is used to evaluate PER, blocking probability, throughput,

and delay.

To provide statistical QoS guarantees, we propose a new efficient admission control policy, which uses PLR as the admission criteria to work in conjunction with ROS in multi-service wireless networks. PLR is an important QoS parameter for real-time and some non-real-time data traffic. Using PLR as a criteria for our admission control policy design provides us an effective and efficient approach for QoS provisioning. By investigating the behaviors of packet queues in ROS, we deduce the mathematical expression of probability distribution of queue length. After simplification and approximation, we can obtain probability distribution function (PDF) results similar to the $M/D/c$ model. Based on the properties of PDF, we design a new admission control policy called ROS-CAC specially for the ROS scheduler. The simulation results based on ns-2 [7] demonstrate the effectiveness of our new admission control policy.

The remainder of this chapter is organized as follows. In Section 6.2, we introduce the system model. In Section 6.3, we provide a review for Rate-guaranteed Opportunistic Scheduling (ROS). After that, we analyze the PLR performance of ROS and present our new admission control policy in Section 6.4. We examine the behaviors and performance of our proposed admission control policy via simulation in Section 6.5 and conclude the chapter in Section 6.6.

6.2 System Model

We consider a time-slotted system, in which Time-Division Multiplexing (TDM) is deployed, consisting of one base station and many mobile hosts which may generate multi-class traffic. At the base station, an efficient rate-guaranteed opportunistic scheduling (ROS) scheme (will be elaborated in Section 6.3) is used as the packet scheduling scheme to allocate bandwidth to mobile hosts. In this system, transmission time is fully slotted. The transmission of packets only starts in the beginning of a time-slot. The scheduler assigns a time-slot to individual users. The users may have different transmission rates, such as that in a hybrid Time-Division Multiple Access (TDMA)/CDMA system [60]. The multiple new requests arrived during the time-slot $n - 1$ will be accumulated by the base station. These requests will be processed by the admission control procedure at the beginning of time-slot n . The sets of existing users and new users are denoted by O and F , respectively. $O = O^v \cup O^d$, where O^v and O^d represent the sets of existing voice users and data users,

respectively. In the following, we describe the channel capacity model and the traffic model.

6.2.1 Channel Capacity

In wireline networks, the link has fixed capacity, i.e. capacity $C = \text{Constant}$. Although in wireless networks the link capacity varies with multiple aspects such as fading, the long-term average capacity ($\bar{C}$) can be viewed as a constant value. Such model is accurate in another regime where the changes of channel states are “covered” by physical layer coding and modulation. Suppose r^v and r^d represent the maximum transmission rate of voice users and data users, respectively. F_A is the set of newly admitted incoming users. ϕ_i is the pre-assigned queue weight for user i . After the new users are admitted, the allocated bandwidth for user i will be [58],

$$r_i = \frac{\phi_i}{\sum_{j \in O^v} \phi_j + \sum_{k \in O^d} \phi_k + \sum_{l \in F_A} \phi_l} \bar{C} \quad (6.1)$$

If $r_i \geq r^v$ for voice users and $r_i \geq r^d$ for data users, the bandwidth requirement is satisfied and no backlog is generated. However, the link capacity is partially wasted and under-utilized in such case. Suppose this link capacity is greater than the transmission requirement of M^v voice users and M^d data users at the same time, i.e. $C \geq M^v r^v + M^d r^d$. Since the probability of all users transmitting at the same time is very small, this system may accommodate more than $M^v + M^d$ users, if temporary congestion is allowed. Since there will exist such a case that $r_i < r^v$ or $r_i < r^d$, the backlog may increase under such condition. Assume each user is allocated buffer with fixed size b_i (for simplicity $b_i = b \ \forall i$). The buffer will overflow if too many users are admitted to use the wireless link.

6.2.2 Traffic Model

On-off traffic model is widely used to describe the burst characteristics of voice and multimedia traffic [60] [54]. In this chapter, we use an on-off model with two states- $G(\text{On})$ and $B(\text{Off})$. The transition probability from G state to B state is b , while the transition probability from B to G is g . The probabilities that the model remains at G and B states are $1 - b$ and $1 - g$, respectively. All packets are assumed to have the same packet length.

In the system considered in this chapter, we assume the transmission state keeps unchanged

during a time slot. The state transition only occurs at the beginning of a time slot. Both the “on” period and the “off” period are assumed to be geometrically distributed, and the average value of these two periods are represented by L_{on} and L_{off} respectively. Since the “on” state transits to “off” state with probability b , and remains at “on” state with probability $1 - b$. Then the probability of a burst contains k time slots is given by:

$$P(burst = k) = b(1 - b)^{k-1} \quad (6.2)$$

Thus we have $L_{on} = 1/b$. Supposing the “off” state remains at current state with probability $1 - g$ and transits to another state with probability g , we have the probability of a “off” period containing k time slots as follows,

$$P(period = k) = g(1 - g)^{k-1} \quad (6.3)$$

Similarly, we have $L_{off} = \frac{1}{g}$. Thus the traffic load of this burst traffic flow is given by:

$$\rho = \frac{L_{on}}{L_{on} + L_{off}} = \frac{g}{g + b - bg} \quad (6.4)$$

6.2.3 Channel Model

In this chapter, the two-state channel model [89] is used to describe wireless channel states. In the “good” state, the number of packets can be transmitted per time-slot is R . Otherwise, no packet can be transmitted in the “bad” state.

6.3 Rate Guaranteed Scheduling (ROS)

In this section, we provide an overview of an efficient rate guaranteed opportunistic scheduling (ROS) scheme for wireless networks [58]. The design goal of ROS is to achieve maximum system throughput while satisfying the fairness constraint of resource allocation, with a low implementation complexity. Due to the time varying nature of the wireless channel, the maximum transmission rates of each user also change over time. Naturally dispatching the user with maximum transmission rate at all times will maximize system throughput. However, the tradeoff between

individual user satisfaction and system throughput is necessary. The users with a long distance to the base-station or high disturbance from other signal sources will be sacrificed with small transmission opportunity. While high transmission power and transmission opportunity is allocated to those users near the base-station.

Our scheme is motivated by the work of [55], in which the authors present a solution to maximize system performance value by exploiting channel states while satisfying time-fraction assignments. However, this scheme has a high computational complexity in evaluating parameters. The authors also do not provide analysis of performance bound in the chapter. Besides work in [55], there are some studies [70] [60] extending scheduling algorithms for wireline networks to wireless networks. They provide short-term or long-term throughput bound for incoming flows. At the same time, they consider the fairness constraint between flows and are able to guarantee some short-term or long-term fairness. In order to satisfy the “fairness” constraint, we employ the measure “relative throughput”, which is defined as the throughput divided by the predefined weight. In more details, the difference of “relative throughput” between users is used as a key parameter in the operation of ROS. The greater this parameter, the more the unfairness service received by users.

ROS scheme chooses the flow with highest priority to transmit at the beginning of each time slot. The priority of flow i is defined as

$$prio_i = R_i + x_i \quad (6.5)$$

where R_i is the maximum transmission rate of flow i at the next time-slot, and

$$x_i = \max_j \{y_j(t)\} - y_i, \quad i, j = 1, 2, \dots, N$$

where N is the number of flows serviced by this node. $y_i(t)$ is the “relative throughput” which is defined as

$$y_i(t) = O_i(t)/\phi_i \quad (6.6)$$

where $O_i(t)$ represents the amount of data of flow i which has been serviced, ϕ_i is the normalized rate-fraction of flow i , i.e., $\phi_i = w_i/w_0$, $w_0 = \min\{w_i\}$, where w_i is the predefined queue weight for flow i .

Suppose the system is a concatenation of M nodes, and flow i is regulated by leaky bucket (σ_i, ρ_i) at the ingress node. The ROS scheme can provide guaranteed service rate r_i for each flow. The service curve can be described as $\beta_{r_i, v_i} = r_i t - r_i v_i$, where $v_i = \frac{F_i}{r_i}$ and $F_i = N D_{max} \phi_i / \sum_j \phi_j$. D_{max} is the upper bound of x_i . The detailed deduction of the upper bound is given in [58]. The service curve provided by an ROS node is given by

$$\beta(t) = s_m t - \sum F_i = s_m t - D_{max} N$$

where s_m is the bandwidth of output link of node m . According to the delay bound analysis for concatenation of guarantee rate (GR) nodes [22], the delay bound of ROS is

$$d_{bound} = \sum_{m=1}^M \frac{D_{max} N}{s_m} + L_{max} \sum_{m=1}^{M-1} \frac{1}{s_m} + \frac{\sum_i \sigma_i}{\min_m s_m} \quad (6.7)$$

where L_{max} is the maximum packet length.

By measuring the “relative throughput” difference among users, ROS can provide the mobile hosts rate-guaranteed service, which is similar to the GR node with loss in the model of service curve with loss. ROS chooses the user with maximum “opportunity” to transmit in one time-slot. We prove that ROS can achieve maximum throughput compared with other GR nodes via theoretical analysis and validate the results via simulation. As a GR node with loss, ROS can also provide delay bound for regulated incoming flows. Besides all these advantages, ROS has another important merit - low implementation complexity.

However, without the aid of admission control policy, the number of flows transmitting in the network could be out of control. When there are too many flows sending packets at the same time, there will be a disaster for network and the QoS requirements of all flows may be violated. Therefore, a coordinated admission control policy is necessary for ROS to provide statistical QoS guarantee. In next section, we will introduce our new admission control policy.

6.4 New Admission Control Policy— ROS-CAC

In this section, we first derive the probability distribution function of the queue length of individual flows at base station and estimate the PLRs of all flows. PLR is then used as the main admission criteria for the following considerations. PLR itself is an important QoS parameter for realtime multimedia traffic. In addition, PLR is closely related with other QoS parameters. When congestion occurs in a network, a longer packet delay leads to more packet losses. There is a tradeoff space between delay requirement and loss probability of a flow, both of which are affected by the buffer size allocated to the flow. When a new flow is admitted, a certain amount of buffer is allocated to it. The packets belonging to this flow are queued in the buffer and the HOL (Head of Line) packet will be transmitted if the flow is chosen for service by the ROS scheduler. The smaller the buffer size, the lower the queueing delay at this node because the packets which have been waiting for too long will be discarded. In other words, a smaller buffer size leads to a shorter queueing delay but a higher packet loss probability. The optimal value of buffer size should satisfy both delay and packet loss constraints. Throughput is also related with packet losses. Obviously, more packet losses lead to lower throughput. In this work, we focus on the packet loss due to buffer overflow at the scheduler. Then our admission problem is changed to estimating the probability of buffer overflow p_i^o . The admission policy can be expressed as

$$p_i^o \leq p_i^d \quad i \in O \cup F \quad (6.8)$$

where p_i^d is the PLR requirement for user i , O and F are sets of active and new users, respectively. Before admitting a new user, the base station computes $p_i^o, i \in F$ for new users and re-estimates $p_i^o, i \in O$ for all active users. If all the updated p_i^o satisfy the requirements, this new user is admitted.

6.4.1 Analysis of PLR

In this subsection, we derive the PLR caused by buffer overflow in ROS. At each time-slot only one user is permitted to transmit its data. Since each user has its own queue, we can investigate the behaviors of one of these queues without loss of generality. Let l be the index of user who is

allocated this queue. Let u_i be the probability that i packets are waiting in this queue, we have,

$$\begin{cases} u_0 = A_0 \sum_{i=0}^R u_i Pr\{d=R\} + pA_0u_0 \\ u_1 = A_0u_{R+1}Pr\{d=R\} + A_1 \sum_{i=0}^R u_i Pr\{d=R\} \\ \quad + p(A_0u_1 + A_1u_0) \\ \vdots \\ u_n = A_n \sum_{i=0}^R u_i Pr\{d=R\} + u_0A_np \\ \quad + \sum_{j=R+1}^{R+n} A_{n+R-j}(u_j Pr\{d=R\} + u_{j-R}p) \end{cases} \quad (6.9)$$

where $A_i = Pr\{i \text{ arrivals in a time-slot}\}$. Here we suppose that the packet arrival rate follows Poisson distribution. p is the probability of user who is not scheduled to transmit, and $Pr\{d=R\}$ is the probability that maximum possible number of departure packets (“ d ”) is equal to R , i.e.

$$\begin{aligned} Pr\{d=R\} &= Pr\{\text{user } l \text{ is scheduled to transmit}\} \\ &\times Pr\{\text{user } l \text{ is in “good” state at the same time}\} \end{aligned} \quad (6.10)$$

In Section 5.4.4 we have proved that $Pr\{\text{user } l \text{ is scheduled to transmit}\} = \phi_l / \sum \phi_j$. The probability of channel staying at “bad” state is small as compared at “good” state. Furthermore, we proved that the probability of scheduled user experiencing “bad” state is much more smaller than that of “good” state in Section 5.4.4. Thus the second term of Equation (6.10), $Pr\{\text{user } l \text{ is in “good” state at the same time}\}$ can be ignored, that is $Pr\{\text{user } l \text{ is in “good” state at the same time}\} \approx 1$. We have,

$$Pr\{d=R\} = \frac{\phi_l}{\sum_j \phi_j} = 1 - p \quad \forall l \quad (6.11)$$

Equation (6.9) can be solved by the generating function $P(z) \equiv \sum_{n=0}^{\infty} u_n z^n$. Defining $Q_R =$

$\sum_{i=0}^R u_i$, we have,

$$\begin{aligned}
 P(z) &= [(1-p)Q_R + u_0p] \sum_{n=0}^{\infty} e^{-\rho} \frac{\rho^n}{n!} z^n \\
 &\quad + \sum_{n=1}^{\infty} z^n \sum_{i=1}^n e^{-\rho} \frac{\rho^{n-i}}{(n-i)!} [(1-p)u_{i+R} + u_i p] \\
 &= [(1-p)Q_R + u_0p] e^{\rho(z-1)} \\
 &\quad + \sum_{i=1}^{\infty} [(1-p)u_{i+R} + u_i p] z^i \sum_{n=i}^{\infty} e^{-\rho} \frac{\rho^{n-i}}{(n-i)!} z^{n-i} \\
 &= [(1-p)Q_R + u_0p] e^{\rho(z-1)} \\
 &\quad + e^{\rho(z-1)} \sum_{i=1}^{\infty} [(1-p)u_{i+R} + u_i p] z^i
 \end{aligned} \tag{6.12}$$

implying

$$\begin{aligned}
 P(z) &= \frac{(1-p)Q_R z^{R-(1-p)} \sum_{k=0}^R u_k z^k}{z^R e^{\rho(1-z)} - (1-p+p z^R)} \\
 &= \frac{(1-p) \sum_{k=0}^{R-1} u_k (z^R - z^k)}{z^R e^{\rho(1-z)} - (1-p+p z^R)}
 \end{aligned} \tag{6.13}$$

where ρ is the mean packet arrival rate (number of packets arriving per time-slot).

It can be known that the poles of Equation (6.13) are distinct [40]. According to Rouché's theorem [40], there are $R-1$ zeros of the numerator inside $|z| = 1$, and the R th is clearly $z = 1$. Thus, the numerator can be written as,

$$N(z) = K(z-1)(z-z_1) \dots (z-z_{R-1}) \tag{6.14}$$

By using $P(1) = 1$ and L'Hôpital's rule,

$$\begin{aligned}
 1 &= \lim_{z \rightarrow 1} P(z) = \frac{K(1-z_1) \dots (1-z_{R-1})}{(1-p)R-\rho} \\
 \Rightarrow K &= \frac{(1-p)R-\rho}{(1-z_1) \dots (1-z_{R-1})}
 \end{aligned} \tag{6.15}$$

Then,

$$P(z) = \frac{(1-p)R-\rho}{(1-z_1) \dots (1-z_{R-1})} \frac{(z-1)(z-z_1) \dots (z-z_{R-1})}{z^R e^{\rho(1-z)} - (1-p+p z^R)} \tag{6.16}$$

The roots $\{z_i, i = 1, \dots, R-1\}$ can be derived from $z^R = \frac{1-p}{e^{\rho(1-z)}-p}$. By evaluating Equation (6.16) at $z = 0$, we have,

$$u_0 = \frac{[(1-p)R-\rho](-1)^{R-1} \prod_{i=1}^{R-1} z_i}{(1-p) \prod_{i=1}^{R-1} (1-z_i)} \quad (R \geq 2) \tag{6.17}$$

$\{u_i, i = 1, \dots, R-1\}$ can be derived from a $(R-1) \times (R-1)$ complex-valued linear system of

equations. These equations are created by setting the numerator of Equation (6.13) to be equal to zero at $R - 1$ roots $\{z_i\}$. Then, $\{u_i, i \geq R\}$ can be derived from recursion on Equation (6.9).

Next, we consider the case of a finite buffer. The packet loss probability of finite buffer can be estimated by calculating the sum of state probabilities $\sum_{i=B+1}^{\infty} u_i$ (supposing B is the buffer size) for the infinite buffer case as in [74].

6.4.2 Admission Control Policy

In this subsection, we derive the packet loss probabilities of each active flow and the new admitted flow based on the results in previous subsection, and thus design the admission policy. The traffic load (in terms of mean packet arrival rate) ρ can be derived from the traffic model of each flow. For the on-off traffic model, the equivalent traffic load can be deduced from Equation (6.4). Then for each flow l , u_n^l , the probability of n packets staying in the buffer of flow l can be derived. Furthermore, we have the packet loss rate for flow l .

$$PLR_l(B) = 1 - \sum_{i=0}^B u_i^l \quad (6.18)$$

where B is the buffer size (in number of packets) for flow l .

By using the PLR result from Equation (6.18), we can design an efficient admission control policy by using the PLRs as the criteria. The admission region is defined in Equation (6.8). If the calculated $PLR_l(B)$ is smaller than the PLR requirement of flow l and all existing flows' PLRs are smaller than their requirements, flow l can be admitted. Otherwise, flow l is rejected.

With the increase of number of flows and types of traffic, the computation complexity of queue length PDF and the corresponding PLR will increase accordingly. A near-optimal but simpler algorithm could be developed and this could be an interesting research topic in the future. Another possible solution to reduce the complexity is to make computation "off-line" - before the operation of system, if the parameters of traffic and channel capacity are known. By calculating the results of all possible combinations of traffic flows, the admission region can be determined. The results can be stored in a table, which can be looked up before an admission decision is made.

6.4.3 Example of Admission Control Operation

Here we use a simple example to illustrate the operation of ROS-CAC. We assume that there are N users, including active and newly admitted users, in the system. All users have the same queue weight, i.e. $\phi_i = \phi_j, \forall i, j \in 1, \dots, N$. Thus all users have the same probability to be scheduled to transmit, i.e. $1 - p = 1/N$. We assume that the maximum number of packets can be transmitted in a time slot $R = 3$. All flows follow the on-off traffic model, and the traffic load ρ of each flow is normalized in terms of number of packets arriving during a time-slot.

At first we resolve roots $\{z_1, z_2\}$ (obviously $z_3 = 1$) of the following equation,

$$z^3 = \frac{1 - p}{e^{\rho(1-z)} - p} \quad (6.19)$$

The results will be a pair of complex value. Suppose,

$$z_1 = x + jy; z_2 = x - jy \quad (6.20)$$

Thus, according to Equation (6.17), we have

$$u_0 = \frac{(3(1 - p) - \rho)(x^2 + y^2)}{(1 - p)((1 - x)^2 + y^2)} \quad (6.21)$$

From the 2×2 complex-valued linear system of equations created from the numerator of Equation (6.13), we have

$$\begin{cases} u_1 = u_0 \frac{x^2 + y^2 + 2x}{-x^2 - y^2} \\ u_2 = u_0 \frac{1 + 2x}{x^2 + y^2} \end{cases} \quad (6.22)$$

$\{u_i, i \geq 3\}$ can be derived from recursion of Equation (6.9).

$$\begin{cases} u_3 = \frac{(1 - pA_0)u_0 - A_0 \sum_{i=0}^2 u_i}{A_0(1 - p)} \\ \vdots \\ u_{3+n} = \frac{1}{A_0(p-1)} \{ A_n \sum_{i=0}^3 u_i (1 - p) + u_n (A_0 p - 1) \\ + \sum_{j=4}^{2+n} A_{n+3-j} (u_j (1 - p) + u_{j-3} p) + u_0 A_n p \} \end{cases} \quad (6.23)$$

Next, we show an example of the calculation result of previous procedure. In this example,

$N = 20, R = 3$. By setting traffic load ρ to three different values, the corresponding three PDF curves are shown in Figure 6.1. Obviously, the lower the traffic load ρ , the higher the probability queue length being short.

Figure 6.1: PDF of queue length in ROS($\rho = 0.1, 0.12$ and 0.14 , respectively)

The PDF results can be used in admission control policy. Suppose the PLR requirement of the flow is no greater than 0.01, traffic load of each flow is $\rho_i = 0.02$ and queue length allocated to each flow is 20 packets. From the obtained PDF results, we can find that when total traffic load $\rho = 0.10$, $PLR = 0.00922$ the PLR requirements of all flows are satisfied. However, when $\rho = 0.12$, $PLR = 0.096$ the PLR requirements are violated. Thus the maximum number of users who can be admitted is 5.

6.4.4 Discussions

When a new incoming flow is admitted by the admission control policy, the base station allocates a certain size of buffer to it. The packets of this flow are served on the FCFS (First Come First Serve) basis in this buffer. The HOL packet will be transmitted when the flow obtains transmission opportunity. The buffer size B is an important parameter for QoS provisioning. First, PLR of each flow is determined by B . Second, it can also affect the waiting time of packets. The maximum waiting time is $D_m = \frac{B}{r_i}$ in a simplified case. Thus We have the maximum buffer size not violating delay requirement $B_d = r_i D_i^{max}$, where D_i^{max} is the maximum queueing delay requirement of flow i .

In short, the buffer management is very important. Supposing the minimum buffer size satisfying the PLR requirement is B_p , the actual allocated buffer size should satisfy following conditions.

$$B \geq B_p \quad \text{and} \quad B \leq B_d \quad (6.24)$$

If B_d is larger than B_p , B can be assigned any value between them.

The computation of maximum waiting time $D_i = \frac{B_d}{r_i}$ is valid only in an ideal continuous fluid model. In ROS, the distribution of waiting time is more complicated since only one user can transmit in a time-slot and no simultaneous transmission is allowed. In long-term average, user i is scheduled to transmit with a specified probability. Another particular characteristic of ROS is that packets leave the buffer in a batch during a time-slot. One batch of packets consists of R packets. When a packet enters the buffer of user i , there are already M_n packets waiting in the buffer. The average waiting time (in terms of time slot) of this packet will be,

$$\bar{D}_i = (M_n \bmod R) \times \bar{T}_i + 1 \quad (6.25)$$

where $\bar{T}_i$ is the average waiting time of HOL packet of this buffer. Since the probability of user i being chosen to transmit is $p_i = \phi_i / \sum \phi_j$, $\bar{T}_i$ can be deduced as follows.

$$\bar{T}_i = 1/p_i = \frac{\sum \phi_j}{\phi_i} \quad (6.26)$$

Instead of fixed buffer size, variable buffer size should be more efficient. For example, in the case of B_d larger than B_p , we have $B_p \leq B \leq B_d$. If B is assigned the minimum value B_p , this flow occupies minimum buffer resource. However, some packets may be prematurely dropped before their delay timer expired in such case. Thus packet loss rate increases. On the other hand, if B is assigned the maximum value B_d , the PLR will be smaller but more buffer resources occupied. Thus, there is a tradeoff between PLR and buffer resource. The optimal size is an interesting issue to study.

6.5 Performance Evaluation

In this section, we use experiments to validate the effectiveness of the proposed joint packet scheduling/admission control framework based on ns-2 [7] simulations. In the experiments, we consider a system with a center cell surrounded by six cells placed hexagonally. The performance results of the center cell will be simulated and analyzed. We assume on-off video traffic flows are transmitting in this system. The on-off traffic model generates packets in a constant rate r_{on} during “on” time periods, while not generating during “off” time periods. The lengths of “on” and “off” time periods follow exponential distribution with mean values of t_{on} and t_{off} , respectively. The mean transmission time of each video flow is set to t_{active} . We use the two-state Markov model to describe channel states. In the “good” state, the maximum number of packets can be transmitted in a time-slot will be $R = 3$. Users arrive at the center cell following Poisson distribution. The queue length for each admitted user is fixed at 20 packets. All packets have the same packet length-256K bytes. Suppose the PLR requirement of the flow is 0.01. Other simulation parameters are listed in Table 6.1.

Table 6.1: Simulation parameters

Parameter	Value
t_{on}	180ms
t_{off}	60ms
r_{on}	128kB/s
t_{active}	180s

6.5.1 Video Traffic Scenario

We simulate the new proposed admission control policy (ROS-CAC) based on the traffic model and channel model mentioned previously. We compare the system performance results in two different conditions: without admission control and with admission control.

The comparison of average packet loss ratio with admission control (AC) and without AC is shown in Figure 6.2. When the traffic load is below $1.67/min$, the PLR difference between two cases is invisibly small. The PLR requirements are satisfied under both cases. However, the PLR of the system without AC increases very fast with the traffic load, and the PLR requirement is violated when the user arrival rate is greater than $2.5/min$. As a comparison, the PLR of the system with

AC remains stable and PLR requirement is always guaranteed to be smaller than 0.01. Obviously, the base station can avoid lots of packet losses with the aid of admission control. Moreover, the average end-to-end delay of packets is significantly reduced, which is illustrated in Figure 6.3. This is because that the number of packets waiting in queues with admission control is much less than that without admission control.

We evaluate the system throughput by calculating the average number of packets successfully transmitted per time-slot. The simulation results of system throughput are demonstrated in Figure 6.4. We can see that the system throughput with admission control is slightly lower than that without AC when new user arrival rate is high. The main reason is that with admission control, less users can transmit simultaneously than that without AC.

Figure 6.2: Packet loss rate in video scenario

6.5.2 Mixed Traffic Scenario

Different from previous simulation experiment with only video traffic flows, in this experiment we add two TCP flows in the simulation. Suppose the queue weight of TCP flow $\phi_t = \frac{1}{2}\phi_v$, where ϕ_v is the queue weight of video flow. The traffic model for video flow is the same as that in previous section. We also compare the system performance results between with AC and without AC.

Figure 6.5 shows the comparison of average packet loss ratio with admission control (AC) and without AC in mixed traffic scenario. When the traffic load is below $0.75/min$, the PLR difference

Figure 6.3: Average end-to-end delay in video scenario

Figure 6.4: System throughput in video scenario

between two cases is invisibly small. The PLR requirements are satisfied under both cases. However, the average PLR of the system without AC increases very fast with the traffic load, and the PLR requirement is violated when the user arrival rate is greater than $1.86/min$. As a comparison, the average PLR of the system with AC remains stable and PLR requirement is always guaranteed to be smaller than 0.01. In Figure 6.6, we illustrate the average end-to-end delay of packets. We can find that the delay is significantly reduced due to smaller waiting queue length. The system throughput is still evaluated by calculating the average number of packets successfully transmitted per time-slot. The simulation results of system throughput are shown in Figure 6.7. Similar to the video traffic scenario, we can see that the system throughput with admission control is slightly lower than that without AC when new user arrival rate is high.

Figure 6.5: Packet loss rate in mixed scenario

6.6 Conclusion

In this chapter, we first review a novel packet scheduling scheme we proposed for wireless networks, called Rate-guaranteed Opportunistic Scheduling(ROS). In order to provide statistical QoS guarantees, we have proposed a new admission control policy to work together with ROS for wireless networks. Through mathematical analysis, we investigated the behavior of buffer at base-station and buffer overflow probability. In order to maintain low packet loss rates for mobile users, the base-station allocates buffer of an appropriate size to individual users. Our proposed new admission

Figure 6.6: Average end-to-end delay in mixed scenario

Figure 6.7: System throughput in mixed scenario

control policy admits new user if all active users' PLR will not be violated. The simulation results have validated the effectiveness of our proposal.

Chapter 7

Optimizing Congestion Control for Multi-hop Wireless Networks

After previous work on wireless packet scheduling and admission control, our focus turns to congestion control, the third key component of traffic control mechanisms. Instead of traditional wireless networks (e.g., cellular networks), multi-hop wireless networks become our research interest, since the latter is currently a hot research area. As a kind of multi-hop wireless networks, wireless mesh network (WMN) provides a new network access solution for both mesh and conventional mobile clients. Recently, WMN with multi-channel MAC (Media Access Control) is proposed to provide higher system capacity by exploiting non-overlapped channels with multiple transceivers at WMN nodes. The integration of WMNs with other networks (e.g. Internet, cellular, IEEE 802.11, sensor networks, etc.) requires an effective and robust congestion control algorithm to maintain network stability and achieve a high network resource utilization. In this chapter, we address the problem of congestion control for WMNs with multi-channel MAC. We formulate the design of congestion control mechanism in such a system as a problem of utilization maximization subject to time constraint. We first derive an end-to-end algorithm as the solution. Based on that, we develop a distributed adaptive algorithm with hop-by-hop congestion information feedback and prove its convergence. The numerical results based on ns-2 simulations validate the stability and demonstrate that our new hop-by-hop congestion control algorithm significantly outperforms the end-to-end algorithm in terms of packet delay and queue length.

7.1 Introduction

As a new solution for last-mile broadband Internet access, wireless mesh network (WMN) becomes a hot research topic recently [47,77]. Some major network/telecommunication equipment companies, such as Motorola [6], Nortel [8], Microsoft [4] and Intel [38], have proposed their own designs and implementations for WMN. Compared with other broadband Internet access technologies, e.g. cable, digital subscriber line (xDSL) and satellite Internet access, WMN has some prominent advantages. First, the deployment of WMNs is easy and cost effective. The initial installment of WMNs may be very simple with a few nodes. Second, the topology of WMNs is highly scalable. New nodes can be added in deployment at anytime the operator desires. Other attractive features of WMNs include self-organization, auto-configuration and self-healing, etc. [14]

WMN shares some common characteristics with Ad-Hoc networks, as it could be considered as a special type of Ad-Hoc networks in some sense [14]. However, there are some critical differences between them [75]. First, topology changes in the WMN are not as frequent as that in Ad-Hoc networks, since the positions of backbone nodes in WMN are fixed after they are deployed [14]. Second, in WMN the traffic characteristics of aggregated flows do not change significantly. Third, traffic flow pattern in WMN is usually one-directional: to/from a wired network backbone. This is determined by prominent data applications- most users retrieve data from Internet or enterprise servers which are usually within the wired infrastructure. In addition, WMN has some important unique characteristics in network topology. All nodes in backbone WMN act as both access points (APs) and routers. However, not all access points in WMN have wireline connections with Internet backbone. Some APs have only wireless links. Traffic aggregation devices (ADs) are deployed at some APs with sufficient processing capacity. Once the route between an AP with AD and another AP with AD is fixed, the aggregated traffic flows between these two APs will share common paths. All above mentioned characteristics of WMNs suggest that the designs of architecture, protocols and other system components for Ad-Hoc networks are not directly applicable for WMNs.

In recent years, WMN architecture with multi-channel MAC has been proposed to utilize multiple non-overlapped channels defined in 802.11 standard [75]. Existing multi-channel MACs fall into three categories [14]: mutli-channel single transceiver MAC, multi-channel multi-transceiver

MAC and multi-radio MAC [12]. In multi-channel single transceiver MAC, each node has only one transceiver and operates on one channel at a time, but different nodes may operate on different channels [80]. In multi-channel multi-transceiver MAC, a WMN node has multiple transceivers which may operate on different channels simultaneously but share a common MAC layer. WMN with multi-radio MAC in which each node is equipped with two or more network interface cards (NICs) has also been developed [75]. The system capacity of WMN with multi-channel MAC is significantly larger than that of traditional single channel network. Therefore, we expect that multi-channel MAC will become more prevalent for WMNs in the future.

In order to achieve high end-to-end throughput and maintain system stability in WMNs, congestion control is essential and should be designed carefully. The main function of congestion control is to regulate source transmission rates so that the total transmission rates on any link of the network do not exceed the available link capacity, or the sum of data rates do not exceed the forwarding capacity for every node. Congestion control is important to maintain network stability and avoid network fatal collapse. While this topic has been intensively studied in the Internet context in the past few years, little relevant work is known in the Ad-Hoc network context [92], especially in the WMN context. Current Internet congestion control schemes (e.g. TCP and its variants) can be interpreted as seeking the global optimization of a certain aggregated utilization function subject to network capacity constraints [62]. If routing information and channel capacity constraints are given, end-to-end control schemes (e.g. TCP) can be developed to solve this problem. On the other hand, for multi-hop wireless network, this problem can be solved by approximate distributed algorithms, which can converge to a stable operation point [27].

Compared with end-to-end schemes, a hop-by-hop congestion control algorithm has shorter response time since the congestion feedback is sent to preceding node which can make prompt adjustment accordingly. Hop-by-hop algorithms need per-flow information at all intermediate nodes, causing the scalability problem for large-scale networks such as Internet. However, in WMN the number of flows at each node is of a much smaller order than that in Internet [92]. Moreover, many existing packet scheduling schemes, e.g. [23, 60], use per-flow queueing at intermediate nodes. With flow aggregation schemes, the number of flows transmitting in WMN can be further reduced. Thus we argue that hop-by-hop control is appropriate and feasible for WMNs. Actually, hop-by-hop

algorithms have been proposed in multi-hop wireless networks context in the literature [92].

In this chapter, we focus on the congestion control for WMNs with multi-channel MAC, in which each node has two NICs operating simultaneously and independently. To the best of our knowledge, there is no similar work on congestion control for such a system. Our motivation comes from the constrained maximization problem of system utilization function. We use time constraint instead of link constraint since the former is stricter than the latter one.

Our contributions in this chapter are three fold. First, we formulate the problem of utilization maximization with time constraint for congestion control design in WMN with multi-channel MAC. Second, we solve the maximization problem with convex utilization function, and derive the so-called end-to-end solution. This end-to-end solution can be described as an algorithm to adjust source transmission rate. Third, we develop a hop-by-hop congestion control algorithm, which has better performance in terms of packet delay, based on the end-to-end solution. Instead of controlling source rate directly, the hop-by-hop adaptive algorithm controls the transmission rates of flows at intermediate nodes. We prove that the transmission rates will converge to a stable point which is the same as that of the end-to-end solution. We conduct simulation experiments based on ns-2 and the numerical results validate the effectiveness of our new hop-by-hop algorithm and demonstrate its better performance in terms of packet delay and queue length.

The rest of this chapter is organized as follows. We describe the issues of congestion control in WMN and introduce some related work in Section 7.2. In Section 7.3, we describe the system model, including multi-channel WMN, time constraint and utilization function. The optimization problem is brought out in Section 7.4, followed by a end-to-end source rate control solution. Furthermore, a new distributed hop-by-hop rate control algorithm is proposed. In Section 7.5, we present the numerical results to demonstrate the effectiveness of the proposed hop-by-hop algorithm. Finally, we conclude this chapter in Section 7.6.

7.2 Congestion Control for WMN

Congestion control for WMN is still a new research topic as current research work in the area of WMN mainly focuses on the lower layer issues, including routing, MAC protocol, etc. [75]. Some characteristics of WMN should be considered in the development of congestion control scheme.

First, as a new solution for “last mile” access network, each node in WMNs acts as an access point, which may be interconnected with various subnets with different infrastructures, protocols and transmission environments. Applying the same congestion control mechanism to heterogeneous subnets will be ineffective. On the other hand, it may be costly and complicated to design different congestion control schemes for different subnets. As a result, the new congestion control mechanism for WMNs should be adaptive to diverse network conditions. Moreover, diverse types of traffic flows may be accumulated at the WMNs’ nodes. Naturally, traffic aggregation, which is similar to that in infrastructure mode of IEEE 802.11 standard, is suitable for this system. With traffic aggregation, it is possible to maintain per-flow states at each WMN nodes for congestion control mechanism. Second, the infrastructure of WMN is semi-permanent, which is more flexible than wired network and cellular network but more stable than traditional Ad-Hoc network since the WMN routers have minimum mobility, while WMN clients may also be stationary or mobile hosts [14]. The heterogeneity of integrated networks and the differences in mobility make the existing congestion control algorithms designed for wired networks and Ad-Hoc networks are not directly applicable to WMNs. In the following we review some existing congestion control mechanisms, including aggregated congestion control and congestion control algorithms for conventional wireless networks, which may provide us some hints in the development of congestion control schemes for WMNs.

There are two different ways to implement congestion control— end-to-end or host-by-host algorithm [92]. The former is more favorite in wireline networks and is usually realized by transport protocols(e.g. TCP). The latter has been studied in both the Internet [68] and the multi-hop wireless networks context [92]. Both methods have their own pros and cons [92]. End-to-end schemes (e.g. TCP) dominate present Internet infrastructure for its good scalability and deployability. The congestion control is realized by adjusting the source rate according to the feedback from destination node to the source node. In hop-by-hop methods, a node sends a notification to preceding node immediately when it detects congestion. Thus the congestion reaction is faster than that in end-to-end schemes and results in a better performance in terms of delay of control loop. Per-flow states record and management are required in intermediate nodes along the path in hop-by-hop schemes. Obviously it is hard to implement such scheme in large scale networks with a large number of flows. Fortunately, in wireless networks, the number of flows per node is of a

much smaller order than that in the Internet. Furthermore, existing packet scheduling schemes for wireless networks usually exploit per-flow states and use per-flow queueing [23, 60], enabling the possibility to share the per-flow state with congestion control mechanism. Therefore, hop-by-hop schemes could be more suitable for multi-hop wireless networks.

Not so much work on congestion control for wireless networks is known. A host-by-host congestion control mechanism for wireless sensor networks is presented in [44]. This mechanism consists of two steps. In the first step, each node detects possible congestion. There are two different ways for congestion detection: queue occupancy monitoring and channel sampling. The former method monitors the queue size at the node. If the queue size is greater than a threshold, the sensor is considered in congestion and then the sensor in congestion sets the congestion bit in the header of its outgoing packets. After overhearing the congestion bit from its parent node, a sensor will stop forwarding data to allow its parent to drain its queues. If the hop-by-hop feedback eventually arrives at the source node, the application level flow control mechanism will adapt the source rate. CODA [86] uses another method to detect congestion. Before the sensor sends a packet, it samples the channel states in a fixed time interval. A utilization factor is computed based on the number of times the channel is busy. If the utilization factor is greater than a threshold, the sensor is considered in congestion. Finally, each node sets the congestion bit in the header of every outgoing packet if congestion is detected. Otherwise, the congestion bit will be cleared. In [92], the authors propose a fair hop-by-hop congestion control algorithm with the time constraints at MAC layer for multi-hop wireless networks. The stability of this algorithm is proved by Lyapunov function. It is also proved that the algorithm has the property of spatial spreading, i.e. the congestion at bottleneck node can be “smoothed” over nearby hops.

7.3 System Model

In this chapter, we consider the utilization maximization problem with time constraint for the congestion control design in WMNs with multi-channel MAC. The underlying system model involves three main elements: WMNs with multi-channel MAC, time constraint and utilization function. In this section, we first describe the architecture of WMNs with multi-channel MAC. Next we present the expression of time constraint and give an example for easy understanding. Finally, we present

the utilization function, which is a variant from TCP model.

7.3.1 Multi-channel WMN

In conventional Ad-Hoc and sensor networks, all mobile hosts use a single channel. However, IEEE 802.11b/g and 802.11a standards define 3 to 12 non-overlapped channels. In [75], the authors also propose an architecture of multi-channel wireless mesh networks, by assuming each node has multiple NICs. Due to the interference from neighboring links as well as from adjacent hop in the same path, the capacity of WMN with single channel is highly constrained. On the other hand, exploiting multi-channels to transmit data simultaneously will definitely improve total system throughput significantly in WMNs. Figure 7.1 shows the architecture of a WMN with multi-channel MAC. Only the gateway nodes have the wired link to Internet backbone; while the other nodes have only wireless links. Traffic aggregation devices are deployed with some of WMN nodes. Figure 7.1 also illustrates the channel assignment results of routing/channel assignment scheme. Totally 3 non-overlapped channels are used in this system. One of these channels is assigned to each link, while each node can use two different channels since they have two NICs. With just two NICs on each node, the system throughput can be improved by factor of 6 to 7 compared with single-channel WMN [75]. Furthermore, in [75] the authors propose a distributed routing/channel assignment algorithm for multi-channel WMNs. In this algorithm, all the gateway nodes with wired link to Internet backbone try to build a spanning tree which starts from the node itself and all mobile hosts try to join one of these spanning trees. These spanning trees are built when the topology of WMN is deployed and rebuilt when any gateway node fails to work. The routes between any two wired gateway nodes are fixed since they consist of only wired links. However, the routes from a mobile host to its root gateway may change when traffic load varies. In this chapter, we consider a similar WMN architecture with two NICs at each node. These two NICs can operate simultaneously in time-division mode. That is, an NIC cannot receive or send multiple flows at the same time. Moreover, for simplicity, we assume that there are enough non-overlapped channels to be utilized by channel assignment scheme, and the channel assignment has been implemented like that in [75] [24]. Thus we do not consider routing/channel assignment algorithm but focus on the congestion control in this chapter.

Figure 7.1: Multi-channel WMN with two NICs

7.3.2 Time Constraint

Instead of a loose link capacity constraint, a tighter constraint-time constraint which is also related with link capacity is introduced in [92]. Time constraint is derived from the channel access time constraint of many wireless MACs' implementations, in which time-division strategy is used. In such a strategy, only one user can access the physical channel at any time instant. Although carrier sense multiple access with collision avoidance (CSMA/CA) is the current standard of MAC for WMNs, its low network capacity and lack of scalability suggest that time-division multiple access (TDMA) or code-division multiple access (CDMA) will eventually replace it [14]. The time constraints exist at MAC/PHY (Physical) layer of WMNs since each NIC has only one transceiver. To be more accurate, one NIC cannot perform multiple transmissions and receptions at the same time. This constraint can be expressed in terms of time fraction used by flows, which is defined as r/c , where r is the transmission rate of a flow and c is the capacity of link traversed by this flow. Suppose there are several links connected to node k . Among these links, the sets of links using NIC 1 and NIC 2 of node k are denoted by L_1 and L_2 respectively. The time constraint at node k can

be expressed as following inequation,

$$\begin{aligned} \sum_{l \in L_1} \frac{\sum_{f \in F_l} r_f}{c_l} &\leq 1 \\ \sum_{l \in L_2} \frac{\sum_{f \in F_l} r_f}{c_l} &\leq 1 \end{aligned} \quad (7.1)$$

where c_l is the capacity of link l ; F_l denotes the set of all flows traversing link l ; r_f is the transmission rate of flow f , $f \in F_l$.

Let us use an example shown in Figure 7.2 to illustrate the meaning of this constraint. Node C is connected to four nodes- A, B, D and E. c_1, c_2, c_3 and c_4 are the capacity of links ($l_{ac}, l_{bc}, l_{dc}, l_{ec}$) from node A, B, D and E to C, respectively. There are four flows f_1, f_2, f_3 and f_4 traversing node C with transmission rates r_1, r_2, r_3 and r_4 , respectively. Suppose links l_{ac} and l_{bc} are bound with NIC 1 at node C; while l_{dc} and l_{ec} are bound with NIC 2. According to Equation (7.1), we have the following constraints,

$$\begin{aligned} \frac{r_1+r_4}{c_1} + \frac{r_1+r_2}{c_2} &\leq 1 \\ \frac{r_3+r_4}{c_3} + \frac{r_2+r_3}{c_4} &\leq 1 \end{aligned} \quad (7.2)$$

Figure 7.2: Example of link constraint

Compared with traditional link constraint, time constraint is a stronger constraint. Link constraint states that the total transmission rate of flows along one link is limited by the capacity of

this link. For comparison, the link constraints for previous example can be expressed as

$$\begin{cases} \frac{r_1+r_4}{c_1} \leq 1 \\ \frac{r_1+r_2}{c_2} \leq 1 \\ \frac{r_3+r_4}{c_3} \leq 1 \\ \frac{r_2+r_3}{c_4} \leq 1 \end{cases}$$

Obviously the link constraints are subsumed by the time constraints, i.e., the link constraints must be satisfied once the time constraints are satisfied. In most cases, time constraint is stronger than link constraint except in the extreme case where there is only one flow and this flow ends at the node. Due to the existence of such a stronger time constraint, it is unnecessary to consider link constraints. The term “1” on the right hand side of Equation (7.2) is based on the assumption of 100 percent utilization of link capacity, which is actually not practically implementable. Suppose $\eta, 0 < \eta < 1$ is the efficiency of MAC protocol, the expressions in Equation (7.2) are changed as,

$$\begin{aligned} \frac{r_1+r_4}{c_1} + \frac{r_1+r_2}{c_2} &\leq \eta \\ \frac{r_3+r_4}{c_3} + \frac{r_2+r_3}{c_4} &\leq \eta \end{aligned}$$

Time constraints should be applied to all nodes in the network. Suppose there are $N(\mathbf{V})$ nodes in the network, the time constraints are represented by $2 \times N(\mathbf{V})$ linear inequations.

7.3.3 Utilization Function

Kelly et al. model network congestion control as a distributed solution of utilization maximization problem [50]. Recent research shows that TCP transmission can be modeled as utility maximization problem when the link capacity is fixed [62]. Utilization function, which can be interpreted as profit of transmission, is defined as a function of transmission rates. Different TCP implementations have different definitions of utilization function. For example, TCP vegas uses

$$U(r_i) = \alpha_i d_i \log(r_i)$$

as the utilization function, where α_i , d_i and r_i are the protocol parameter, round trip time (RTT) and transmission rate of flow i , respectively. TCP Reno's utilization function can be expressed as [62]

$$U(r_i) = -\frac{a^2}{\tau_i^2 r_i}$$

where τ_i is the round trip time of flow i and a is a constant. The variants of these functions have also been used in the development of congestion control schemes. In [50] the utilization function is defined as,

$$U(r_i) = w_i \log(r_i)$$

where w_i is the weight of flow i . In this chapter, we choose a variant of the TCP Reno as the objective utilization function in the optimization design of congestion control scheme for WMNs with multi-channel MAC. We define the utilization function for flow i as,

$$U(r_i) = -\frac{W_i}{r_i}$$

where W_i is a predefined constant weight parameter for flow i . The RTT term is ignored in this expression for simplicity. This does not affect the generality of the proposed congestion control algorithm. Under the assumption that the routing and channel assignment have been performed perfectly, the RTT can be considered to be roughly constant for a specific flow, and thus does not affect the optimization problem in the congestion control design. Indeed, it can be combined in the weight factor.

7.4 Proposed Congestion Control Algorithm

In this section, we present our new proposal- rate controller for WMNs. The transmission rate adjustment algorithm at intermediate nodes is the core component of the hop-by-hop congestion control mechanism. The algorithm can be derived from an optimization problem. The optimization objective could be system throughput, link utilization or other utilization function of transmission rates. These utilization functions are constrained by channel capacity.

7.4.1 Problem Formulation

Our design objective is to maximize transmission utilization function while satisfying link constraints. Let $N(\mathbf{V})$, $N(\mathbf{L})$ and $N(\mathbf{F})$ be the total number of nodes, links and transmitting flows in the system, respectively, where $\mathbf{V}$, $\mathbf{L}$ and $\mathbf{F}$ denote the sets of nodes, links and flows in the system, respectively. Suppose $U(\cdot)$ is the utilization function. The optimization problem can be formulated as follows,

$$\text{maximize } U(\vec{r}) \quad (7.3)$$

subject to time constraint,

$$\mathbf{MA}\vec{r} \leq \eta \mathbf{1}$$

over

$$\vec{r} \geq 0$$

where $\vec{r}$ is the vector of transmission rates of all transmitting flows. $\mathbf{A}$ is a $N(\mathbf{L}) \times N(\mathbf{F})$ matrix, which represents the association between flows and links. The definition of elements in matrix $\mathbf{A}$ is listed below.

$$a_{ij} = \begin{cases} \frac{1}{c_i}, & \text{if flow } f_j \text{ traverses link } l_i \\ 0, & \text{otherwise.} \end{cases}$$

$\mathbf{M}$ is an $2 * N(\mathbf{V}) \times N(\mathbf{L})$ matrix representing the association between links and nodes. Suppose the channel capacity of link i is denoted by c_i . The elements of $\mathbf{M}$ are defined as follows,

$$\begin{cases} m_{(2i)j} = \begin{cases} 1, & \text{if link } l_j \text{ is incident with node } n_i, \text{ and use NIC 1} \\ 0, & \text{otherwise.} \end{cases} \\ m_{(2i+1)j} = \begin{cases} 1, & \text{if link } l_j \text{ is incident with node } n_i, \text{ and use NIC 2} \\ 0, & \text{otherwise.} \end{cases} \end{cases}$$

We define the utilization function as follows.

$$U(\vec{r}) = \sum_{f \in \mathbf{F}} \left(-\frac{W_f}{r_f} \right)$$

where W_f and r_f are the weight parameter and rate of flow f respectively. The utilization function is strictly concave due to the assumption that $W_f > 0, f \in \mathbf{F}$.

Our end-to-end congestion control algorithm comes from the solution to the problem represented by Equation (7.3). Based on the end-to-end algorithm, we develop a new hop-by-hop distributed rate control scheme, which has better performance in terms of response time and occupied queue length.

7.4.2 End-to-end Algorithm

In the mathematical model, the source rates are used as the primal variable in the utilization optimization problem. As a result, the rate (congestion) control mechanism, which is derived from the optimization model, adjusts source rate directly according to the feedback from the destination. This is so called “end-to-end” algorithm. Instead of controlling source rate directly, hop-by-hop algorithms adjust the forwarding rates at intermediate nodes along the path to alleviate network congestion. It is a challenging task to prove the convergence of a hop-by-hop algorithm. Our idea is to develop the new hop-by-hop algorithm based on an end-to-end algorithm which is proved to be stable and converging.

The end-to-end congestion control scheme is derived from the solution to the optimization problem in Equation (7.3). First, we derive the Lagrange multipliers expression for the optimization problem,

$$L(\vec{r}, \mu) = \sum_{f \in \mathbf{F}} \left(-\frac{W_f}{r_f} \right) + \mu^T (\mathbf{M} \mathbf{A} \vec{r} - \eta \mathbf{1}) \quad (7.4)$$

where μ is the Lagrange multiplier. As shown in Figure 7.3, we define $V(j, f)$ as the set of downstream nodes traversed by flow f from node j . Suppose the source node of flow f is $s(f)$, then $V(s(f), f)$ represents the set of all nodes in the path of flow f . For each intermediate node in the path of flow f , flow f enters the node via the input link and leaves it via the output link. We define $h_i(n, f)$ and $h_o(n, f)$ as the input and output links of flow f on node n respectively, if flow f goes through n . For example, $h_i(C, 1) = 1$ and $h_o(C, 1) = 2$, in the example of Figure 7.2. At the source node of flow f , $h_i(s(f), f) = \infty$; while at the destination node of flow f (denoted by $d(f)$), $h_o(d(f), f) = \infty$. Furthermore, we use $w_1^i(n, f), w_1^o(n, f), w_2^i(n, f), w_2^o(n, f)$ to denote the

association between two NICs at node n and two links ($h_i(n, f)$ and $h_o(n, f)$), which is defined as,

$$w_1^i(n, f) = \begin{cases} 1, & \text{if } l_i(n, f) \text{ uses NIC 1} \\ 0, & \text{otherwise.} \end{cases}$$

The definitions of other three notations are similar.

Figure 7.3: Flow notations

Differentiating $L(\vec{r}, \mu)$ in Equation (7.4) and setting the derivative to zero yields,

$$\begin{aligned} \frac{dL}{dr_f} &= d(\sum_{f \in F} (-\frac{W_f}{r_f}) + \mu^T (MA\vec{r} - \eta\mathbf{1})) / dr_f \\ &= \frac{W_f}{r_f^2} - \sum_{j \in V(s(f), f)} ((\frac{w_1^i(j, f)}{c_{h_i(j, f)}} + \frac{w_1^o(j, f)}{c_{h_o(j, f)}}) \mu_{2j} + (\frac{w_2^i(j, f)}{c_{h_i(j, f)}} + \frac{w_2^o(j, f)}{c_{h_o(j, f)}}) \mu_{2j+1}) \\ &= 0 \end{aligned} \quad (7.5)$$

Obviously, the solution to this optimization problem should satisfy the following condition:

$$(r_f^*)^2 = \frac{W_f}{\sum_{j \in V(s(f), f)} ((\frac{w_1^i(j, f)}{c_{h_i(j, f)}} + \frac{w_1^o(j, f)}{c_{h_o(j, f)}}) \mu_{2j} + (\frac{w_2^i(j, f)}{c_{h_i(j, f)}} + \frac{w_2^o(j, f)}{c_{h_o(j, f)}}) \mu_{2j+1})} \quad (7.6)$$

Based on these results, we can design our end-to-end congestion control mechanism to adjust the source rates. Suppose the transmission rate of flow f at time t is denoted by $r_f(t)$, which should be adjusted according to

$$\frac{dr_f(t)}{dt} = \kappa(W_f - r_f^2(t)) \sum_{j \in V(s(f), f)} ((\frac{w_1^i(j, f)}{c_{h_i(j, f)}} + \frac{w_1^o(j, f)}{c_{h_o(j, f)}}) \mu_{2j}(t) + (\frac{w_2^i(j, f)}{c_{h_i(j, f)}} + \frac{w_2^o(j, f)}{c_{h_o(j, f)}}) \mu_{2j+1}(t))) \quad (7.7)$$

where

$$\begin{aligned}\mu_{2j}(t) &= p_{2j}(\sum_{k \in \mathbf{S}(j)} r_k(t) (\frac{w_1^i(j,k)}{c_{h_i(j,k)}} + \frac{w_1^o(j,k)}{c_{h_o(j,k)}})) \\ \mu_{2j+1}(t) &= p_{2j+1}(\sum_{k \in \mathbf{S}(j)} r_k(t) (\frac{w_2^i(j,k)}{c_{h_i(j,k)}} + \frac{w_2^o(j,k)}{c_{h_o(j,k)}}))\end{aligned}\quad (7.8)$$

where $\mathbf{S}(j)$ is the set of all flows going through node j . $p_{2j}(y)$ and $p_{2j+1}(y)$ are equal to the same price function $m_j(y)$, which represents the price of link resources consumed by flows traversing node j . The price function is an indication of congestion state at a node, and this indication should be referred in the adaption of source rate [50]. We will discuss the price function in more detail in next subsection.

Note that $\mu_{2j}(t)$ and $\mu_{2j+1}(t)$ denote the price for using NIC 1 and NIC 2 on node j respectively. Furthermore, $r_f(t)(\frac{w_1^i(j,f)}{c_{h_i(j,f)}} + \frac{w_1^o(j,f)}{c_{h_o(j,f)}})$ represents the MAC time fraction of NIC 1 on node j used by flow f ; while $r_f(t)(\frac{w_2^i(j,f)}{c_{h_i(j,f)}} + \frac{w_2^o(j,f)}{c_{h_o(j,f)}})$ represents the MAC time fraction of NIC 2 on node j used by flow f . As a result, $r_f^2(t) \sum_{j \in V(s(f),f)} ((\frac{w_1^i(j,f)}{c_{h_i(j,f)}} + \frac{w_1^o(j,f)}{c_{h_o(j,f)}}) \mu_{2j}(t) + (\frac{w_2^i(j,f)}{c_{h_i(j,f)}} + \frac{w_2^o(j,f)}{c_{h_o(j,f)}}) \mu_{2j+1}(t))$ can be interpreted as the total cost of flow f at node j . In Equations (7.7) and (7.8), the aggregated price function of all nodes traversed by flow f is controlled to approach a predefined parameter W_f .

Next we establish the Lyapunov function

$$\Psi(\vec{r}) = \sum_{f \in \mathbf{F}} -\frac{W_f}{r_f} - \sum_{j \in \mathbf{V}} (\int_0^{\sum_{k \in \mathbf{S}(j)} r_k(t) (\frac{w_1^i(j,k)}{c_{h_i(j,k)}} + \frac{w_1^o(j,k)}{c_{h_o(j,k)}})} p_{2j}(y) dy + \int_0^{\sum_{k \in \mathbf{S}(j)} r_k(t) (\frac{w_2^i(j,k)}{c_{h_i(j,k)}} + \frac{w_2^o(j,k)}{c_{h_o(j,k)}})} p_{2j+1}(y) dy) \quad (7.9)$$

for the system of differential Equations (7.7)-(7.8). Then we can prove that the extremum point $\vec{r}^*$ is the stable point of this system.

Theorem 7.1 *The strictly concave function $\Psi(\vec{r})$ is a Lyapunov function for the system of differential Equations (7.7)-(7.8). The unique value $\vec{r}^*$ maximizing $\Psi(r)$ is a stable point of the system, to which all trajectories converge.*

Proof: $r_f \geq 0$, $W_f > 0$ and $p_j(y)$ ensure that $\Psi(\vec{r})$ is a strictly concave function. We have,

$$\begin{aligned}\frac{\partial}{\partial r_f} \Psi(r) &= \frac{W_f}{r_f^2} - \sum_{j \in V(s(f),f)} [(\frac{w_1^i(j,f)}{c_{h_i(j,f)}} + \frac{w_1^o(j,f)}{c_{h_o(j,f)}}) p_{2j}(\sum_{k \in \mathbf{S}(j)} r_k(t) (\frac{w_1^i(j,k)}{c_{h_i(j,k)}} + \frac{w_1^o(j,k)}{c_{h_o(j,k)}})) \\ &\quad + (\frac{w_2^i(j,f)}{c_{h_i(j,f)}} + \frac{w_2^o(j,f)}{c_{h_o(j,f)}}) p_{2j+1}(\sum_{k \in \mathbf{S}(j)} r_k(t) (\frac{w_2^i(j,k)}{c_{h_i(j,k)}} + \frac{w_2^o(j,k)}{c_{h_o(j,k)}}))]\end{aligned}\quad (7.10)$$

setting these derivatives to zero identifies the maximum value. We define an intermediate function

as,

$$\begin{aligned} \Phi(f, t) = & \sum_{j \in V(s(f), f)} \left[\left(\frac{w_1^i(j, f)}{c_{hi}(j, f)} + \frac{w_1^o(j, f)}{c_{ho}(j, f)} \right) p_{2j} \left(\sum_{k \in S(j)} r_k(t) \left(\frac{w_1^i(j, k)}{c_{hi}(j, k)} + \frac{w_1^o(j, k)}{c_{ho}(j, k)} \right) \right) \right. \\ & \left. + \left(\frac{w_2^i(j, f)}{c_{hi}(j, f)} + \frac{w_2^o(j, f)}{c_{ho}(j, f)} \right) p_{2j+1} \left(\sum_{k \in S(j)} r_k(t) \left(\frac{w_2^i(j, k)}{c_{hi}(j, k)} + \frac{w_2^o(j, k)}{c_{ho}(j, k)} \right) \right) \right] \end{aligned} \quad (7.11)$$

Then we have

$$\frac{\partial}{\partial r_f} \Psi(r) = \frac{W_f}{r_f^2} - \Phi(f, t) \quad (7.12)$$

In addition, from Equation (7.7), we have

$$\frac{dr_f(t)}{dt} = \kappa(W_f - r_f^2(t)\Phi(f, t)) \quad (7.13)$$

Combining Equation (7.12) and Equation (7.13) yields

$$\begin{aligned} \frac{d}{dt} \Psi(\vec{r}(t)) &= \sum_{f \in \mathbf{F}} \frac{\partial \Psi}{\partial r_f} \cdot \frac{d}{dt} r_f(t) \\ &= \kappa \sum_{f \in \mathbf{F}} \frac{1}{r_f^2(t)} (W_f - r_f^2(t)\Phi(f, t))^2 \end{aligned} \quad (7.14)$$

which ensures that $\Psi(\vec{r}(t))$ is strictly increasing with t , until $\vec{r}(t) = \vec{r}^*$, the maximum value of $\Psi(\vec{r})$. Thus, the function $\Psi(\vec{r})$ is a Lyapunov function of system of the Equations (7.7)-(7.8). The theorem follows ([50], Theorem 1). $\blacksquare$

As a result, the global stability of our end-to-end algorithm in Equation (7.7) has been proven, and the stable point of our end-to-end algorithm is $\vec{r}^*$ which maximizes $\Psi(r)$.

7.4.3 Price Function

As we have discussed earlier in Section 7.4.2, each router in WMN is associated with a price function $p_j(\cdot)$. Since each node in WMNs acts as both end-host and router, it also has a corresponding price function. Price function indicates the price of a time fraction used by a flow according to total transmission time accessed at this node. Furthermore, this function should satisfy,

- $p_j(y) \in [0, 1]$, since it only marks a fraction of flows;
- It is an increasing function. The more the time utilized, the greater the price function should be;

- It is Lipschitz continuous. This condition makes price function “smooth”.

In this chapter, we adopt the following expression as the price function,

$$p_j(y) = \max(0, \frac{y - \tilde{\tau}}{y}) \quad (7.15)$$

where the input parameter y is the sum of time fraction used by incoming and outgoing flows in one NIC at node j . Therefore, $p_j(y)$ marks the portion of flows which exceeds the threshold $\tilde{\tau}$. $\tilde{\tau}$ should be less than 1 since it controls the desired time utilization of the link. Furthermore, $\tilde{\tau}$ is an important parameter to solve the optimization problem (7.3). The setting of this parameter should ensure that $\tilde{\tau}^*$ becomes the solution of the optimization problem [92].

7.4.4 Hop-by-hop Congestion Control Algorithm

So far, we have derived the optimal solution for the end-to-end congestion control, where the congestion state information is feedback hop-by-hop from the destination node or congested node to the source node. Compared with end-to-end algorithms which control the source rate directly, hop-by-hop algorithms have shorter response time for congestion control. In a hop-by-hop mechanism, the preceding nodes, instead of source node, adjust their transmission rates after receiving congestion indication. The idea behind hop-by-hop control algorithm is that all intermediate nodes operate a congestion control algorithm for the flows passing through them. In the following, we develop the hop-by-hop congestion control algorithm based on Equation (7.7).

We first introduce the definition of “congestion state” of flow f on node j as

$$\alpha_{j,f}(t) = (\frac{w_1^i(j, f)}{c_{h_i(j, f)}} + \frac{w_1^o(j, f)}{c_{h_o(j, f)}})\mu_{2j}(t) + (\frac{w_2^i(j, f)}{c_{h_i(j, f)}} + \frac{w_2^o(j, f)}{c_{h_o(j, f)}})\mu_{2j+1}(t) \quad (7.16)$$

$\alpha_{j,f}(t)$ consists of two terms, which are related with NIC1 and NIC2 respectively. The first term is the product of the occupied time fraction at NIC1 and the price function of NIC1. The second term has similar meaning related to NIC2.

The end-to-end algorithm in Equation (7.7) controls the source rate directly with full knowledge of $\alpha(t)$ and capacity of all links in the path of the flow. The knowledge is feedback hop-by-hop from the destination node to the source node along the path. In such case, an intermediate node

in the path of the flow can receive only a partial knowledge, i.e., the information of $\mu(t)$ from its downstream nodes. Thus the congestion control algorithm should operate based on this partial knowledge. We assume all intermediate nodes have the exact knowledge of W_f for each flow that is incident on the node, as well as the local link capacities. With the $\mu(t)$ information from downstream nodes, we modify the algorithm in Equation (7.7) as follows.

$$\frac{dv_f^k}{dt} = \kappa(W_f - (a_f^k(t))^2) \sum_{j \in V(n,f)} \left(\left(\frac{w_1^i(j,f)}{c_{hi}(j,f)} + \frac{w_1^o(j,f)}{c_{ho}(j,f)} \right) \mu_{2j}(t) + \left(\frac{w_2^i(j,f)}{c_{hi}(j,f)} + \frac{w_2^o(j,f)}{c_{ho}(j,f)} \right) \mu_{2j+1}(t) \right) \quad (7.17)$$

$$a_f^k(t) = \min(v_f^k(t), a_f^{k-1}(t)) \quad (7.18)$$

where $a_f^k(t)$ and $v_f^k(t)$ are the actual and “virtual” transmission rates of flow f at the k -th hop at the path respectively, and n is the source node of k -th hop. We introduce the concept of “virtual” rate ($v_f^k(t)$) as the theoretical adaption target of transmission rate. Combining with actual available bandwidth ($a_f^{k-1}(t)$), we can obtain the practical transmission rate of flow f ($a_f^k(t)$). The definition of $\mu(t)$ is similar to that in Equation (7.8), but the source rates are replaced by the actual transmission rates at the node.

The congestion controller on each intermediate node sums up the congestion states ($\alpha(t)$) of all downstream nodes in the path. Based on the results, the controller calculates the maximum possible transmission rates for every flow which traverses the node. To utilize bandwidth efficiently, the controller chooses the smaller one of the maximum possible rate and the data arriving rate (transmission rate of the preceding node) to be the actual transmission rate.

In the following, we prove the convergence and stability of this hop-by-hop congestion control algorithm.

Theorem 7.2 *The transmission rates determined by the hop-by-hop congestion controller represented by Equations (7.17) and (7.18) converge to the stable point $\bar{r}^*$ given in Theorem 7.1, i.e. for each hop k and each flow f , $a_f^k(t) \rightarrow r_f^*$ as $t \rightarrow \infty$.*

Proof: Equation (7.18) implies that $a_f^k(t) \leq a_f^{k-1}(t)$. In other words, $a_f^k(t)$ decreases along the path from the source to the destination. Furthermore, the Lagrange multiplier part of Equation (7.17) decreases along the path, since the closer to the destination the less number of non-

negative terms added. These two factors imply

$$\frac{d}{dt}(v_f^k(t) - v_f^{k-1}(t)) = \frac{d}{dt}v_f^k(t) - \frac{d}{dt}v_f^{k-1}(t) \geq 0$$

Combining with the initial value of $v_f^k(t)$, we have

$$v_f^k(t) \geq v_f^{k-1}(t) \quad (7.19)$$

Furthermore, Equation (7.18) implies,

$$a_f^k(t) \leq v_f^k(t) \quad (7.20)$$

Hence, we obtain $v_f^k(t) \geq v_f^{k-1}(t) \geq a_f^{k-1}(t)$ and

$$\begin{aligned} a_f^k(t) &= \min(v_f^k(t), a_f^{k-1}(t)) \\ &= a_f^{k-1}(t) \end{aligned}$$

Obviously, the actual transmission rates $a_f^k(t)$, $f \in \mathbf{F}$ are the same along all hops at the path. Furthermore, at the source node, the hop-by-hop congestion controller operates in the same way as the controller defined in Equation (7.7). In other words, the rate dynamic of flow f in this hop-by-hop controller is the same as that in the controller defined in Equation (7.7). According to Theorem 7.1, our proof completes. ■

7.5 Performance Evaluation

In this section, we use simulation experiments based on ns-2 [7] to demonstrate the effectiveness of our new congestion control algorithm. Response time, convergence rate and the queue length at the bottleneck node, etc., are used as the performance metrics of congestion control schemes in our simulation experiments.

7.5.1 Simulation Scenario

We consider a network topology, which represents a tree-like WMN, as shown in Figure 7.4. Our main interest is the performance of our new congestion control algorithms on a bottleneck WMN

node. In this topology, there are $2N$ source nodes and two destination nodes. Each source node has only one flow to transmit. The source nodes are partitioned into two sets. Source nodes 1 to N are sending flows to destination node 2; while source nodes $N + 1$ to $2N$ are sending flows to destination node 1. The bottleneck node B is located just preceding to these two destination nodes, and all flows traverse node B . Among the links connected to node B , those above the dashed line are using NIC 1, while the rest are using NIC 2. There are L wireless hops between each of the source nodes and its destination node. In this experiment, we set $L = 5$ and $N = 2$. The capacity of all links is set to 100 kbytes/sec. The one way transmission delay time per-hop is 4 milli-second. Thus the round trip time between source and destination nodes is 40 milli-second. Suppose all flows have the same weight value, $W_f = W, f \in \mathbf{F}$. Other parameters in this experiment are listed in Table 7.1.

Figure 7.4: Simulation topology

Parameters	Value
κ	1
W	12
$\tilde{\tau}$	0.42
η	1

Table 7.1: Simulation parameters

First, we examine the transient behaviors of the aggregated transmission rates using the end-to-end and hop-by-hop algorithms respectively. Figure 7.5 shows the aggregated transmission rate,

which can also be interpreted as the system throughput, of all four flows. The simulation time duration is expressed in time units, which equals to $100ms$. We can observe that the aggregated transmission rates can converge to almost the same stable constant value for both algorithms at around 2.8 ms . Before reaching the peak points, the aggregated rates of both algorithms increase roughly linearly, and these two peak points are very close. After reaching the peak points, the network enters the congestion period. During the congestion period, the end-to-end algorithm reaches the stable point faster than the hop-by-hop algorithm. The aggregated rate of hop-by-hop scheme falls below the stable point during $3 \sim 4.6\text{ ms}$, then re-increases to the stable point as shown in Figure 7.5; while the end-to-end algorithm converges to the stable point without further fluctuation. The reason is that the transmission rate control is based on overall network congestion conditions in end-to-end algorithm. But in hop-by-hop algorithm, the intermediate node determines the rates based on the congestion condition of its downstream nodes. This causes that the hop-by-hop algorithm has shorter response time but coarser control accuracy. After about 4.8 ms , both algorithms enter the stable period, in which the aggregated transmission rate has only very small variance. Since we set $\eta = 1$, the aggregated transmission rate reaches almost 100 kbps , i.e., the system utilization is very close to 100% , after the system enters stable period. This result is in consistence with Theorem 7.2: the hop-by-hop algorithm can converge to the same stable point as that in end-to-end algorithm.

Figure 7.5: Aggregated transmission rates of all flows(hop-delay=4ms)

Next, we examine the queue length at the bottleneck node and the packet delay. Figure 7.6 shows the queue lengths at the bottleneck node for both algorithms. Compared with the end-to-end algorithm, the hop-by-hop algorithm occupies a significant smaller buffer size. Furthermore, the occupied queue length decreases quickly after the peak point. As a result, the queueing delay at bottleneck node is notably decreased as shown in Figure 7.7, where x axis is time in 0.5 unit and y axis represents the average delay of packets arrives in the past 0.5 unit time. We can observe that although the hop-by-hop algorithm has slower convergence than the end-to-end algorithm, it outperforms significantly the latter in terms of queue length at bottleneck node and packet delay.

Figure 7.6: Queue length at the bottleneck(hop-delay=4ms)

Next, we change the simulation scenario by increasing hop transmission delay to 8ms to examine the performance of our algorithms under long RTT scenario. The numerical results are shown in Figures 7.8, 7.9 and 7.10. In Figure 7.8, we can see that there is a small fluctuation in the aggregated transmission rates for the end-to-end algorithm, which does not exist in previous scenario, before reaching stable point in the end-to-end algorithm. The hop delay can also affect the performance of our hop-by-hop congestion control algorithms, as the convergence becomes slower. But the algorithm can still converge to the stable point. From the results of queue length at bottleneck node and packet delay shown in Figures 7.9 and 7.10, we can find that the hop-by-hop algorithm performs even more significantly better than the end-to-end algorithm in the large delay scenario. As a result, we argue that the hop-by-hop algorithm is more suitable for the WMN with multi-

Figure 7.7: Average packet end-to-end delay(hop-delay=4ms)

channel MAC.

Figure 7.8: Aggregated transmission rates of all flows(hop-delay=8ms)

7.6 Conclusion

In this chapter, we have proposed a new hop-by-hop congestion control algorithm for WMN with multi-channel MAC. In such networks, several non-overlapped channels are used and each node is equipped with two NICs. Instead of link constraint, we use a stronger time constraint, i.e. a

Figure 7.9: Queue length at the bottleneck(hop-delay=8ms)

Figure 7.10: Average packet end-to-end delay(hop-delay=8ms)

physical NIC can be occupied by only one user at any time instant. Our new algorithm solves the system throughput maximization problem with time constraint for such a system. Furthermore the stability and convergence of this new algorithm is theoretically proved. The numerical results of simulation experiments validate the effectiveness of our new algorithm and our theoretical analysis. We also find that the hop-by-hop algorithm can significantly outperform the end-to-end algorithms in terms of the queue length and the packet delay.

Chapter 8

Conclusions and Future Work

In this chapter, we will conclude our research work in this Ph.D thesis. In addition, some possible future research directions and our initial ideas will be presented.

8.1 Conclusions

The research work in this thesis focuses on the traffic control mechanisms for future wireless networks. Our contributions include the following research projects.

1. Channel-State-Aware WISPER for CDMA Networks

As the first step of the research work for wireless packet scheduling, we propose and study a channel-state-aware WISPER (CSA-WISPER) scheme for CDMA networks in Chapter 3. In order to provide QoS guarantees for multiple class traffic in CDMA networks, an efficient scheduling scheme must consider not only wireless channel variability but also special properties of CDMA networks, such as multi-class packets in the same slot. The CSA-WISPER utilizes the two-state channel model, and tries to guarantee fairness among all the flows. We use simulations based on NS-2 to evaluate the performance of several proposed mechanisms. The numerical results show that CSA-WISPER can achieve higher throughput and lower packet delay than original WISPER.

2. Channel-State Dependent Fair Scheduling for CDMA/HDR Networks

Next, a new packet scheduling scheme designed specially for high data rate CDMA network is proposed in Chapter 4. This scheme is named Channel States Dependent Fair Service for CDMA (CSDFSC). First, we investigate the appropriate multiplexing method for high data rate CDMA network. There are two main multiplexing schemes: TDM and CDM. In TDM, mobile users transmit in high data rates but only one user can transmit at one time; while in CDM multiple users transmit at the same time to share channel bandwidth with low data rates. Both schemes have their own advantages and disadvantages. According to our mathematical analysis, TDM instead of CDM should be used in most situations. In other words, one user transmitting in a timeslot at full rate is more efficient than multiple users transmitting simultaneously in a timeslot with low transmission rates.

CSDFSC uses the multi-state Markov model as the CDMA wireless channel state model, and tries to emulate GPS to provide fair services for best-effort traffic flows. Furthermore, flow weights are used to differentiate guaranteed and best-effort traffic flows. After comparing CSDFSC with WISPER via ns-2 simulation, we find that CSDFSC performs much better than WISPER in terms of channel capacity, link resource utilization, packet loss ratio, fairness between best-effort traffic, end-to-end delay of guaranteed traffic and so on. Because CSDFSC uses the multi-state channel model, the scheduler knows quite well the current channel states and can effectively avoid transmission loss. Obviously CSDFSC will experience less packet losses and achieve higher channel utilization than WISPER. The good performance in providing fair service to TCP flows shows that CSDFSC emulates GPS quite well and can guarantee the bandwidth proportion shared by each flow. In WISPER guaranteed traffic flows experience similar end-to-end delay as best-effort traffic flows. However in CSDFSC guaranteed traffic flows experience a much smaller end-to-end delay than best-effort traffic flows, and the delay is also smaller than that of guaranteed traffic flows in WISPER. From these results, we can obtain the conclusion that the queue weight setting mechanism in CSDFSC is effective.

3. Rate-guaranteed Opportunistic Scheduling for Wireless Networks

Rate-guaranteed Opportunistic Scheduling scheme (ROS) is introduced in Chapter 5. By measuring the “relative throughput” differences between flows, ROS can provide rate-guaranteed service for mobile hosts and the ROS scheduler is proved to be a GR node with loss, whose

definition is extended from the model of service curve with loss. By exploiting wireless channel states, ROS chooses the flow with the maximum “opportunity” to transmit in the next time-slot. We prove that ROS can achieve the maximum throughput compared with other GR nodes by theoretical analysis and numerical results. As a GR node with loss, ROS can also provide a statistical delay bound for regulated incoming flows. Besides all these advantages, ROS has another important merit - low implementation complexity.

4. Admission Control Coordinated with ROS

In order to provide statistical QoS guarantees, we propose a new admission control policy to work in conjunction with ROS for wireless networks in Chapter 6. As an important QoS requirement, packet loss rate is used as the admission criteria in our proposed admission control policy. In this thesis, we consider a system in which each flow is allocated a fixed length buffer at the router. Through mathematical analysis, we investigate the behavior of buffer length at the base-station and buffer overflow probabilities of flows. The actual PLRs of all flows can be derived from the buffer overflow probabilities. Our proposed new admission control policy will admit new user if all active users’ PLR requirements are not violated after admission and new user’s PLR requirement is also satisfied. The simulation results validate the effectiveness of our proposed admission control algorithm.

5. Congestion Control for WMN with Multi-channel MAC

We present a new hop-by-hop congestion control algorithm for WMNs with multi-channel MAC in Chapter 7. In WMNs with multi-channel MAC, several non-overlapped channels are used and each node is equipped with two NICs. Instead of the link constraint, we use a stronger time constraint, i.e. a physical NIC can be occupied by only one user at any time instant. We propose a new end-to-end algorithm to solve the system throughput maximization problem with time constraint for such a system. Based on the proposed end-to-end algorithm, we develop a distributed hop-by-hop algorithm. Furthermore, the stability and convergence of these two algorithms are theoretically proved. The numerical results of simulation experiments validate the effectiveness of our new algorithms and our theoretical analysis. We also find that the proposed hop-by-hop algorithm can significantly outperform the end-to-end

algorithm in terms of queue length and packet delay.

8.2 Future Work

The next step of our research will focus on the integration of these packet scheduling, admission control and congestion control mechanisms. A traffic control framework consisting of packet scheduling, admission control, congestion control and other necessary control mechanisms, is a possible solution. Some problems which should be addressed by this framework are described below.

- Cross-layer design. Recently there has been increased interest in protocols for wireless networks that rely on significant interactions between various layers of the network stack. This so-called “cross-layer design” may be appropriate for our traffic control framework, since traffic control mechanisms operate at various layers. By exchanging and utilizing information between different layers, “cross-layer design” usually achieves tempting performance improvement. However, it should be used carefully without violating the network architecture too much [49], since the unintended cross-layer interactions may cause some undesirable consequences to system performance and stability. There are some principles should be considered in a cross-layer design. The dependency relation between different layers should be examined carefully and the time-scale separation of different layer mechanisms should be enforced to reduce the conflicts between these mechanisms.
- Power-aware traffic control. Power constraints exist for most wireless hosts, since they usually use battery as the power supply. Therefore, practical wireless communication protocols/mechanisms should be power-efficient. There has been some significant work which considers power-efficiency in the design of traffic control mechanisms [20, 27]. Our optimal solution for wireless scheduling or congestion control may be extended to meet both original optimal design goal and minimum power consumption of mobile hosts. The extended power-aware traffic control mechanisms should adapt intelligently according to not only energy balance or the remaining battery life, but also the corresponding QoS requirements.
- Traffic control based on the general model of multi-hop wireless networks. As the multi-hop wireless networks (e.g. wireless mesh network and wireless sensor network) become the trend

of future wireless networks, an important direction for future research is to define a general model for the multi-hop wireless networks. From such modeling and analysis, we can obtain results which are helpful not only for the design of efficient traffic control mechanisms, but also for the development of effective routing and MAC protocols.

Author's Publications

- *Fei Long*, Gang Feng, Chee Kheong Siew, "Channel States Dependent Fair Service: A New Packet Scheduling Algorithm for CDMA," Computer Network, Volume 49, Issue 2 ,October 2005, Pages 201-216.
- *Fei Long*, Gang Feng and Junhua Tang, "A Joint Scheduling/Admission Control Framework for Wireless Networks," Journal of Communications and Networking, vol. 7, Dec. 2005, pp. 408-416.
- Chee-Kheong Siew, Gang Feng, *Fei Long*, and Meng-Hwa Er, "Congestion Control Based on Flow-state-dependent Dynamic Priority Scheduling", IEE Proc. Communications, Vol.152, Issue 5. 2005, pp. 548-558.
- *Fei Long*, Gang Feng, Chee Kheong Siew, "A Modified BER Scheduling Scheme for Wireless Multimedia Access Control Protocol," Proceedings of The Ninth IEEE International Conference on Communications Systems (ICCS 2004), pp 205-209.
- *Fei Long* and Gang Feng, "Hop-by-hop Congestion Control for Wireless Mesh Networks with Multi-channel MAC," submitted to Globecom 2007.
- *Fei Long* and Gang Feng, "Efficient Rate Guaranteed Opportunistic Scheduling," submitted to IEEE Trans. on Vehicular Technology for publication.
available at <http://www.ntu.edu.sg/home5/pg03753229/lftvt.pdf>.
- *Fei Long* and Gang Feng, "Optimizing Congestion Control for Wireless Mesh Networks with Multi-channel MAC," submitted to IEEE Trans. on Vehicular Technology for publication.
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