Packet Level Quality of Service Analysis of Multiclass Services in a WCDMA Mobile Network

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A THESIS SUBMITTED

FOR THE DEGREE OF MASTER OF ENGINEERING

DEPARTMENT OF ELECTRICAL AND COMPUTER ENGINEERING

NATIONAL UNIVERSITY OF SINGAPORE

2003

Acknowledgements

With this opportunity I would like to express my sincere appreciation to all people who have helped me to finish this thesis.

First of all, I would like to express my great gratitude to my supervisors, Dr. CHEW Yong Huat and Dr. WONG Tung Chong. They provided me with this research project and kindly guided me with my work for two and half years. Dr. CHEW and Dr. WONG have provided a lot of valuable advices and ideas which open my mind and help me to overcome all difficulties along the way for my research work. It would be impossible for me to complete this thesis without their long-term assistance and guidance.

Furthermore, it gives me pleasure to thank Xiao Lei and Yao Jianxin, who are my partners in our research group. I often discussed problems that I encountered in my work with them and gained many benefits from their suggestions.

Next, I am also grateful to Yang Yang, who is my lab-mate. He helped me with some programming issues and techniques with Mathematica and Matlab softwares.

Besides, many thanks for Wang Xiaofeng, one of my good friends in NUS. He gave me much help when I first came to Singapore and helped me adapt to the new environment quickly.

Finally, I specially give my hearty gratitude to my families, including my parents and my elder brother. They encouraged me to pursue this Master of Engineering degree in NUS and supported me throughout the past few years. Their selfless help and kind concern play a critical role in my studies and work.

Contents

Summary

In the future Universal Mobile Telecommunications System (UMTS) network, Quality of Service (QoS) provisioning is a critical issue. In contrast to earlier generations of telecommunication systems, the UMTS network can enable a variety of services with different QoS requirements within each mobile user simultaneously. Wideband CDMA (WCDMA) is chosen as the multiple access technology and the air interface of UMTS. This thesis studies the QoS performances in the WCDMA system. Due to the unique characteristics of the UMTS network, a complete and detailed QoS architecture is proposed to deal with all related topics on QoS provisioning. All services in the UMTS network are classified into four classes in the proposed QoS architecture. The services consist of the conversational class, streaming class, interactive class and background class. These four classes are different in terms of their QoS requirements. Although QoS provisioning issues have long attracted a lot of research interests and many discussions have been made in this area, no analytical work has been done to solve the QoS provisioning problems for all the four UMTS classes. The objective of this thesis is to address the packet level QoS issues at the network layer of the WCDMA system with deterministic mathematical methods. Besides, it is also our aim to give a QoS-based call admission control (CAC) algorithm at the packet level of the network layer and to obtain the corresponding feasible admission regions (ARs). In this thesis, we study the wireless channel between mobile users and base stations and focus our work on the uplink of the WCDMA system.

Firstly, this thesis introduces the rudimentary UMTS network and its QoS architecture. We develop two system models for analysis based on them. The two system models are called single-connection system model and multi-connection system model, respectively. Only a single service is permitted within each mobile user in the single-connection system model, while multi-connection multiclass services are permitted within each mobile user in the multi-connection system model. Assuming perfect power control, efficient power distribution algorithms are developed in the two system models. The Go-Back-N (GBN) automatic retransmission request (ARQ) mechanism is used for the services of the interactive and background classes. The effects of the GBN ARQ in the WCDMA channel are examined in details. The outage probability of each class is formulated for each service in the single-connection and multi-connection system models, taking into consideration of the effects of the GBN ARQ.

Secondly, we present the packet level QoS performances, including packet loss rate and average delay, for all services in the WCDMA system. The packet level QoS performances are directly associated with the data link layer QoS attributes, such as outage probability. Accurate mathematical formulas are developed for the outage probabilities, the packet loss rates and the average delays of each service in the two system models.

Lastly, a QoS-based CAC algorithm is given, satisfying the packet level QoS requirements of all admitted services. Furthermore, we derive the ARs for the two system models based on this CAC scheme and appropriate system parameters. The ARs can assure that any admitted service in the WCDMA system is able to achieve its required QoS levels.

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3G	Third Generation
3GPP	Third Generation Partnership Project
ACK	Acknowledgement
AMPS	Advanced Mobile Phone System
AR	Admission Region
ARQ	Automatic Repeat Request
ASK	Amplitude Shift Keying
BE	Best Effort
BER	Bit Error Rate
BPSK	Binary Phase Shifting Keying
BS	Base Station
CAC	Call Admission Control
CBR	Constant Bit Rate
CDMA	Code Division Multiple Access
CDF	Cumulative Distribution Function
CN	Core Network
CS	Circuit Switching
DCH	Dedicated Channel

DS-CDMA	Direct Sequence CDMA
ETSI	European Telecommunications Standards Institute
FCFS	First Come First Serve
FDD	Frequency Division Duplex
FDMA	Frequency Division Multiple Access
FEC	Forward Error Correction
FSK	Frequency Shift Keying
FTP	File Transfer Protocol
GSM	Global System for Mobile Communications
GGSN	Gateway GPRS Support Node
GMSC	Gateway MSC
GPRS	General Packet Radio System
HLR	Home Location Register
IEEE	Institute of Electrical and Electronic Engineers
IMT-2000	International Mobile Telephony, 3rd Generation Networks
ITU	International Telecommunication Union
IP	Internet Protocol
IS-95	Interim Standard 95
ITU	International Telecommunication Union
LC	Load Control
MAC	Medium Access Control
MAI	Multiple Access Interference

ME	Mobile Equipment
MS	Mobile Station
MSC	Mobile Services Switching Center
NACK	Negative Acknowledgement
Node B	UMTS Base Station
PC	Power Control
PDC	Personal Digital Cellular
PDF	Probability Density Function
PLMN	Public Land Mobile Network
PN	Pseudo Noise
PS	Packet Switching
PSK	Phase Shift Keying
OVSF	Orthogonal Variable Spreading Factor
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
RAB	Radio Access Bearer
RAN	Radio Access Network
RNC	Radio Network Controller
RRM	Radio Resource Management
SIM	Subscriber Identity Module
SINR	Signal to Interference Plus Noise Ratio
SGSN	Serving GPRS Support Node
SMS	Short Message Service

SS	Spreading Spectrum
TDD	Time Division Duplex
TE	Terminal Equipment
TS	Technical Specification
UE	User Equipment
UL	Uplink
UMTS	Universal Mobile Telecommunication Service
USIM	UMTS Subscriber Identity Module
UTRA	Universal Terrestrial Radio Access
UTRAN	UMTS Terrestrial Radio Access Network
VBR	Variable Bit Rate
VLR	Visitor Location Register
WCDMA	Wideband Code Division Multiple Access
Web-browsing	World Wide Web Browsing

List of Symbols

a_{off}	Location parameters for the Pareto off period of a non-real-time service.
a_{on}	Location parameters for the Pareto on period of a non-real-time service.
$a_{k,off}$, $k \in \{3,4\}$	Location parameter for the Pareto off period of a web-browsing and data service, respectively.
$a_{k,on}$, $k \in \{3,4\}$	Location parameter for the Pareto off period of a web-browsing and data service, respectively.
A	Area of a square cell.
α	Transition rate from the off state to the on state of a low-bit-rate video minisource.
α_k , $k \in \{1,3,4\}$	Transition rates from the off state to the on state of voice, web-browsing and data, respectively.
B	Buffer size of a non-real-time service.
B_k , $k \in \{3,4\}$	Buffer sizes for of web-browsing and data, respectively.
BER_k^* , $k \in \{1,2l,2h,3,4\}$	BER requirements of voice, video (low-bit-rate), video (high-bit-rate), web-browsing and data, respectively.
β	Transition rate from the on state to the off state of a low-bit-rate

	video minisource.
β_k, $k \in \{1,3,4\}$	Transition rates from the on state to the off state of voice, web-browsing and data, respectively.
c_{off}	Shape parameters for the Pareto off period of a non-real-time service.
c_{on}	Shape parameters for the Pareto on period of a non-real-time service.
$c_{k,off}$, $k \in \{3,4\}$	Shape parameter for the Pareto off period of a web-browsing and data service, respectively.
$c_{k,on}$, $k \in \{3,4\}$	Shape parameter for the Pareto on period of a web-browsing and data service, respectively.
$D_{i,k}$, $k \in \{1,2l,2h,3,4\}$	Delays of voice, video (low-bit-rate), video (high-bit-rate), web-browsing and data, respectively within *i*th mobile user in the multi-connection system model.
D_k, $k \in \{1,2l,2h,3,4\}$	Delays of voice, video (low-bit-rate), video (high-bit-rate), web-browsing and data, respectively in the single-connection system model.
D_k^*, $k \in \{1,2l,2h,3,4\}$	Delay requirements of voice, video (low-bit-rate), video (high-bit-rate), web-browsing and data, respectively.
ε_m, ε_d	Two independent Guassian random variables with zero mean and σ^2 variance.
η	Average power of AWGN.
G_k,	Spreading gains of voice, video (low-bit-rate), video (high-bit-rate),

	web-browsing and data, respectively.
γ_k^*, $k \in \{1, 2l, 2h, 3, 4\}$	SINR requirements of voice, video (low-bit-rate), video (high-bit-rate), web-browsing and data, respectively.
I_i, $k \in \{1, 2l, 2h, 3, 4\}$	Intercell interference of a voice, video (low-bit-rate), video (high-bit-rate), web-browsing and data service.
$I_{i,j}$, $k \in \{1, 2l, 2h, 3, 4\}$	Intercell interference of a voice, video (low-bit-rate), video (high-bit-rate), web-browsing and data service within the *i*th mobile user.
$I_{intercell}$	Intercell interference.
k	Number of packets that are transmitted in the channel conditioned on that there are *l* packets in a Pareto on period.
l	Number of packets in a Pareto on period.
$l_{i,k}$, $k \in \{1,2l,2h,3,4\}$	The instantaneous number of active spreading codes used by all voice, video (low-bit-rate), video (high-bit-rate), web-browsing and data services within the *i*th mobile user, respectively in the multi-connection system model.
l_k, $k \in \{1, 2l, 2h, 3, 4\}$	The instantaneous number of active spreading codes used by all voice, video (low-bit-rate), video (high-bit-rate), web-browsing and data services, respectively in the single-connection system model.
L_k, $k \in \{1,2,3,4\}$	Number of bits in voice, video, web-browsing and data packets, respectively.
λ	Transition rate from the off state to the on state of a high-bit-rate video minisource.

m_i	Number of retransmissions for the *i*th packet conditioned on that there are *l* packets in an on period.
M	Number of low-bit-rate spreading codes used by a video service
M_{re}	Maximum number of retransmissions.
M_{rek}, $k \in \{3,4\}$	Maximum number of retransmissions for web-browsing and data, respectively.
μ	Transition rate from the on state to the off state of a high-bit-rate video minisource.
$n_{i,k}$, $k \in \{1,2,3,4\}$	Number of voice, video, web-browsing and data services within the *i*th mobile user in a cell in the multi-connection system model.
n_{tr}	Number of transmissions that occur before the last packet in an on period arrives if this ob period contains *l* packets.
N	Number of mobile users in a cell.
N_k, $k \in \{1,2,3,4\}$	Number of voice, video, web-browsing and data services in a cell in the single-connection system model.
$N_{of}(l)$	Number of overflowed packets in the on period conditioned on that there are *l* packets in the on period.
$P_{loss,i,k}$, $k \in \{1,2l,2h,3,4\}$	Packet loss rates of voice, video (low-bit-rate), video (high-bit-rate), web-browsing and data, respectively within *i*th mobile user in the multi-connection system model.
$P_{loss,k}$, $k \in \{1,2l,2h,3,4\}$	Packet loss rates of voice, video (low-bit-rate), video (high-bit-rate), web-browsing and data, respectively in the single-connection system model.

system model.

$P_{loss,k}{}^{*}$, $k \in \{1, 2l, 2h, 3, 4\}$	Packet loss rate requirements of voice, video (low-bit-rate), video (high-bit-rate), web-browsing and data, respectively.
p_{onk}, $k \in \{1, 2l, 2h, 3, 4\}$	Activity factors of voice, video (low-bit-rate), video (high-bit-rate), web-browsing and data, respectively.
$p_{onk,c}$, $k \in \{3, 4\}$	Activity factors of web-browsing and data in the channel, respectively in the single-connection system model.
$p_{onk,i,c}$, $k \in \{3, 4\}$	Activity factors of web-browsing and data in the channel within *i*th mobile user, respectively in the multi-connection system model.
$P_{out,i,k}$, $k \in \{1, 2l, 2h, 3, 4\}$	Outage probabilities of voice, video (low-bit-rate), video (high-bit-rate), web-browsing and data, respectively within *i*th mobile user in the multi-connection system model.
$P_{out,k}$, $k \in \{1, 2l, 2h, 3, 4\}$	Outage probabilities of voice, video (low-bit-rate), video (high-bit-rate), web-browsing and data, respectively in the single-connection system model.
p_{re}	Retransmission probability of a non-real-time service.
$\psi_{j,k}$, $k \in \{1, 2l, 2h, 3, 4\}$	Instantaneous number of active spreading codes used by the *j*th voice service, active low-bit-rate and high-bit-rate spreading codes used by the *j*th video service, active spreading codes used by the *j*th web-browsing service and active spreading codes used by the *j*th data service, respectively in the single-connection system model.
$\psi_{i,j,k}$,	Instantaneous number of active spreading codes used by the *j*th

$k \in \{1, 2l, 2h, 3, 4\}$	voice service, active low-bit-rate and high-bit-rate spreading codes used by the *j*th video service, active spreading codes used by the *j*th web-browsing service and active spreading codes used by the *j*th data service within the *i*th mobile user in the multi-connection system model.
r_d	Distance between an intercell service and the intracell base station
r_m	Distance between an intercell service and its own base station
s	Number of packets that are transmitted during the acknowledgement time.
$S_{i,k}$, $k \in \{1, 2l, 2h, 3, 4\}$	Received powers of voice, video (low-bit-rate), video (high-bit-rate), web-browsing and data, respectively within *i*th mobile user in the multi-connection system model.
S_k, $k \in \{1, 2l, 2h, 3, 4\}$	Received power levels of voice, video (low-bit-rate), video (high-bit-rate), web-browsing and data, respectively in the single-connection system model.
σ^2	Variance of a Guassian random variable.
θ	Increased ratio of received power solution
$t_{arrival}$	Arrival time of a Pareto on period that contains *l* packets for a non-real-time service.
$t_{begin,i}$	Beginning transmission time of the *i*th packet within a Pareto on period that contains *l* packets.
$t_{finish,i}$	Finishing transmission time of the *i*th packet within a Pareto on period that contains *l* packets.

t_{off}	Length of the off period of a non-real-time service in the source.
$t_{off,c}$	Length of the off period of a non-real-time service in the channel.
t_{offk}, $k \in \{3,4\}$	Lengths of the off periods of web-browsing and data, respectively in the source traffic.
$t_{offk,c}$, $k \in \{3,4\}$	Lengths of the off periods of web-browsing and data, respectively in the channel.
t_{on}	Length of the on period of a non-real-time service in the source.
$t_{on,c}$	Length of the on period of a non-real-time service in the channel.
t_{onk}, $k \in \{3,4\}$	Lengths of the on periods of web-browsing and data, respectively in the source traffic.
$t_{onk,c}$, $k \in \{3,4\}$	Lengths of the on periods of web-browsing and data, respectively in the channel.
$t_{tr,i}$	Transmission time of the *i*th packet within a Pareto on period that contains *l* packets.
T	Packet duration of a non-real-time service.
T_a	Packet acknowledgement time of a non-real-time service.
T_k, $k \in \{1, 2l, 2h, 3, 4\}$	Packet durations of voice, video (low-bit-rate), video (high-bit-rate), web-browsing and data, respectively.
T_{ka}, $k \in \{3, 4\}$	Packet acknowledgement times of web-browsing and data, respectively.
W	Spread Spectrum bandwidth chip rate of the WCDMA system.

Chapter 1

Introduction

In the past few decades, mobile telecommunications have evolved into multiple cellular mobile systems. The first generation systems, such as the Advanced Mobile Phone System (AMPS) etc., are based on analog technology. They are only intended to carry voice messages between two users. The second generation systems, such as Global System for Mobile Communications (GSM), Personal Digital Cellular (PDC) and Interim Standard 95 (IS-95), are based on digital technology. They usually serve voice messages and sometimes low-bit-rate data communications, such as short messaging service (SMS). With the tremendous growth of a variety of traffic, a more advanced telecommunication system is needed to satisfy the enormous demand of future communications. Therefore, the Third Generation (3G) telecommunication system is proposed and developed. In contrast to the first generation and the current second generation telecommunication systems, the 3G system experiences many significant changes and improvements. It is intended to serve a wide range of multimedia communications and have many more advantages.

The 3G telecommunication system enables high-speed data communications, variable bit rate transmissions, a high spectrum efficiency, a good service quality, a worldwide roaming capability and multiple connections within a mobile user.

In the International Telecommunications Union (ITU), the third generation system is called International Mobile Telecommunications-2000 (IMT-2000). Particularly, it is named Universal Mobile Telecommunications System (UMTS) in Europe. Besides UMTS, the two 3G systems are CDMA2000 and TD-SCDMA. They are proposed in the United States and China, respectively. In this thesis, our studies are focused on the UMTS network.

In the UMTS network, the wideband Code Division Multiple Access (WCDMA) is commonly referred to as the multiple access technology. WCDMA technology is able to support a high bit rate of over 384 kbps in most environments and over 2 Mbps in good conditions. Such a high data bit rate facilitates a lot of new applications like audio, video, file downloading, and Internet surfing.

In order to complete the detailed standardization of UMTS, the Third Generation Partnership Project (3GPP) is established to produce globally applicable technical specifications. At the same time, a lot of efforts are made on developing protocols and algorithms to solve various practical problems in the UMTS network. For instance, Quality of Service (QoS) provisioning in the UMTS network is a critical area that attracts a lot of interests and leaves plenty of room for further studies. The UMTS network accommodates many multimedia services that differ a lot in terms of their QoS requirements. As an effort to study the area of QoS provisioning, the objective of this thesis is to investigate the QoS issues in the UMTS network. In our studies, we will first present the issues and problems of QoS in the UMTS network, and then propose appropriate system models for the network, and finally analyze the QoS performances of various services. In the following sections of this chapter, we introduce the basic Quality of Service issues, the aims of our research, and the organization of the subsequent chapters in this thesis.

1.1 Basic QoS Issues

QoS indicates the level of the performance that the system needs to guarantee during the whole duration of a service in UMTS. The UMTS network can be divided into multiple layers, such as the physical layer, the data link layer, the network layer and some other higher layers according to the functionalities. Each layer is in charge of some functions for a service. These layers jointly fulfill the QoS guarantees in the UMTS network. Some QoS parameters are considered to quantify the QoS performances in each layer. Thus, the detailed QoS architecture of the UMTS is presented in [12] and the QoS provisioning in UMTS is subject to the simultaneous satisfaction of the QoS constraints in all layers.

The QoS parameters at the physical layer include Bit Error Rate (BER) of a service.

The QoS parameters at the data link layer include signal-to-interference-plus-noise ratio (SINR) and the outage probability of a service.

The QoS provisioning at the network layer mainly includes two parts: call level and packet level. The call level QoS parameters usually consist of blocking probabilities of new and handoff services and forced termination probability of handoff services. The packet level QoS parameters consist of average delays and packet loss rates.

Furthermore, a call admission control (CAC) algorithm can be developed based on QoS provisioning. CAC is a process that decides whether a network can admit a new service, while still satisfies the QoS requirements of all existing services in the network. CAC is used to determine the admission region (AR) of the network. An efficient CAC algorithm may widen the AR, increase the system capacity and thus maximize the operation profit.

1.2 Previous Works

The main work of this thesis is to analyze the QoS performances in the uplink of the wideband CDMA system in the UMTS network. Many literatures have sufficiently introduced the UMTS network and the WCDMA multiple access technology. For example, [1-6] give a comprehensive description of the basic principles of UMTS and WCDMA. Besides, 3GPP provides detailed standards of the whole UMTS network. Compared to the second generation telecommunication systems, the UMTS network extends the QoS provisioning of current voice service to multiclass services. WCDMA is chosen as the multiple access technology in the wireless channel of the UMTS network.

Before the emergence of WCDMA, a lot of efforts have been made on studying the QoS performances of DS-CDMA systems in the past decades. In [7, 47-50], the delay and throughput performance of a DS-CDMA network are analyzed for voice and data services. Poisson processes are assumed for both voice and data traffic in [7, 47]. In [48-50], an exponential on/exponential off process and a Poisson process are assumed for voice and data traffic, respectively. In [8], a method is presented to accommodate the voice and data services simultaneously. A voice service is modeled as an exponential on/exponential off process, while a data service generates a packet randomly in each slot with a certain probability in their traffic models. Markov chains are used to solve for the average delays and packet loss rates of each service. In [9], a medium access control (MAC) layer protocol for a DS-CDMA system is proposed to provide the QoS guarantees for multiclass services in a wireless network. For each service, the packet arrival process is Poisson distributed and a Markov chain model is developed to derive the average delays. In [10], the author considers a DS-CDMA system that supports multiclass services. Forward error correction (FEC) method and automatic retransmission request (ARQ) mechanism are implemented to achieve fewer errors. All services are modeled as Poisson processes. This paper investigates the SINR, the average delay and the BER characteristics of each service.

In [11], the QoS performances are evaluated in the UMTS network. The authors develop a MAC protocol for voice, video and data services in the UMTS network. The voice service is modeled as an exponential on/exponential off process, the video service is approximated by Maglaris' model with a one-dimensional Markov chain [13], and the data service is modeled as a Poisson process. This paper studies the packet loss rate and the average delay for each service in the framework of its MAC protocol. For the voice service, analytical results are obtained in terms of average delay and the packet loss rate. However, only computer simulation results are available for video and data services in terms of average delay and the packet loss rate.

The above works on QoS usually adopt simple traffic models, such as exponential on/exponential off process for voice, one-dimensional Markov chain for video and Poisson process for data in [7-11] and [47-50]. At the same time, an infinite buffer is implemented for data in [10], which is not a realistic assumption in practice. Besides, no analytical results are given for video and data services in [11].

The call admission control (CAC) issue for DS-CDMA is addressed in [45-46]. However, the CAC schemes in [45-46] are simply SINR threshold based and cannot guarantee the QoS levels, such as packet loss rate and delay, of all the admitted services.

In [67], the system capacity of a DS-CDMA with voice and data services is evaluated on the satisfaction of outage probability for voice and delay for data. However, it only considers two classes and adopts simple traffic models by randomly generating voice and data packets.

The main contribution in this thesis is to analyze the QoS performances of four traffic classes in the uplink of a multi-cell WCDMA system. Our analysis is based on more realistic traffic models and a finite buffer for packet retransmissions. Furthermore, a CAC method is described on our QoS analytical platform. This method differs from [67] as we extend the CAC scheme by using more realistic traffic models and supporting four classes.

1.3 Aims of Thesis

As we have introduced in section 1.1, the QoS provisioning is jointly fulfilled in different layers of the UMTS network. In this thesis, our work is mainly focused on the packet level QoS performances at the network layer. Within the whole UMTS network, we focus our analysis on the wireless network and the uplink of the WCDMA system. The packet level QoS performances, such as the packet loss rate and the average delay, are evaluated for multiclass services in the uplink. The following issues are the key interests in our studies and presented in greater details.

- Firstly, we emphasize the analysis of the uplink. The uplink refers to the reverse link from the mobile users to the base stations via the wireless channel. The multiple access interference (MAI) in the WCDMA system is more severe in the uplink than that in the downlink, which refers to the forward link from the base stations to the mobile users. Thus, the uplink is the focus of our analytical emphasis.

- Outage is an important QoS concept that indicates that the achieved system does not achieve the required performance in the data link layer. The outage probability is a measurement to define the level of outage and is referred to as the portion of time that the achieved SINR is below the SINR requirements or the achieved BER is above the BER requirements in the WCDMA system.
- Delay is a QoS parameter at the packet level in the network layer. A packet is usually required to be successfully received by the destination within a certain time. Within the WCDMA system, delay refers to the period between the instant when a packet is generated and the instant when it is successfully received.
- Packet loss occurs in the wireless channel between the mobile users and the base stations in the WCDMA system. The packet loss rate is a parameter to accurately quantify the level of the packet loss.

Furthermore, the system capacity is also addressed in terms of call admission region in this thesis. Our system capacity is based on QoS requirements in terms of the packet loss rate and delay. Regarding this issue, the High Data Rate (HDR) algorithm is proposed in [66] by Qualcomm as an approach to achieve a high capacity in a CDMA system, especially in the downlink. In the HDR algorithm, each mobile user is allowed to measure the received SINR from multiple base stations. The base station with the highest SINR is selected so that the interference to the users in other base stations is reduced. In addition, error-correcting coding techniques are implemented to data users with low SINR to suppress interference but result in longer delay. HDR scheme optimizes the packet transmissions and achieves a high throughput by allocating different delays to users with different SINR values and data rates. In [66], the throughput of the HDR system is presented by simulations and measurements under particular coding and modulation techniques. Similar to other existing papers, SINR is the main factor in determining its system capacity. HDR first measures the received SINR and estimates the supportable data rates, followed by optimizing the packet transmissions though appropriate delay allocation to each user. Because this process involved signaling, measurements and prediction, it is not easy to be analyzed. Our thesis has a different contribution compared to [66] since we focus more on provisioning of QoS analytically.

On the other hand, scheduling is a discipline that can allocate resources to different connections and decide the service order. It allows connections to share the resources and provides performance guarantee. For example, wireless weighted fair queuing (WFQ) [63-65] is a scheduling method used in wireless networks. Discussion on scheduling is beyond the scope of this thesis.

1.4 Thesis Organization

In my thesis, the packet level QoS performances are investigated. The subsequent chapters are organized as follows.

Chapter 2 introduces the architecture of the UMTS network. The UMTS network can be divided into multiple subsystems. We discuss the functions of each subsystem. The UMTS radio access network (UTRAN) is emphasized and wideband CDMA is explained. In addition, Chapter 2 also deals with the Quality of Service architecture of the UMTS network. The definitions of the UMTS QoS classes are given. Four different classes are classified. Voice, video, web-browsing and data services are chosen as typical examples of these four classes. Furthermore, traffic models are established for each of them to proceed with further analyses.

Chapter 3 explains the Go-Back-N (GBN) automatic retransmission request (ARQ) that is used in the WCDMA system. GBN ARQ is used to retransmit erroneous as an effort to improve the transmission reliability of the interactive and background classes. The QoS performances are analyzed in the GBN ARQ system. The analytical results in Chapter 3 are referred to in the subsequent chapters.

In Chapter 4, we first propose two appropriate system models, which are the single-connection system model and the multi-connection system model. In the former, each mobile user can only have one connection. In the latter, each mobile user can have multiple connections within different traffic classes. Then the medium access control (MAC) and the radio link control (RLC) methods are introduced. We derive the outage probability of each service in the WCDMA system according to the two system models and the MAC/RLC schemes

Chapter 5 evaluates the packet level QoS performances for voice, video, web-browsing and data services respectively in both system models. The packet loss rate and average delay are analyzed and formulated mathematically for each service.

Chapter 6 provides the numerical results of the QoS performances that are developed in both Chapter 4 and Chapter 5 for the two system models. Simulations are used to verify the accuracy of the analytical results that are derived. At the same time, a QoS-based call admission control scheme is described and discussed, and admission regions are derived at the packet level of the network layer for the two system models.

Finally, Chapter 7 concludes the thesis and introduces future works.

Chapter 2

UMTS Networks and QoS Architecture

As one of the proposals for Third Generation systems, Universal Mobile Telecommunications System (UMTS) shows advantages in many aspects. In the scope of this chapter, we present a basic overview of UMTS. At the same time, the Quality of Service architecture in the UMTS network is introduced in greater details. On the basis of the UMTS QoS classes' classifications, we describe a traffic model for each of them. The contents in this chapter are generalized as follows.

In section 2.1, we give a brief description of the UMTS network and evaluate the functions of each subsystem. In section 2.2, we present the main principles of the wideband CDMA technology as the air interface of the wireless channel. In section 2.3, the QoS classes are classified in the UMTS network. In section 2.4, we define a traffic model for each QoS class to facilitate further analyses. In section 2.5, the conclusion is given for this chapter.

2.1 UMTS Framework

UMTS is the European version of the Third Generation (3G) mobile communication system. The architecture of the UMTS network is given in [1]. Functionally, the UMTS network has three subsystems to address different operations. The subsystems consist of UMTS Terrestrial Random Access Network (UTRAN), Core Network (CN) and User

Equipment (UE). UTRAN is responsible for all radio access procedures. CN is responsible for switching and routing of services and connects external networks. UE refers to the user equipment and interfaces with the UTRAN.

In [1], the system architecture of the UMTS network is illustrated as Figure 2.1:

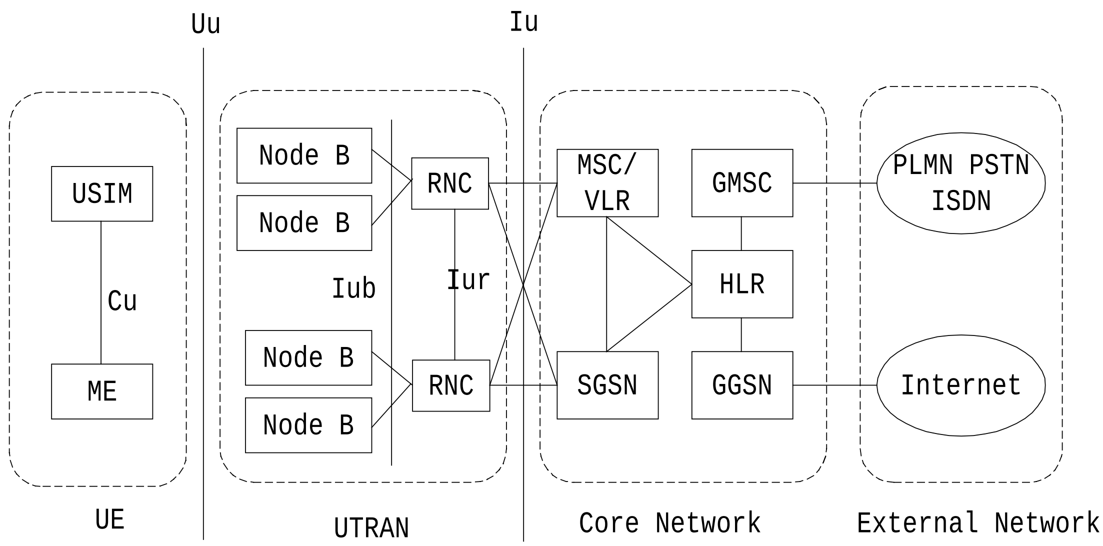

Figure 2.1 UMTS Architecture

UE contains two parts such as the Mobile Equipment (ME) and the UMTS Subscriber Identity Module (USIM) as shown in Figure 2.1. ME is referred to as a mobile user. The USIM is referred to as a smart card that stores the identity, authentication, encryption keys and other user information of the subscriber.

UTRAN is the radio access network of UMTS. It consists of Nodes B and radio network controllers (RNCs). A Node B is a base station transceiver. It is responsible for one or more cells. A Node B exchanges signals with a number of UEs and communicates with RNCs. Its functions also include transmitting the system information, making error detection and correction, finishing channel coding and performing radio resource management, etc. Besides the Nodes B, the RNC is also a controlling element in the

UTRAN. Its functions cover the management of the Nodes B, the system information control, the scheduling of system information and the call admission control, etc. Generally, RNC can control one or a few Nodes B.

CN is another subsystem of the UMTS network. It contains the Mobile Services Switching Center (MSC), Visitor Location Register (VLR), Home Location Register (HLR), Gate MSC (GMSC), Serving GPRS (General Packet Radio Service) Support Node (SGSN), and Gateway General Packet Radio Service Support Node (GGSN). The functions of these entities are given as follows.

- MSC is operated to serve the circuit-switched data. Its functions include paging, dynamic resource allocation and handover management, etc. The MSC serves all circuit-switched flows. In the RNC, circuit-switched data streams are forwarded to the MSC.
- VLR cooperates with the MSC. It works as a database and stores information about roaming mobile users in the MSC area. One VLR may handle the visitor register of several MSC areas.
- HLR works as a database located in the home system of a subscriber and keeps the service profile of the subscriber.
- SGSN provides the functionality that is similar to the MSC/VLR but the SGSN is used for the packet-switched services instead of the circuit-switched services. From RNC to CN, all packet-switched data streams are forwarded to SGSN.
- GMSC refers to a switching gateway that connects the UMTS network to external circuit-switched networks. All circuit-switched data streams between external networks and internal networks must go through the GMSC.

- GGSN is similar to GMSC except that it is a switching gateway that connects the UMTS network to external packet-switched networks. Packet-circuit data streams between external networks and internal networks must pass through the GGSN.

External networks are the networks outside the UMTS network. They can be divided into two types. The Public Switched Telephone Network (PSTN) is used for the transmissions of circuit-switched services, while the Internet Protocol (IP) network is used for the transmission of packet-switched services.

UE, UTRAN and CN cooperate to fulfill the functionality of the UMTS network. The three subsystems are connected together with various interfaces. The Iu interface connects the CN and the UTRAN in the UMTS network. The Cu interface is an electrical interface between the USIM and the ME. The Uu interface is a wideband radio interface, through which the UE accesses the UTRAN. Iur interface links two RNCs and permits soft handover between RNCs. The Iub interface connects a Node B to the RNC. Thus, we can see clearly that we mainly study the QoS performances in the UTRAN and the Uu air interface is our subject of interest.

Through the Uu air interface, WCDMA is chosen as the multiple access technology of the UTRAN. In the next section, we introduce the characteristics of the WCDMA technology.

2.2 Wideband CDMA Air Interface

As a subsystem of the UMTS network, the UTRAN is responsible for wireless access and the radio resource management in the UMTS network. The UTRAN encompasses two modes: Frequency Division Duplex (FDD) and Time Division Duplex (TDD). In the FDD mode, the uplink and the downlink use separate frequency bands and wideband

CDMA is selected as the radio access technology. In the TDD mode, both the uplink and the downlink use the same frequency band and TD/CDMA is selected as the radio access technology. In this thesis, all our studies are based on the FDD mode and the WCDMA technology. In the following, we will introduce the principles of WCDMA in four aspects.

2.2.1 WCDMA Basic Concept

WCDMA is a wideband Direct Sequence Code Multiple Access (DS-CDMA) technology. It is proposed as the multiple access technology in the FDD mode of the UTRAN system.

In comparison with the general DS-CDMA systems that have been deployed in the second generation systems, such as IS-95A/B, WCDMA is characterized by a wide bandwidth of 5 MHz and a constant high chip rate of 3.84 Mcps. The wideband frequency is chosen because it can provide a high data rate of 144 kbps to 384 kbps and even 2 Mbps in good conditions. The wide bandwidth of the spread spectrum system resolves more multipaths problems and thus improves the system performance. In addition, the WCDMA features also include a fast power control in both the uplink and the downlink and the capability to vary the data rates and the system parameters during the connection time of a service.

2.2.2 Spreading and Scrambling

Spreading and scrambling are two important procedures in the WCDMA system. In the uplink of the WCDMA system, before the information data is transmitted out from mobile users, it must be multiplied with both the spreading codes and the scrambling codes. Within a mobile user, a service can be transmitted through a Dedicated Channel

(DCH), which is identified by a spreading code. Spreading codes with various spreading gains can enable different data rates and separate different DCHs. The chip rate of all DCHs is the same through the spreading and is equal to 3.84 Mcps. The signal bandwidth is extended to 5 MHz. Next, all services from the same mobile user are multiplied by a common code that is called the scrambling code. This process is named as scrambling. The scrambling does not change the bit rate of each service and does not increase the transmission bandwidth. The usage of the scrambling is to separate different mobile users in the uplink.

Similarly, all services in the downlink also experience both the spreading and the scrambling. The spreading provides multiple choices of the data rates for services. All services from the same base station are multiplied by a common scrambling code. Different from the usage in the uplink, the usage of the scrambling in the downlink is to separate signals from different base stations. The spreading and the scrambling are illustrated in Figure 2.2.

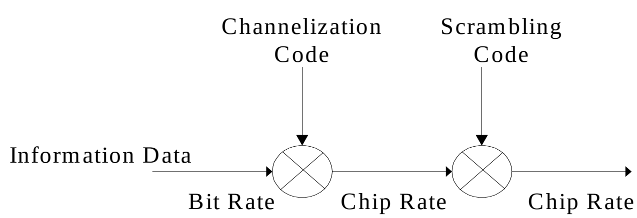

Figure 2.2 Spreading and Scrambling

The input signals must be correlated with replicas of both the spreading codes and the scrambling codes to obtain the desired data information at the receiver of a base station or a mobile user.

According to [1], the spreading codes in the WCDMA system are based on Orthogonal Variable Spreading Factor (OVSF) technique in both the uplink and the downlink. The scrambling codes can be a kind of long code that is called the Gold code.

2.2.3 Modulation and Channel Coding

A modulation scheme defines how the data bits are mixed with the carrier signals, which is usually a sine wave. Generally, there are three basic ways to modulate a carrier signal in a digital sense. They are amplitude shift keying (ASK), frequency shift keying (FSK), and phase shift keying (PSK). The Quadrature Phase Shift Keying (QPSK) modulation is adopted in the WCDMA system.

As the wireless transmissions are unreliable, the channel coding is usually implemented for a service that is sensitive to errors. The channeling coding is a method of adding redundancy to the information signals. The channel coding is to improve the quality of reliable communications. When the packets are transmitted over a noisy channel to the destination, errors can be checked and corrected through channel coding. In [33], two types of channel coding methods are defined for services. The first is convolutional coding. The coding gain is selected as either $\frac{1}{2}$ or $\frac{1}{3}$. The other type of coding method is Turbo coding. Its coding gain is fixed at $\frac{1}{3}$.

2.2.4 Radio Resource Management

Radio Resource Management (RRM) is responsible for the utilization of all radio resources in the WCDMA system. The aim of RRM is to guarantee the QoS of each service in the system as well as offering high capacity in networks. The functions of

RRM cover call admission control (CAC), power control (PC), load control (LC) and scheduling.

The network is supposed to provide a high system capacity and to satisfy the QoS requirements of all services in the WCDMA system. The QoS performances are regarded as the criteria for CAC. Therefore, the call admission control is required to determine admission of a new call based on QoS satisfactions. An efficient CAC scheme has to admit as many mobile users as possible and to guarantee their QoS performances simultaneously.

The performances of the WCDMA system are directly associated with the power levels of all mobile users in the system. Therefore, an efficient power control method needs to be deployed. The power information of all mobile users should be exchanged between users and base stations quickly and the power levels should be adjusted dynamically.

Besides, RRM also includes other areas, such as scheduling, load control (LC). Detailed descriptions of these areas are available in [34].

2.3 UMTS QoS Class

Future modern telecommunication networks, such as UMTS, are expected to support a wide variety of multimedia services including speech, audio, video, image, text, and data. For each multimedia service, the user must specify a set of parameters to characterize its service performances that the network is able to provide over the duration of the connection. Such parameters are called the Quality of Service (QoS) attributes and they quantify the end-to-end network performance for a specific service. Therefore, a major issue in deploying the UMTS network is to guarantee QoS requirements for all services simultaneously.

According to the definition of 3GPP, the network services are based on end-to-end connections. A communication is established from an end user to another end user across the UE, UTRAN, CN and external networks. An end user may have a certain QoS requirements. Based on [1] and [12], the network establishes a bearer service with clearly defined characteristics and functionalities for two end users in order to satisfy the required QoS levels. A bearer service refers to a basic telecommunication service that offers the transmission capability of signals between two end users. A bearer service can either be packet-switched or circuit-switched. Its characteristics include the traffic type, the transmission information and the supported bit rate. A bearer service covers all aspects that are related to the provisioning of QoS. These aspects consist of the signaling and the QoS management functionality, etc. 3GPP proposes the architecture of a bearer service that can be shown as a tree structure as follows [1].

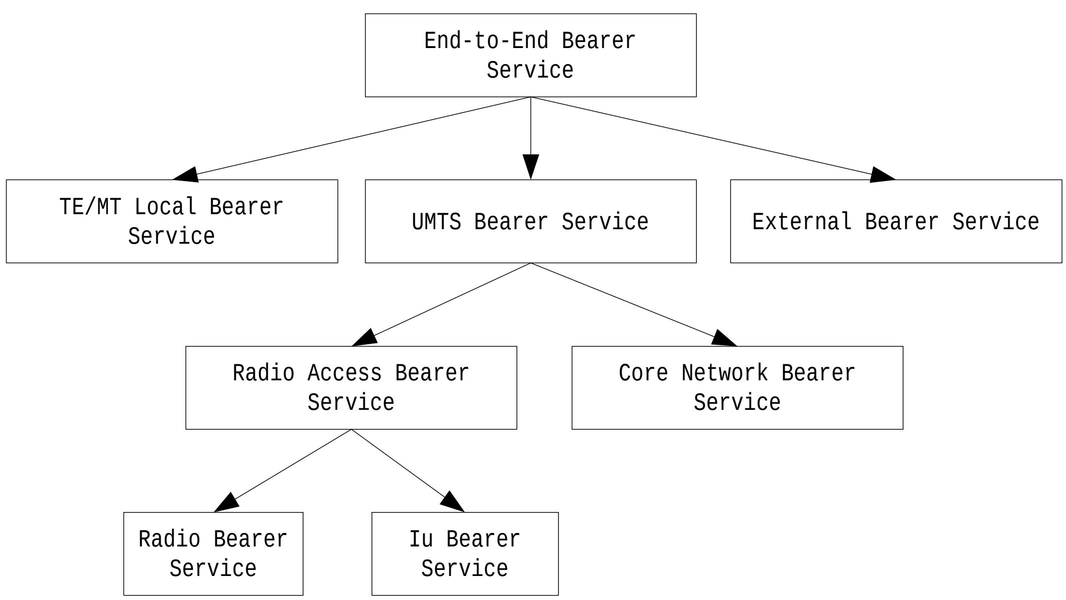

Figure 2.3 Architecture of a Bearer Service

Multiple sub-bearers may jointly complete the functions of an end-to-end bearer service, as illustrated in Figure 2.3. The sub-bearers include the Terminal Equipment/Mobile Terminal (TE/MT) local bearer service, the UMTS bearer service and external bearer services. Each sub-bearer is responsible for different parts of an end-to-end network. Since we are studying the QoS performances in the UMTS network, our interests are focused on the UMTS bearer service.

A UMTS bearer service is further composed of two parts: radio access bearer service and core network bearer service. The radio access bearer service is based on the radio access interface and provides the transport of signaling and the user data between mobile users and the UTRAN with adequate QoS levels. The core network bearer service refers to the service in the wireline channel of the UMTS network and connects the core network with external network with the required QoS constraints.

In [12], the UMTS QoS classes are defined for all kinds of UMTS bearer services. A framework of four traffic classes is established for all UMTS bearer services. The four traffic classes are named as conversational class, streaming class, interactive class and background class. The characteristics of each class are described in the following [12].

2.3.1 Basic Classes

1. Conversational Class

   The conversational class includes bi-directional, symmetric and real-time services. A typical example of this class is telephony voice. With the advent of Internet services and IP networks, voice over IP (VoIP) and video conferencing are also covered by this class. Real-time services are characterized by a low transfer time of a packet. The maximum transfer delay bound is strictly restricted. Failure to satisfy such a bound results in an unacceptable quality.

2. Streaming Class

   The real-time video service is known as a typical application of this class. It is a unidirectional and asymmetric service. The receiver can present a video service to a user during the period of transmission. Additionally, the streaming class requires a looser transfer delay than that of the conversational class and a limited level of packet loss rate.

3. Interactive Class

   The interactive class is a kind of best effort class and usually refers to the case that a user is retrieving data from a remote host. Examples of this class can be web- browsing, data retrieval and server access. The interactive class is highly asymmetric. Only one direction is used to transmit traffic and the other direction is mostly used for signaling. As a non-real-time traffic, the interactive class is not sensitive to delay, while it tolerates fewer transmission errors and a lower packet loss rate than those from real-time classes.

4. Background Class

   The background class is also a kind of best effort class and consists of services that do not have any precise delay requirements. It is also a non-real-time traffic and does not expect to obtain data within a certain time. Similar to the interactive class, the background class is very asymmetric. Email, SMS, Multimedia Messaging Service (MMS), facsimile and download of files are typical examples of this class. In contrast to other classes, the background class requires the lowest packet loss rate during transmission.

In general, these four classes are distinguished based on their different delay and packet loss requirements. The above classes are listed from the most delay-sensitive services to the least delay-sensitive services. The conversational class and the streaming class are usually regarded as real-time services, which require stringent transfer delay. The interactive class and the background class are considered as non-real-time services, which have no delay constraints but require strictly low packet loss rates.

2.3.2 QoS Attributes

The QoS attributes of a UMTS bearer service describe the QoS parameters provided by the UMTS system to the user of the UMTS bearer service. A UE can request specific QoS attributes from the network at the establishment of a UMTS bearer service. The QoS attributes determine the constraints for all QoS classes. In this thesis, the main QoS attributes include BER, delay and packet loss rate of each class. In [12], an introduction of the QoS attributes of the four QoS classes is given in greater details.

2.4 Traffic Models

In the previous sections, four QoS classes are presented. In order to analyze the QoS performances of each class mathematically, traffic models are required to be established to characterize these classes. This section deals with the traffic modeling problem. Traffic modeling constitutes the important step towards understanding and solving QoS-related issues in the UMTS network. The central idea of traffic modeling is to construct appropriate traffic models that describe important statistical properties of different classes.

Traffic models also influence multiple access control method and resource allocation method in the network. As introduced in section 2.3, QoS classes in the UMTS network are classified as conversational class, streaming class, interactive class and background class. In our analyses, voice, video, web browsing, data services respectively are chosen as their typical representatives. In this section, the traffic models are presented and analyzed.

2.4.1 Voice Model

According to [20], a voice service is usually modeled as an exponential on/exponential off process. Packets are transmitted at a fixed rate during the on state, while no packet is transmitted during the off state. The Markov chain for such a source is illustrated as Figure 2.4.

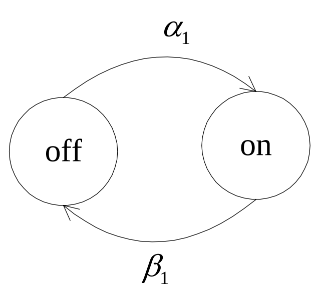

Figure 2.4 Traffic Model of a Voice Service

α_1 means the transition rate from the off state to the on state and β_1 means the transition rate from the on state to the off state. Let the activity factor, which is the probability that a voice service is in the on state, be p_{on1} and the following holds.

$$p_{on1} = \frac{\alpha_1}{\alpha_1 + \beta_1} \tag{2.1}$$

A single spreading code is assumed to be assigned to a voice service during the on period and is assumed to be withdrawn by the base station during the off period in the WCDMA system.

2.4.2 Video Model

Video is a continuous variable bit rate (VBR) traffic and its rate varies over time. In [13], Maglaris proposes a one-dimensional Markov chain to describe a video service. According to [13], a video source can be approximated by a number of identical exponential on/off minisources. However, this model is usually suitable for a video service with a low data rate. In [14], Sen improves the traffic model proposed in [13]. In [14], the data rate of video traffic is approximated by the combination of multiple identical low-bit-rate on/off minisources and a high-bit-rate on/off minisource. The low-bit-rate on/off minisources have the same activity factors, which is different from the activity factor of the high-bit-rate on/off minisource. As given in [14], a 2-dimensional Markov chain is established to approximate the variation of the bit rate as follows.

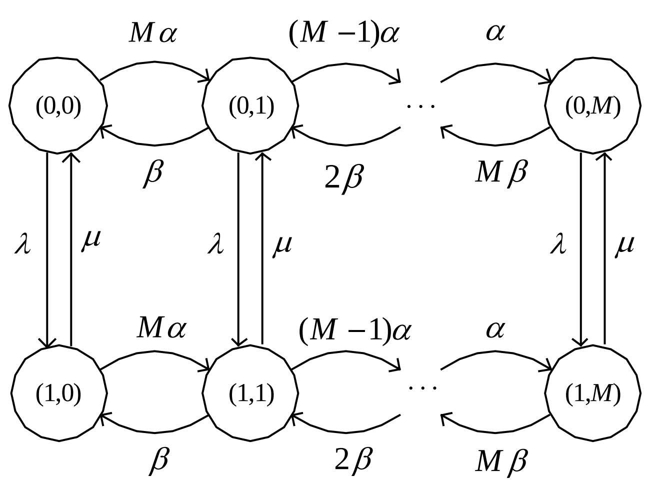

Figure 2.5 Traffic Model of Video Services

Based on Figure 2.3, a number of spreading codes, including one high-bit-rate spreading code and M low-bit-rate spreading codes, are assumed to be assigned to a video service to satisfy its varying rate requirements in the WCDMA system. When the instantaneous bit rate is varying, the number of the required spreading codes is varying too. Each low-bit-rate on/off minisource is identified by a low-bit-rate spreading code and the high-bit-rate on/off minisource is identified by a high-bit-rate spreading code.

A low-bit-rate on/off minisource can be modeled as an exponential on/exponential off process as shown in Figure 2.6.

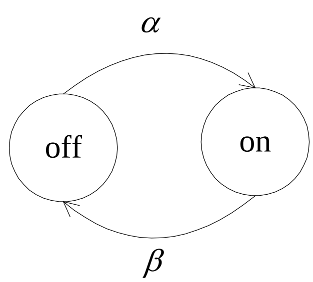

Figure 2.6 Low-bit-rate on/off Minisource of a Video Service

Let α denote the transition rate from the off state to the on state and let β denote the transition rate from the on state to the off state. So the activity factor p_{on2l} of a low-bit-rate on/off minisource is given by equation (2.2).

$$p_{on2l} = \frac{\alpha}{\alpha + \beta} \tag{2.2}$$

The high-bit-rate on/off minisource can be also modeled as an exponential on/ exponential off process as illustrated in Figure 2.7.

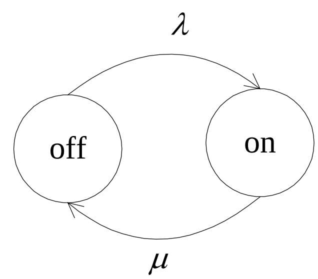

Figure 2.7 High-bit-rate on/off Minisource of a Video Service

Let λ denote the transition rate from the off state to the on state and let μ denote the transition rate from the on state to the off state. So the activity factor p_{on2h} of the high-bit-rate on/off minisource is given by equation (2.3).

$$p_{on2h} = \frac{\lambda}{\lambda + \mu} \tag{2.3}$$

Based on the Markov chain in Figure 2.4, suppose that the bit rate state (*i, j*) indicates a bit rate level in which a video service uses *j* low-bit-rate spreading codes and *i* high-bit-rate spreading codes, then the state probability at (*i, j*) can be expressed as $p_{i,j}$.

$$p_{i,j} = \binom{1}{i}\left(\frac{\lambda}{\lambda+\mu}\right)^{i}\left(1-\frac{\lambda}{\lambda+\mu}\right)^{1-i}\binom{M}{j}\left(\frac{\alpha}{\alpha+\beta}\right)^{j}\left(1-\frac{\alpha}{\alpha+\beta}\right)^{M-j} \tag{2.4}$$

2.4.3 Web-Browsing Model

Web-browsing is a kind of non-real-time service. According to [68], one web-browsing service can be modeled as a Pareto on/Pareto off process. Packets are transmitted at a fixed rate during the on state, while no packet is transmitted during the off state. The on/off process is not Markovian, as the on and off periods are Pareto-distributed. However, its states and transition rates can be approximately illustrated in Figure 2.8.

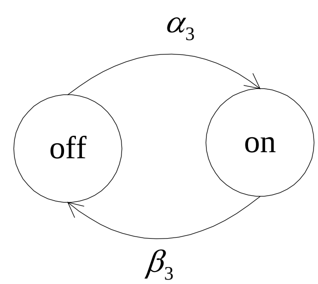

Figure 2.8 Traffic Model of Web-browsing Services

Let α_3 denote the transition rate from the off state to the on state and let β_3 denote the transition rate from the on state to the off state, respectively. For a Pareto on/ Pareto off source, the lengths of its on and off states are denoted by t_{on3} and $t_{off\,3}$, respectively. The probability density function of t_{on3} is denoted by $f_3(t)$ and the mean of t_{on3} is denoted by $E[t_{on3}]$, which are given by equations (2.5) and (2.6), respectively [19].

$$f_3(t) = c_{3,on} a_{3,on}{}^{c_{3,on}} t^{-c_{3,on}-1},\ t \geq a_{3,on}, \tag{2.5}$$

where $c_{3,on}$ is the shape parameter and $a_{3,on}$ is the location parameter.

$$E[t_{on3}] = \frac{1}{\beta_3} = \frac{c_{3,on} a_{3,on}}{c_{3,on} - 1} \tag{2.6}$$

The length of its off state is denoted by $t_{off\,3}$. We assume that the probability density function and the mean of $t_{off\,3}$ are given by

$$g_3(t) = c_{3,off} a_{3,off}{}^{c_{3,off}} t^{-c_{3,off}-1},\ t \geq a_{3,off}\ , \tag{2.7}$$

where $c_{3,off}$ is the shape parameter and $a_{3,off}$ is the location parameter,

and
$$E[t_{off\,3}] = \frac{1}{\beta_3} = \frac{c_{3,off} a_{3,off}}{c_{3,off} - 1} \tag{2.8}$$

The activity factor of a web-browsing service is denoted by p_{on3} and is given by

$$p_{on3} = \frac{E[t_{on3}]}{E[t_{on3}] + E[t_{off3}]}. \tag{2.9}$$

Just like voice services, a single spreading code is assigned to a web-browsing service during its on state and is withdrawn during its off state in the WCDMA system.

2.4.4 Data Model

Data is also a kind of non-real-time service. Similar to a web-browsing service, one data service is modeled as a Pareto on/Pareto off process. The on/off process of data service is not Markovian and can be approximately as shown in Figure 2.9.

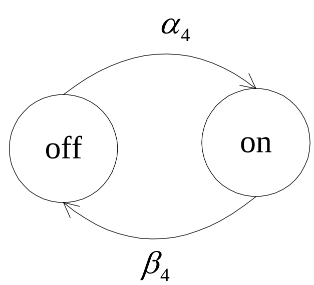

Figure 2.9 Traffic Model of Data Services

Let α_4 denote the transition rate from the off state to the on state and let β_4 denote the transition rate from the on state to the off state, respectively. For a Pareto on/Pareto off source, the lengths of its on and off states are denoted by t_{on4} and t_{off4}. The probability density function and mean of t_{on4} are denoted by $f_4(t)$ and $E[t_{on4}]$, which are given by equation (2.10), and (2.11), respectively [19].

$$f_4(t) = c_{4,on} a_{4,on}{}^{c_{4,on}} t^{-c_{4,on}-1},\ t \geq a_{4,on} \tag{2.10}$$

where $c_{4,on}$ is the shape parameter and $a_{4,on}$ is the location parameter.

$$E[t_{on4}] = \frac{1}{\beta_4} = \frac{c_{4,on} a_{4,on}}{c_{4,on} - 1} \tag{2.11}$$

The length of its off state is denoted by $t_{off\,4}$. We assume that the probability density function and the mean of $t_{off\,4}$ are given by

$$f_4(t) = c_{4,off}\, a_{4,off}{}^{c_{4,off}}\, t^{-c_{4,off}-1},\; t \geq a_{4,off} \tag{2.12}$$

where $c_{4,off}$ and $a_{4,off}$ are the shape parameter and location parameter,

and
$$E[t_{on4}] = \frac{1}{\beta_4} = \frac{c_{4,off}\, a_{4,off}}{c_{4,off}-1} \tag{2.13}$$

The activity factor p_{on4} of data service is expressed as follows.

$$p_{on4} = \frac{E[t_{on4}]}{E[t_{on4}] + E[t_{off\,4}]} \tag{2.14}$$

Similar to voice and web-browsing service, a single spreading code is assumed to be assigned to a data service during its on state and is withdrawn by the base station during its off state.

In summary, the traffic models of all service classes used in this thesis are given. Exponential on/exponential off process and two-dimensional Markov chain are used to model the voice and video traffic, respectively. These two models are adopted by many existing works, such as [14], [20] and [22]. For web-browsing and data, we adopt Pareto on/Pareto off model, which is proposed in [68]. Pareto on/Pareto off model is regarded as more realistic than exponential on/exponential off model in approximating the statistics of web-browsing and data, because these two services have a heavy-tailed probability density function for its on and off periods. In the next few chapters, we will obtain the QoS attributes using this more realistic traffic model.

2.5 Conclusion

We generalize the UMTS network architecture and the main characteristics of the WCDMA technology in this chapter. At the same time, this chapter gives an introduction of the QoS architecture in the UMTS network. According to four different QoS traffic classes, we select typical applications for these classes and define appropriate traffic models for each of them. We will analyze QoS performances based on these traffic classes in the following chapters.

Chapter 3

Analysis of Go-Back-N ARQ

Packets are transmitted from mobile users to base stations over the wireless channels in the uplink of a WCDMA system. In contrast to wireline communications, wireless communications are not reliable and are prone to fluctuations of the channel conditions. Multiple access interference (MAI) deteriorates the packet transmissions and destroys the packets in the WCDMA system, even if ideal power control is used in the system. The information bits within a packet may become erroneous during the transmissions due to MAI. A specific level of bit error rate (BER) requirement is defined in 3GPP technical specification [12]. When the achieved BER within a packet is above the required level, the corresponding packet is considered as erroneous. With regard to the four traffic classes presented in Chapter 2, their erroneous packets are treated differently. The base station first checks whether the received packets are correct or erroneous in the uplink of a WCDMA system. For real-time classes, such as the conversational class and the streaming class, the correct packets are forwarded from the base station to their destinations, while the erroneous packets are discarded directly as packet loss. Due to the delay-sensitive constraint of real-time classes, no automatic retransmission request (ARQ) mechanism is initiated in case of errors. Mobile users merely transmit packets to the base stations one by one in an orderly manner. Forward Error correction (FEC) schemes are appropriate for real-time traffic. However, for non-real-time classes, such as the interactive and the background classes, the operations are more complicated. Since non-real-time classes are non-delay-sensitive and require much more stringent packet loss rate constraints, an appropriate ARQ mechanism is necessary for them. It is necessary that each transmitted packet of a non-real-time class is acknowledged by the base station. The erroneous packets are required to be retransmitted, which results in the reduction of the packet loss rates.

There are three main ARQ schemes in a general telecommunication network. They are the Stop-and-Wait (SW) ARQ, Go-Back-N (GBN) ARQ and Selective Repeat (SR) ARQ, respectively. Stop-and-Wait ARQ is a kind of discontinuous retransmission mechanism used for a half-duplex operation. A newly arrived packet is transmitted from the sender to the receiver only after its previous packets have been positively acknowledged. In [37-39], a detailed description and analysis for Stop-and-Wait ARQ method is given. The other two methods, Go-Back-N ARQ and Selective Repeat ARQ are categorized into the continuous retransmission mechanism for a full-duplex operation. The difference between them and Stop-and-Wait ARQ is that the packet transmissions and the acknowledgement transmissions proceed simultaneously with continuous ARQ methods. The sender does not need to receive an acknowledgement of a packet before it transmits the subsequent packets. In [16-18, 40-41], the performances of GBN systems are discussed. General packet arrival and Poisson arrival are assumed in [16-18, 40] and [41], respectively. All these papers assume that an infinite buffer is used. References [42-44] deal with the performances of SR ARQ. In [42-44], it is assumed that finite buffers are used and general packet arrival is considered. If we compare GBN ARQ with SR ARQ, GBN ARQ requires that both the particular erroneous packet and all its subsequent packets are retransmitted, while SR ARQ only requires the erroneous packet to be retransmitted.

Obviously, Stop-and-Wait ARQ is the simplest among the above three methods but it is most inefficient for communication networks. Selective Repeat ARQ is most efficient but is very complicated. In this thesis, we implement Go-Back-N ARQ method for non-real-time classes, such as the interactive and background classes, in the WCDMA system.

In the subsequent sections of this chapter, the performances of Go-Back-N ARQ are analyzed mathematically with some assumptions and constraints. Section 3.1 explains the principles of Go-Back-N ARQ and applies the GBN ARQ to a Pareto on/Pareto off process. Section 3.2 analyzes the lengthened activity factor of the non-real-time on/off traffic in the Go-back-N ARQ system. Section 3.3 analyzes the packet loss rate performance of the Pareto on/Pareto off traffic in the Go-back-N ARQ system. Section 3.4 deals with the average delay performance of the Pareto on/Pareto off traffic in the Go-Back-N system. Section 3.5 makes some discussion on the analytical work. Finally, section 3.6 concludes this chapter.

3.1 Go-Back-N ARQ Introduction

Go-Back-N (GBN) ARQ is a kind of continuous ARQ method and error control mechanism employed in the data communication networks. In a WCDMA system, due to the nature of non-real-time traffic and the requirement of high reliability, the interactive and background classes are implemented with GBN ARQ. Compared to the other ARQ methods, GBN is particularly attractive because of its simplicity and efficiency. It enables the newly arrived packets to be sent continuously regardless of the acknowledgements of the previous packets. When a negative acknowledgement is received by the sender, both the corresponding packet and its subsequent packets are continuously retransmitted. The operation of GBN ARQ can be explained jointly through the operation of the mobile users (the senders) and the base stations (the receivers) as follows.

At the sender, a finite buffer is equipped to accommodate the newly arrived packets. The sender transmits the packet at the head of the buffer to the receiver over the WCDMA channel. Before the sender obtains an acknowledgement of that packet from the receiver, the subsequent packets are sent out sequentially. The receiver sends a positive or negative acknowledgement back to the sender for each packet. If a positive acknowledgement (ACK) is sent back, the sender realizes a particular packet is correctly received and thus removes that packet from the head of the buffer. The next packet will occupy this position. If a negative acknowledgement (NACK) is sent back, both the particular packet and all its subsequent packets are retransmitted sequentially. A maximum number of retransmissions is defined for any erroneous packet. When a packet is retransmitted for the maximum times and is still erroneous, it is removed from the buffer and discarded. All packets are served with the First Come First Serve (FCFS) policy in the buffer.

An ACK is returned to the sender at the receiver in the case of correct transmissions. If an erroneous packet is received, the receiver sends a NACK to the sender and requires the sender to retransmit the relevant packets. The receiver guarantees that the NACK is not sent for more than the maximum number of retransmissions for a particular packet. The Go-Back-N ARQ method is illustrated in Figure 3.1.

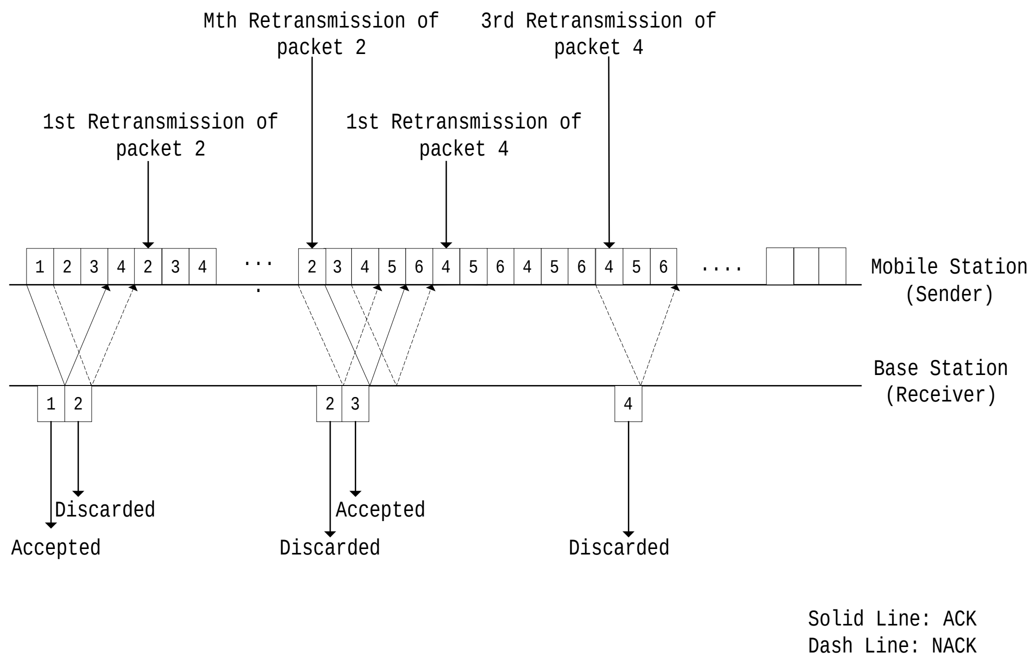

Figure 3.1 Go-Back-N ARQ Illustrations

Non-real-time classes, such as the interactive and background classes, are both modeled as Pareto on/Pareto off processes in the UMTS network. As we have discussed in Chapter 2, Pareto on/Pareto off process is a kind of on/off process in which the length of the on and off period are Pareto distributed. The source traffic alternates between the on state and the off state. Packets are generated continuously in the on state, while no packet is generated in the off state. In order to be transmitted into the channel, the packets from a non-real-time service are first stored in a finite buffer. Then, each packet in the buffer is sent to a WCDMA channel. The transmission operation of all packets fulfills the Go-Back-N ARQ method. Since the GBN ARQ is a continuous ARQ, the traffic is still an on/off process in the WCDMA channel. However, due to the retransmissions of the packets during the on period, the length of the on period in the WCDMA channel is longer than that of the on period in the source traffic. The procedure can be illustrated in Figure 3.2.

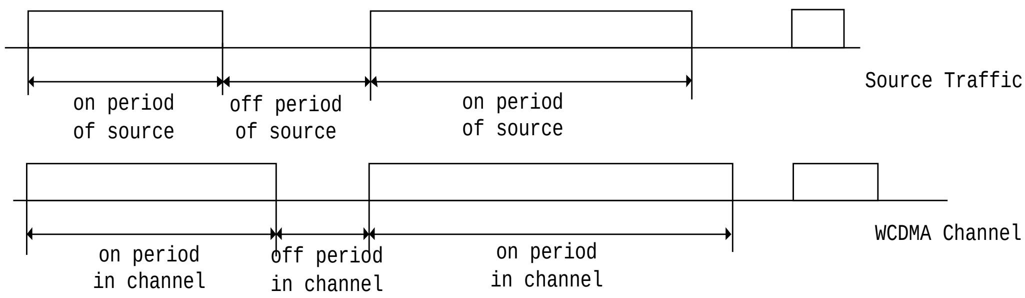

Figure 3.2 Lengthening of on Period in WCDMA Channel

According to Figure 3.2, the traffic in the WCDMA channel is still an on/off process but the on period is lengthened due to the retransmissions. Accordingly, the Pareto distributed off period is reduced, since the summation of average on period and average off period is unchanged. Thus, the lengthened on period results in a lengthened activity factor in the WCDMA channel.

The buffer will overflow if there are many retransmissions, as a finite buffer is used for each non-real-time service. The buffer overflow results in packet loss. Besides this buffer overflow, the packets that are unsuccessfully retransmitted up to the maximum times are also discarded as packet loss. Thus, a non-real-time service may experience packet loss from both buffer overflow and erroneous transmissions in the WCDMA channel. With regard to delay, any newly arrived packet experiences queuing time, transmission time, retransmission time and acknowledgement time from its arrival to its removal from buffer. The period between the arrival of a packet and the removal of the packet is referred to as the delay of a particular packet.

In the following sections, we will discuss the lengthened activity factor in WCDMA channel, the packet loss rate and the average delay. First, we list out all necessary assumptions and system parameters before proceeding with further analysis.

Assumptions

1. All ACK/NACK messages are correctly received by the sender.

2. The buffer at the receiver is finite. The buffer size in terms of packets is no less than the number of packets that can be transmitted in the acknowledgement time. This is because this assumption can guarantee that a non-real-time service is still an on/off traffic in the WCDMA channel. If this assumption is not satisfied, the transmission of the on state in the WCDMA channel will become discontinuous and thus the traffic will not be an on/off process any more in the channel.
3. The off period is long enough that the buffer is empty with the arrival of each new on period. This assumption is critical for our analysis in this chapter. In section 3.5, this assumption will be explained in greater details.

System Parameters

1. The buffer size is B in terms of number of packets.
2. Each on period contains l packets and l is a random variable.
3. Packet error probability is defined as p_{re} . p_{re} indicates the probability that a packet becomes erroneous in the system.
4. Let the maximum number of retransmissions be M_{re}.
5. Suppose the on state in source traffic is denoted by t_{on} and the off period in source traffic is denoted by t_{off} . The activity factor in the source traffic is denoted by p_{on}.
6. Suppose the on period in the WCDMA channel is denoted by $t_{on,c}$ and the off period in the WCDMA channel is denoted by $t_{off,c}$. The activity factor in the WCDMA channel is denoted by $p_{on,c}$.
7. Each on period is made up of a number of packets with the same size. Suppose the packets are generated continuously during the on period and thus there is no overlapping interval between two consecutive packets. The packet duration is denoted as T .

8. When a packet is transmitted from a mobile user to the base station, the sender needs to wait for an acknowledgement from the receiver. Suppose the acknowledgement is T_a. Assume that the ratio of T_a to T is an integer *s*. (e.g., $s = \frac{T_a}{T}$). According to the second assumption, $B \geq s$ holds.

9. If there are *l* packets during an on period, then let the overflowed packets be $N_{of}(l)$, which is a random variable related to *l*.

10. The probability density function (PDF) of t_{on} and t_{off} are given by $f(t_{on}) = c_{on} a_{on}^{c_{on}} t_{on}^{-c_{on}-1}$, $t_{on} \geq a_{on} > 0$ and $f(t_{off}) = c_{off} a_{off}^{c_{off}} t_{off}^{-c_{off}-1}$, $t_{off} \geq a_{off} > 0$. The cumulative distribution function (CDF) of t_{on} and t_{off} are given by $F(t_{on}) = 1 - a_{on}^{c_{on}} t_{on}^{-c_{on}}$ and $F(t_{off}) = 1 - a_{off}^{c_{off}} t_{off}^{-c_{off}}$. Thus, $E[t_{on}]$ is $\frac{c_{on} a_{on}}{c_{on} - 1}$ and $E[t_{off}]$ is $\frac{c_{off} a_{off}}{c_{off} - 1}$. In the following, its PDF and CDF are shown in Figures 3.3 and 3.4, respectively.

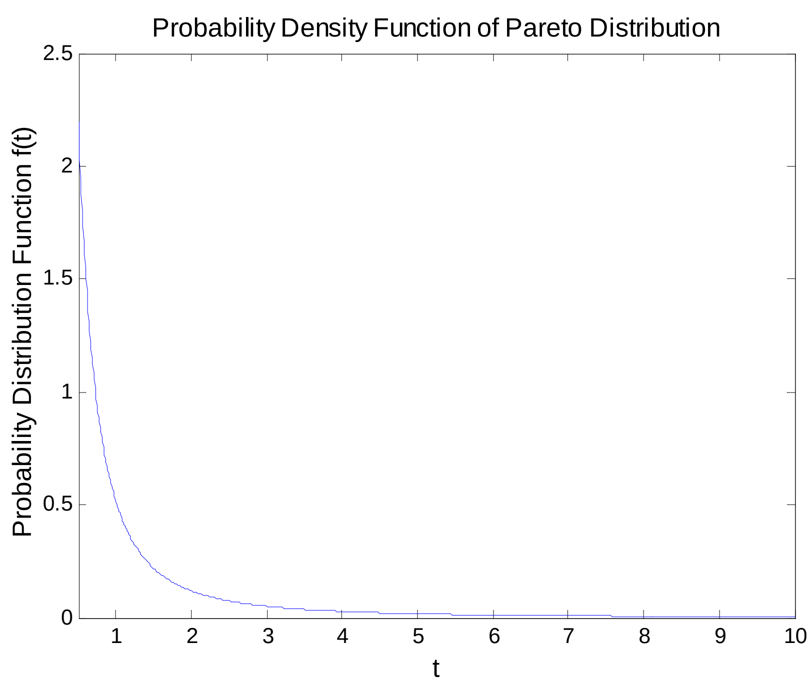

Figure 3.3 Probability Density Function of Pareto Distribution (*a*=0.5, *c*=1.1)

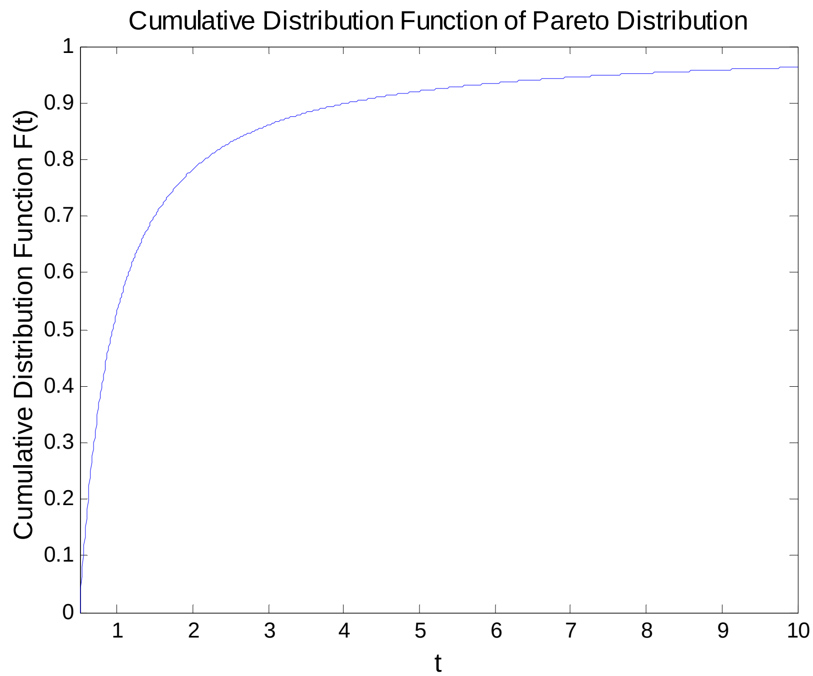

Figure 3.4 Cumulative Distribution Function of Pareto Distribution (a=0.5, c=1.1)

3.2 Analysis of the Lengthened Activity Factor

If the GBN ARQ is implemented for a Pareto on/Pareto off traffic, the average on period in the WCDMA channel will be lengthened. Packets are generated and then queued into the buffer sequentially. Since the buffer size is finite, the buffer may be full when a new packet arrives. In this case, the newly arrived packet is not transmitted into the WCDMA channel at all and is dropped directly as an overflowed packet. For those packets that enter the buffer, each packet in the buffer is continuously transmitted until it is correctly received or it is unsuccessfully retransmitted for M_{re} times. Thus, it is clear that the on period in the WCDMA channel is only influenced by those packets that enter buffer and are transmitted in the channel. Thus, the lengthened on period should be a function of the packet error probability p_{re}, the buffer size B, the maximum number of retransmissions M_{re} and (l-$N_{of}(l)$).

Let k be equal to $l\text{-}N_{of}(l)$. Thus, k, $(k \le l)$, is a random variable and denotes the number of packets that can be transmitted into the channel within an on period, if there are l packets during this on period. As the packets are transmitted sequentially in the GBN ARQ system, the packets in the buffer are transmitted in the channel according to the arriving sequences. Suppose among the k packets in the on period, the retransmission number of the ith ($1 \le i \le k$) packet is denoted by m_i ($0 \le m_i \le M_{re}$). m_i is a random variable and is associated with the packet error probability p_{re}. For each packet, m_i is independent and has the same distribution as follows.

$$Pr\{ m_i = n \} = \begin{cases} (1-p_{re})p_{re}, n=1 \\ (1-p_{re})p_{re}{}^2, n=2 \\ \ldots \\ (1-p_{re})p_{re}{}^{M_{re}-1}, n=M_{re}-1 \\ (1-p_{re})p_{re}{}^{M_{re}} + p_{re}{}^{M_{re}+1} = p_{re}{}^{M_{re}}, n=M_{re} \end{cases} \tag{3.1}$$

The above equation (3.1) is straightforward for n=1, 2, …, $M_{re}-1$. In the last expression of equation (3.1), $(1-p_{re})p_{re}{}^{M_{re}}$ denotes the probability that this packet is successfully transmitted in the last retransmission, while $p_{re}{}^{M_{re}+1}$ denotes the probability that this packet is unsuccessfully retransmitted for all the M_{re} retransmissions and the packet is discarded by the system. With equation (3.1), the mean of retransmissions for the ith is given by

$$\begin{aligned} \bar{m} &= E[m_i] \\ &= \sum_{m_i=0}^{M_{re}-1} m_i(1-p_{re})\,p_{re}{}^{m_i} + p_{re}{}^{M_{re}} M_{re} \\ &= \frac{p_{re} - p_{re}{}^{M_{re}+1}}{1-p_{re}}. \end{aligned} \tag{3.2}$$

Based on the GBN ARQ, the retransmissions of an erroneous packet not only influence the corresponding packet, but also result in the retransmissions of its succeeding packets during the period of the acknowledgement time. As stated in the system parameters, the acknowledgement time in the GBN ARQ system is supposed to be equal to the length of *s* packets. That is, an unsuccessful transmission may cause the corresponding packet and the succeeding *s* packets to be retransmitted. Before all previous packets in the buffer have been transmitted, the newly arrived packets have to be stored and queued in the buffer. Figure 3.5 shows the operation of the packet arrivals, the packet transmissions and the packet removals during an on period.

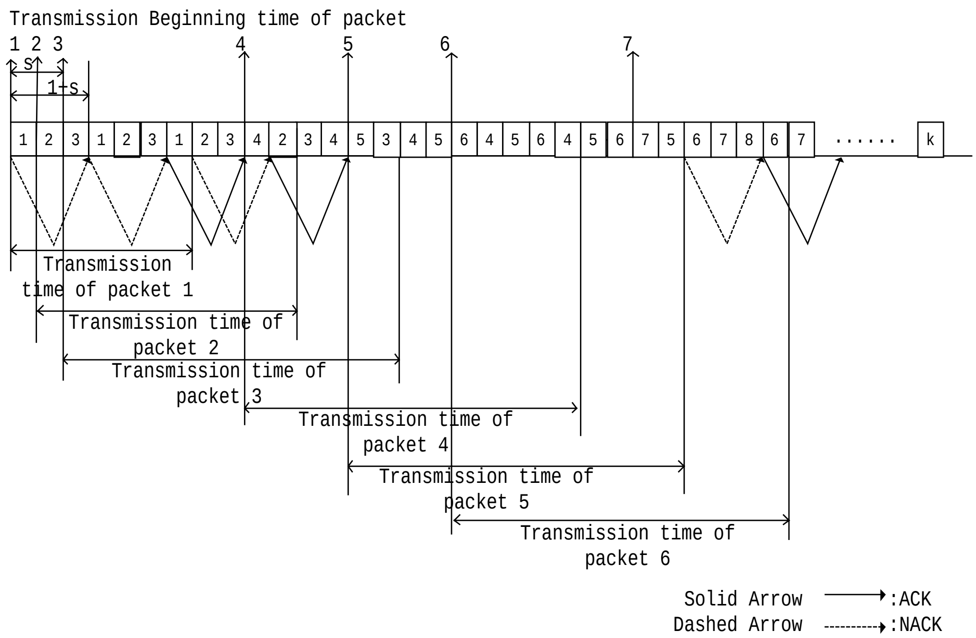

Figure 3.5 Packet Transmission Operations in the Go-Back-N ARQ System

In Figure 3.5, the packets that are transmitted in the WCDMA channel are numbered from 1 to *k* according to their sequence of entering the buffer. The lengthened on period in the WCDMA channel is equal to the interval between the beginning transmission time of the first packet and the finishing transmission time of the *k*th packet.

From Figure 3.5, the acknowledgement time T_a is equal to sT. Among all the k packets in an on period, the beginning transmission time of the first 1+s packets are the same as their arrival times. For example, if the arrival time of the first packet is $t_{arrival}$, the beginning transmission time of the first packet is also $t_{arrival}$. The beginning transmission time of the second packet is $t_{arrival}+T$. The beginning transmission time of the (1+s)th packet is $t_{arrival}+sT$. From the (s+2)th packet and the succeeding packets, their beginning transmission times are determined by the retransmissions of all previous packets. For example, the beginning transmission time of the (s+2)th packet is $t_{arrival}+\{s+[1+m_1(1+s)]\}T$. The beginning transmission time of the (s+3)th is $t_{arrival}+\{s+[1+m_1(1+s)]+[1+m_2(1+s)]\}T$. Similarly, the beginning transmission time of the kth packet is $t_{arrival}+\{s+\sum_{q=1}^{k-s-1}[1+m_q(1+s)]\}T$.

Considering the first packet, its total retransmissions are caused by the unsuccessful retransmissions of itself and the retransmission number is expressed by m_1. Thus, the total transmission time of the first packet is $[1+m_1(1+s)]T$. For the second packet, its total retransmissions are caused by both itself and the first packet. Thus the total retransmission number is equal to $1+m_1+m_2$ and its total transmission time is $[1+(m_1+m_2)(1+s)]T$. For the (1+s)th packet, its total retransmissions are caused by both itself and its previous s packets. Thus the total retransmission number is equal to $\sum_{q=1}^{1+s}m_q$ and the total transmission time of the (1+s)th packet is $[1+\sum_{q=1}^{1+s}m_q(1+s)]T$. Furthermore, for the (2+s)th packet and all the succeeding packet, their cases are similar to the (1+s)th packet. Similarly, the total retransmissions of the kth packet, $(k\le l)$, are determined by itself and its previous s packets. The total retransmission number is equal to $\sum_{q=k-s}^{k} m_q$ and thus its retransmission time is expressed as $[1+(1+s)\sum_{q=k-s}^{k} m_q)]T$.

Suppose $t_{arrival}$ is the arrival time of the first packet during the on period. Suppose $t_{begin,i}$ denote the beginning transmission time of the *i*th packet during the on period. Suppose $t_{tr,i}$ denote the total transmission time of the *i*th packet during the on period. In the following Table 3.1, we generalize $t_{begin,i}$ and $t_{tr,i}$ for all *k* packets, $(k \le l)$, that are transmitted in the WCDMA channel during an on period.

Table 3.1 Packet Beginning Transmission Time and Transmission Time in the WCDMA Channel

The *i*th packet	Beginning transmission time $t_{begin,i}$	Transmission time $t_{tr,i}$
1	$t_{arrival}$	$[1+(1+s)]\,(\sum_{q=1}^{1} m_q\,)]T$
2	$t_{arrival}+T$	$[1+(1+s)(\sum_{q=1}^{2} m_q\,)]T$
. . .	. . .	. . .
s+1	$t_{arrival}+sT$	$[1+(1+s)(\sum_{q=1}^{s+1} m_q\,)]\ T$
s+2	$t_{arrival}+\{s+[1+m_1(1+s)]\}\,T$	$[1+(1+s)(\sum_{q=2}^{s+2} m_q\,)]T$
. . .	. . .	. . .
k	$t_{arrival}+\{s+\sum_{q=1}^{k-s-1}[1+m_q(1+s)]\}T$.	$[1+(1+s)(\sum_{q=k-s}^{k} m_q)]T$

In Table 3.1, we give the beginning transmission time and the transmission time of each packet. We have assumed that if there are l packets during an on period of the source, k packets, $(k \le l)$, during the on period are transmitted in the WCDMA channel. Therefore, the lengthened on period can be formulated as a function of k. As Go-Back-N ARQ is a kind of continuous ARQ, it guarantees that the traffic in the channel is an on/off process. The lengthened on period in the WCDMA channel is the period from the beginning transmission time of the first packet to the finishing transmission time of the kth packet.

Suppose $t_{finish,i}$ denote the finishing transmission time of the ith ($1 \le i \le k$) packet. $t_{finish,i}$ can be given by

$$t_{finish,i} = t_{begin,i} + t_{tr,i}\ . \tag{3.3}$$

$t_{finish,i}$ is a finite variable. This is because $t_{begin,i}$ ($1 \le i \le k$) is a finite value that denotes the beginning transmission time of ith packet in on period with l packets, and we assume a maximum number of retransmissions (M_{re}), which makes $t_{tr,i}$ also finite. The finishing transmission time of the kth packet is given by

$$t_{finish,k} = t_{begin,k} + t_{tr,k} \tag{3.4}$$

Accordingly, the lengthened on period in the WCDMA channel is defined as $t_{on,c}$, which is given as follows.

$$\begin{aligned}
t_{on,c} &= t_{finish,k} - t_{begin,1} \\
&= t_{begin,k} + t_{tr,k} - t_{begin,1} \\
&= t_{arrival} + [1 + (1+s)\sum_{i=k-s}^{k} m_i]T + \{s + \sum_{i=1}^{k-s-1}[1 + m_i(1+s)]\}T - t_{arrival} \\
&= [k + (1+s)(\sum_{i=1}^{k} m_i)]T
\end{aligned} \tag{3.5}$$

As explained in section 3.1, the off period of the source traffic is Pareto distributed and independent of the on period. Thus, the distribution and average length of the off period in the WCDMA channel is reduced due to the lengthening of the on period. The summation of the on and of period in the WCDMA channel should be the same as the summation of the on and off periods in the source. Therefore, suppose that the off period in the WCDMA channel is defined as $t_{off,c}$. The following equation holds.

$$E[t_{on,c}] + E[t_{off,c}] = E[t_{on}] + E[t_{off}] \tag{3.6}$$

Thus, based on equations (3.5) and (3.6), the lengthened activity factor of the on/off process in the WCDMA channel is given by

$$p_{on,c} = \frac{E[t_{on,c}]}{E[t_{on,c}] + E[t_{off,c}]} . \tag{3.7}$$

From equation (3.7), the lengthened activity factor in the WCDMA channel $p_{on,c}$ is a function of $k, (k \le l)$, which is a random variable and is associated with the buffer size *B*, the packet error probability p_{re} and the maximum number of retransmissions M_{re}. Clearly, in order to obtain $p_{on,c}$ with equations (3.57), it is necessary to calculate the mean of *k*, which is denoted by $E[k]$.

Conditioned on that there are *l* packets in the on period, we assume that *i* packets have been removed from buffer when the *l*th packet arrives. If the *i*th ($0 \le i \le k$) packet finishes its transmission at $t_{finish,i}$, According to Go-Back-N ARQ mechanism, it is removed from the buffer at $t_{removal,i}$, which is given by

$$\begin{aligned} t_{removal,i} &= t_{finish,i} + T_a \\ &= t_{finish,i} + sT \end{aligned} \tag{3.8}$$

Figure 3.6 illustrates the above procedures.

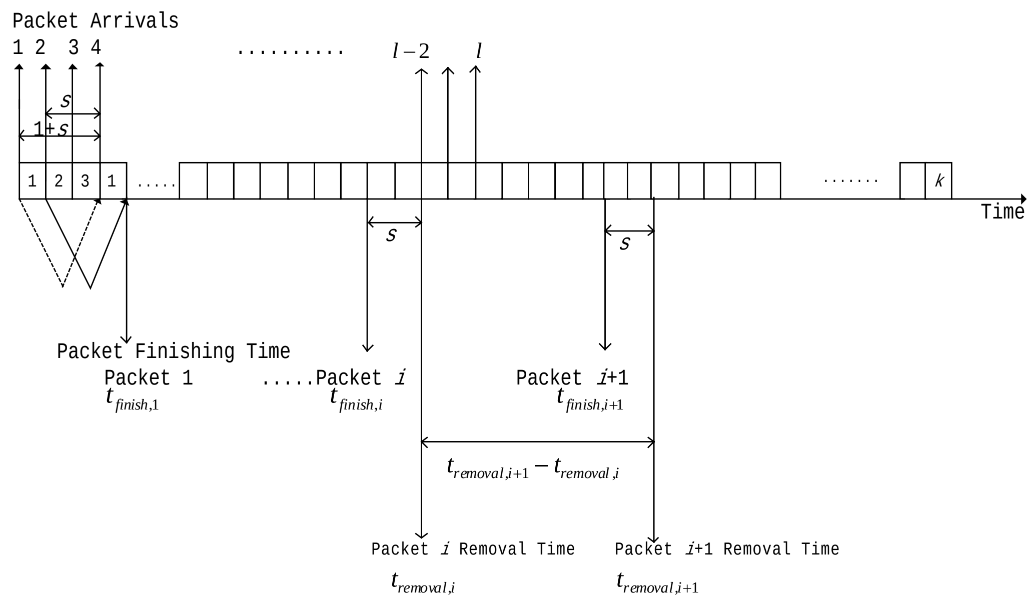

Figure 3.6 Packet Removal Operations in the Go-Back-N ARQ System

As we assumed that i packets are removed from the buffer before the lth packet arrives, the following equations are satisfied based on equation (3.8) and Table 3.1.

$$t_{removal,i} = t_{finish,i} + sT = [i + \sum_{q=1}^{i} m_q(1+s) + s]T \leq (l-1)T \tag{3.9}$$

$$t_{removal,i+1} = t_{finish,i+1} + sT = [i+1+ \sum_{q=1}^{i+1} m_q(1+s) + s]T \geq (l-1)T \tag{3.10}$$

Because m_q ranges from zero to M_{re}, the following equations (3.11) and (3.12) are satisfied, when m_q is equal to zero and M_{re}, respectively.

$$i \leq l-1-s \tag{3.11}$$

$$i \geq \max\{\frac{l-1-s}{1+(1+s)M_{re}} - 1, 0\} \tag{3.12}$$

Therefore, the following equation (3.13) is derived to express the range of the variable i.

$$\max\{\frac{(l-1-s)}{1+(1+s)M_{re}} - 1, 0\} \leq i \leq l-1-s \tag{3.13}$$

Among the i ($1 \le i \le k$) packets that have been removed from the buffer before the lth packet arrives, suppose j ($1 \le j \le i$) packets are correctly received by the receiver. Thus, *i-j* packets are unsuccessfully retransmitted for M_{re} times and are discarded by the receiver as packet loss. Therefore, the retransmission number ranges between 0 and M_{re} for any correct packet, while the retransmission is obviously M_{re} for any discarded packet. Therefore, in the following we will present the possible range of j.

From equation (3.9), the following two inequalities are obtained, when m_q of the j successful packets is equal to 0 and m_q of the (*i-j*) unsuccessful packets is equal to M_{re}.

$$\begin{cases} i+(i-j)M_{re}(1+s)+s \le l-1 \\ j \le i \end{cases}$$

Thus, the range of j is given by

$$i-\frac{l-1-s-i}{(1+s)M_{re}} \le j \le i . \tag{3.14}$$

In the Go-Back-N ARQ system, the total number of transmissions of a particular packet is not only caused by itself but also caused by its previous s packets. For example, the total number of transmissions for the ith packet is equal to $1+\sum_{q=i-s}^{i} m_q$.

The first $\sum_{q=i-s}^{i-1} m_q$ transmissions are caused by the erroneous transmissions of its previous s packets ([16], [40], [41]) and the final $1+m_i$ transmissions are caused by the ith packet itself. Thus, the first $\sum_{q=i-s}^{i-1} m_q$ transmissions of the ith packet do not influence whether the ith packet can be removed from the buffer or not. For the ith packet, only the final $1+m_i$ transmissions can determine whether it can be removed from the buffer. Therefore, for all the i packets that are removed from the buffer before the lth arrives, the sum of transmissions of all the i packets, which are caused by themselves, is denoted by n_{tr}. With rearrangements of equation (3.9), n_{tr} is given by

$$n_{tr} = \sum_{q=1}^{i}(1+m_q) \le i + \frac{l-1-s-i}{1+s} . \tag{3.15}$$

When the lth packet during the on period arrives, i packets have finished their transmissions and have been removed from the buffer, and the finite buffer can store a maximum of B packets. Then, $l-i-B$ denotes the number of overflowed packets in all the l packets. $\binom{i}{j}(1-p_{re}{}^{M_{re}+1})^{j}(p_{re}{}^{M_{re}+1})^{i-j}$ denotes the probability that j packets are correctly received out of all the i removed packets before the lth packet during the on period arrives. $\binom{n_{tr}}{j}(1-p_{re})^{j}\,p_{re}{}^{n_{tr}-j}$ denotes the probability that before the lth packet in the on period arrives, there are j correct transmissions in all the n_{tr} transmissions.

From equations (3.13), (3.14), and (3.15), the mean of $N_{of}(l)$ in the on period is denoted by $E[N_{of}(l)]$ and can be formulated by

$$E[N_{of}(l)] = \frac{\sum_{i=i_{\min}}^{i_{\max}}\sum_{j=j_{\min}}^{i}\binom{n_{tr}}{j}(1-p_{re})^{j}\,p_{re}{}^{n_{tr}-j}\binom{i}{j}(1-p_{re}{}^{M_{re}+1})^{j}(p_{re}{}^{M_{re}+1})^{i-j}\,max(l-i-B,0)}{\sum_{i=i_{\min}}^{i_{\max}}\sum_{j=j_{\min}}^{i}\binom{n_{tr}}{j}(1-p_{re})^{j}\,p_{re}{}^{n_{tr}-j}\binom{i}{j}(1-p_{re}{}^{M_{re}+1})^{j}(p_{re}{}^{M_{re}+1})^{i-j}}, \tag{3.16}$$

where

$$i_{\min} = \max\{\frac{l-1-s}{1+(1+s)M_{re}}-1, 0\},$$

$$i_{\max} = l-1-s ,$$

and

$$j_{\min} = i - \frac{l-1-s-i}{(1+s)M_{re}}.$$

Furthermore, as the on period is Pareto distributed, the probability that an on period has *l* packet, denoted by $p(l)$, is approximately given by.

$$p(l) = \Pr\{t = lT\} = \int_{lT}^{(l+1)T} c_{on} a^{c_{on}} t^{-c_{on}-1} dt,\ t \geq a_{on} \tag{3.17}$$

From equations (3.16) and (3.17), the mean of *k*, $E[k]$, is given by

$$E[k] = \sum_{l=a_{on}/T}^{\infty} \{p(l)(l - E[N_{of}(l)])\}. \tag{3.18}$$

Therefore, the mean of the lengthened on period in the WCDMA channel can be formulated from equation (3.5) and (3.18), and is given by

$$E[t_{on,c}] = \sum_{l=a/T}^{\infty} \{p(l)[1 + \frac{(p_{re} - p_{re}{}^{M_{re}+1})(1+s)}{1-p_{re}}]E[k]T\}. \tag{3.19}$$

Therefore, with the packet error probability as p_{re}, the lengthened activity factor in the WCDMA channel, $p_{on,c}$, is formulated based on equations (3.7), (3.16), (3.18) and (3.19) and is given as follows.

$$p_{on,c} = \frac{\sum_{l=a/T}^{\infty} \{p(l)[1 + \frac{(p_{re} - p_{re}{}^{M_{re}+1})(1+s)}{1-p_{re}}]E[k]T\}}{E[t_{on}] + E[t_{off}]}$$

In order to facilitate further analysis in subsequent chapters, based on equations (3.7) and (3.19), we simplify the lengthened activity factor in the WCDMA channel, $p_{on,c}$, as a function, $AfFun(T, T_a, B, c_{on}, a_{on}, c_{off}, a_{off}, p_{re}, M_{re})$, which is given by

$$p_{on,c} = AfFun(T, T_a, B, c_{on}, a_{on}, c_{off}, a_{off}, p_{re}, M_{re}). \tag{3.20}$$

3.3 Analysis of Packet Loss Rate

In the Go-Back-N system, the packet loss is due to two sources. The first source is unsuccessful retransmissions. If a packet is retransmitted for M_{re} times and is still erroneous, it is discarded as packet loss. The second source is the finite buffer overflow. As the Go-Back-N system implements a finite buffer, the arriving packets may be dropped, if the buffer becomes full during the transmissions. Therefore, the total packet loss is the sum of the unsuccessful packets and the overflowed packets. In the following, we first calculate the packet loss during the on period that contains l packets. As the length of the on period is Pareto distributed and l is a random variable, we next calculate the average packet loss rate over time by summing up all possible packet loss rates with the corresponding probabilities.

Equation (3.16) gives the average number of overflowed packet, $E[N_{of}(l)]$, on the condition that the on period contains l packets and the packet error probability is p_{re}. Then, the number of unsuccessful packets due to erroneous retransmissions is defined to be $N_{error}(l)$. The mean of $N_{error}(l)$ is given by

$$E[N_{error}(l)] = \{l - E[N_{of}(l)]\} p_{re}^{M_{re}+1}. \tag{3.21}$$

Therefore, with the packet error probability p_{re} and the on period containing l packets, the packet loss number is denoted by $N_{loss}(l)$. The mean of $N_{loss}(l)$ is given by

$$E[N_{loss}(l)] = E[N_{error}(l)] + E[N_{of}(l)]. \tag{3.22}$$

Then, conditioned on that there are l packets in the on period, the packet loss rate is given by

$$P_{loss}(l) = E[N_{loss}(l)]/l. \tag{3.23}$$

To derive the average packet loss rate over time, we need to sum up all probabilistic packet loss rates. According to equations (3.17) and (3.23), the average packet loss rate in the WCDMA channel is P_{loss}, which is given by

$$P_{loss} = \sum_{l=a/T}^{\infty} [P_{loss}(l)\, p(l)]. \tag{3.24}$$

In order to facilitate the analysis in the subsequent chapters, we simplify the average packet loss rate in the WCDMA channel as a function, $PlossFun(T, T_a, B, c_{on}, a_{on}, c_{off}, a_{off}, p_{re}, M_{re})$, which is given by

$$P_{loss} = PlossFun(T, T_a, B, c_{on}, a_{on}, c_{off}, a_{off}, p_{re}, M_{re}). \tag{3.25}$$

3.4 Analysis of Delay

In the Go-Back-N ARQ system, the packet delay is defined as the whole period between the arrival and the removal of a packet. Packet delay is further comprised of three parts: queuing delay, transmission delay and acknowledgement delay. Firstly, the queuing delay of a packet is due to the waiting period from the arrival time to the beginning transmission time of that packet. In the GBN ARQ system, when packets are generated, they are queued in the buffer and served on the FCFS policy. Secondly, the transmission delay of a packet refers to the period between the beginning transmission time and the finishing transmission time. According to our assumption, the transmission delay is restricted by the maximum number of retransmissions, M_{re}. Thirdly, as a correctly transmitted packet has to wait for an additional T_a time to receive an ACK from the receiver before it can be removed from the buffer, the acknowledgement time, T_a, is also part of the packet delay. The packet delay is illustrated in Figure 3.7.

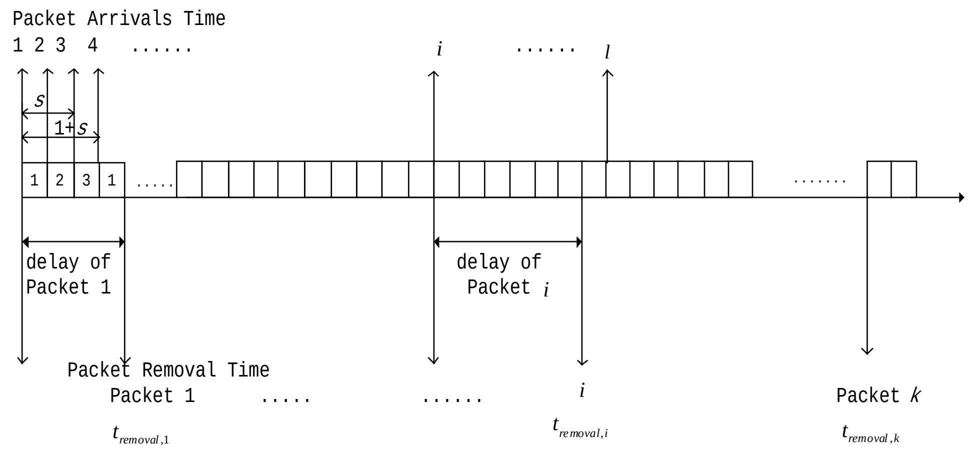

Figure 3.7 Packet Delay Illustration in the WCDMA Channel

In the assumptions in section 3.1, we assume that each new on period arrives with an empty buffer. If an on period contains *l* packets, the buffer length is increasing with the arrivals of these packets. The number of packets in the buffer may reach a maximum value during the on period in the WCDMA channel and stay at this state until the *l*th packet arrives. After the arrival of the *l*th packet, no new packet is generated in the on period. Thus, the number of packets in the buffer then decreases from the maximum value to zero. The on period in the WCDMA channel ends when all packets finishes transmissions. After that, the off period in the WCDMA channel starts. The buffer is empty from the removal of the last packet in the buffer until the arrival of the next on period. In each on/off cycles, the buffer length is varying similarly. Thus, we can investigate the number of packets in the buffer on conditioned on that there are *l* packets during an on period as follows.

$$
B_{length}(t\,|\,l)=\begin{cases}
min\{1,B\} & t_{arrival}\le t\le t_{arrival}+T\\
min\{2,B\} & t_{arrival}+T\le t\le t_{arrival}+2T\\
\cdots\cdots\cdots\cdot & \\
min\{\frac{t_{removal,1}}{T},B\}, & t_{arrival}+t_{removal,1}-T\le t\le t_{arrival}+t_{removal,1}+T\\
min\{\frac{t_{removal,1}}{T}+1,B\}, & t_{arrival}+t_{removal,1}+T\le t\le t_{arrival}+t_{removal,1}+2T\\
\cdots\cdots\cdot & \\
min\{\frac{t_{removal,2}}{T},B\}, & t_{arrival}+t_{removal,2}\le t\le t_{arrival}+t_{removal,2}+T\\
\cdots\cdots\cdots & \\
min\{\frac{t_{removal,i}}{T}-(i-1),B\}, & t_{arrival}+t_{removal,i}-T\le t\le t_{arrival}+t_{removal,i}+T\\
min\{\frac{t_{removal,i}}{T}-i,B\}, & t_{arrival}+t_{removal,i}+T\le t\le t_{removal,i}+2T\\
\cdots\cdots & \\
min\{k-i-1,B\}, & t_{arrival}+(l-2)T\le t\le t_{arrival}+(l-1)T\\
min\{k-i,B\}, & t_{arrival}+(l-1)T\le t\le t_{arrival}+t_{removal,i+1}\\
\cdots\cdots & \\
min\{1,B\}, & t_{arrival}+t_{removal,k-1}\le t\le t_{arrival}+t_{removal,k}\\
0, & t_{arrival}+t_{removal,k}\le t\le t_{arrival}+t_{on,c}+t_{off,c}.
\end{cases}
$$

(3.26)

In equation (3.26), $B_{length}(t\,|\,l)$ is the number of packets in the buffer if an on period has l packets. From (3.26), the mean of $B_{length}(t\,|\,l)$ over time can be expressed as $\bar{B}_{length}(l)$, which is given by

$$
\bar{B}_{length}(l)=E\{\sum_{j=1}^{\frac{t_{removal,1}}{T}} min\{j,B\}+\sum_{j=\frac{t_{removal,1}}{T}}^{\frac{t_{removal,2}}{T}-1} min\{j,B\}+\ldots\ldots\sum_{j=\frac{t_{removal,i-1}}{T}-(i-2)}^{\frac{t_{removal,i}}{T}-(i-1)} min\{j,B\}
$$

$$
+\sum_{j=\frac{t_{removal,i}}{T}-(i-1)}^{l-i-1} min\{j,B\}+[\frac{t_{removal,i+1}}{T}-(l-1)]min\{k-i,B\}+[\frac{t_{removal,i+2}-t_{removal,i+1}}{T}]min\{k-(i+1),B\}
$$

$$
+\ldots\ldots+min\{2,B\}[\frac{t_{removal,k-1}-t_{removal,k-2}}{T}]+min\{1,B\}(\frac{t_{removal,k}-t_{removal,k-1}}{T})/(\frac{t_{on,c}+t_{off,c}}{T})\}, \qquad (3.27)
$$

which can be solved with equation (3.8) and Table 3.1.

Furthermore, if the on period has l packets, the average arrival rate is assumed to be $\lambda(l)$.

$$\lambda(l) = \frac{E[k]}{E[t_{on}] + E[t_{off}]} = \frac{l - E[N_{of}(l)]}{E[t_{on}] + E[t_{off}]} \tag{3.28}$$

Based on equation (3.28), the average packet arrival over the time, denoted by λ, is given by

$$\lambda = \sum_{l=a_{on}/T}^{\infty} [\lambda(l)\, p(l)]. \tag{3.29}$$

Similarly, the average buffer length B over the time is based on equation (3.27) and is roughly given by

$$B = \sum_{l=a_{on}/T}^{\infty} [\bar{B}_{length}(l)\, p(l)]. \tag{3.30}$$

According to Little's theory, the average delay of a packet, D, can be obtained by the ratio of the average number of packets in the buffer to the average arrival rate of packets. D can be given by

$$D = B / \lambda\ . \tag{3.31}$$

In order to facilitate further analysis in the subsequent chapters, we simplify the average packet delay in the WCDMA channel as a function, $DelayFun(T, T_a, B, c_{on}, a_{on}, c_{off}, a_{off}, p_{re}, M_{re})$, which is given by

$$D = DelayFun(T, T_a, B, c_{on}, a_{on}, c_{off}, a_{off}, p_{re}, M_{re})\ . \tag{3.32}$$

3.5 Discussions

In the above analyses of the packet loss rate, average delay and lengthened activity factor, the third assumption in section 3.1 is a very critical condition. According to the third assumption in section 3.1, it must be guaranteed that the off period t_{off} in the source traffic is long enough and each newly arrived on period encounters an empty buffer. This assumption is made because our analysis is based on the probabilistic sum of the analysis in each individual on period. That is, we generalize the parameters, such as packet loss rate, delay and lengthened activity factor, for each possible on period and then obtain the average of these parameters over time. Thereby, each on period is investigated independently to simplify our analytical work. However, this assumption results in some constraints of our approach. Strictly, it is possible that the new on period is coming before the former on period has finished, if there are too many retransmission occurrences. Thus, in this case, the newly arrived on period may encounter a non-empty buffer and the assumption 3 in section 3.1 is violated.

Fortunately, a practical system commonly has a low packet error probability in terms of a few percents and the lengthening of on periods in the channel is not significant, compared to the off period. For instance, t_{off} for interactive and background classes is usually larger than a few seconds, while t_{on} is only lengthened by a few hundreds milliseconds in the channel. Thus, t_{off} is long enough so that the probability that a newly arrived on period encounters a non-empty buffer is negligible. Hence, the third assumption in section 3.1 is reasonable.

3.6 Conclusion

This chapter mainly studies the Go-Back-N ARQ mechanism in the WCDMA system. The Pareto on/Pareto off process is assumed to approximate the source traffic of a non-real-time service. A finite buffer is provided to each non-real-time service. The Go-back-N ARQ results in the lengthening of the activity factor of the traffic in the WCDMA channel. We calculate the lengthened activity factor and address the analyses of the QoS performances, including the packet loss rate and the average delay, in the Go-Back-N ARQ system. The results obtained from this chapter provide a basis for the QoS analysis of non-real-time services in the WCDMA system and will be referred to in subsequent chapters.

Chapter 4

Analysis of Outage Probability

In this chapter, we address the data link layer QoS issue by investigating the outage probabilities for multiclass services in the uplink of the WCDMA cellular mobile network. The data link layer is the second layer of the WCDMA system and is directly related to the packet level QoS in the network layer. The main measurement of the data link layer QoS refers to the outage probability.

In this chapter, we consider a cellular system with multiple cells. Each cell serves a number of mobile users. The received signal at the base station in the reference cell is interfered by both mobile users in its own cell and mobile users in other neighbouring cells. When the interference in a cell exceeds a predetermined level, or the SINR of received signal is below a predefined level, the link layer QoS performance is not tolerable any more and outage occurs. Our objective in this chapter is to analyze the outage probability in the uplink. Since we have assumed in Chapter 2 that voice, video, web-browsing and data services exist in the system, we give mathematical formulas of outage probabilities for these services. This chapter is organized as follows.

- Before we investigate the outage probability performance, we propose two system models that contain a variety of services in the WCDMA system in section 4.1. One of them is a single-connection system model in which each mobile user supports only one connection of service. Different classes of services are served within different mobile users. Therefore, a service is interfered by all other services in the system. The other model is a multi-connection system model in which one mobile user can support multi-connection multiclass services simultaneously. In this model, a service is only interfered by services of other mobile users, not including the other services within the same mobile user. This is because synchronous orthogonal spreading codes are used for services in the same mobile user. We study and generalize the distinct approaches used for these two system models.

- In section 4.2, we describe a medium access control/radio link control (MAC/RLC) method in the WCDMA system. The MAC/RLC method is responsible for resource allocation, spreading code assignment, packet access and transmission mode in the WCDMA system.
- In section 4.3, an efficient power distribution algorithm is designed. With this algorithm, perfect power control is assumed in the system. We calculate the desired received power of each traffic class in both system models.
- In section 4.4, the analytical expressions of outage probabilities are derived for each traffic class in both system models.
- In section 4.5, we evaluate the characteristics of the on/off processes in the WCDMA channel for web-browsing and data services. Because web-browsing and data are both considered as non-real-time services, Go-Back-N ARQ is implemented in case of transmission errors. The Go-Back-N ARQ lengthens the on period of web-browsing and data services in the WCDMA channel. The lengthened on period increases the activity factors of web-browsing and data services. In section 4.5, we derived the lengthened activity factors of web-browsing and data services, and formulate the

outage probabilities of all traffic classes as a function of the lengthened activity factors in both system models. Furthermore, since the lengthened activity factors and the outage probabilities are intertwined, an iteration method is proposed to solve for the outage probabilities and the activity factors in the WCDMA channel.

- We conclude this chapter in section 4.6.

4.1. System Model

A cellular network consists of a number of cells and mobile users. We assume that each cell serves the same number of mobile users and services. Mobile users are uniformly located in each cell and are served by a base station that is situated in the center of the cell. Mobile users and base stations can communicate via the WCDMA channels. Here, we only consider the uplink of the WCDMA system. Mobile users transmit signals to their own base stations in the uplink. The system permits the transmissions of multiclass services simultaneously in the WCDMA system. Each mobile user can support a single-connection service or multi-connection services simultaneously. As given in Chapter 3, voice, video, web-browsing, and data are chosen as representatives of the four different QoS traffic classes. When a service within a mobile user is transmitting packets to the base station in a cell, other mobile users in the same cell and neighbouring cells contribute interference to the system and decrease the QoS performance of that particular service. In this section, we propose two system models to describe the network. The former is a single-connection system model and the latter is a multi-connection system model. The common assumptions are given in the followings.

Assumptions

- All mobile users are uniformly located in each cell.

- An Additive White Guassian Noise (AWGN) channel is assumed.
- Perfect power control is implemented for each service and the desired received powers are achieved at the base station.
- Convolutional coding is used for each class as the method of error correction.
- Both the ETSI [20] and 3GPP [62] specifications adopt the square cell (Manhattan) model in their simulations. Hence, in this thesis, the square cell model, which is also commonly being adopted in the literature [58-61], is used.

4.1.1 Single-Connection System Model

In this model, there are a number of mobile users in a cell. We assume that each mobile user only supports a single-connection service. All mobile users share the uplink channel to the base station and thus interfere with each other. As given in Chapter 2, we assume that each voice, web-browsing or data service reserves a single spreading code. On the other hand, a video service reserves a high-bit-rate spreading code and M low-bit-rate spreading codes. Additionally, when signals are transmitted from mobile users to the base station using spreading codes, perfect power control is assumed in each code channel. That is, the transmit power of each service is adjusted dynamically against the effect of the variation of the channel. Wherever a mobile user is in a cell, the transmit powers of its traffic classes are changed quickly to maintain the received powers at a desired level. Additive White Guassian Noise (AWGN) channel is assumed in the system. The four classes also differ in their bit error rate (BER) and signal-to-interference- plus-noise ratio (SINR) requirements, which are specified for each class.

Only one service is transmitted within each mobile user in the single-connection system model. A service in the mobile user is first spread with spreading code, then is scrambled with a scrambling code, and finally is transmitted over the AWGN channel. Figure 4.1 illustrates the procedures of transmissions within the mobile users.

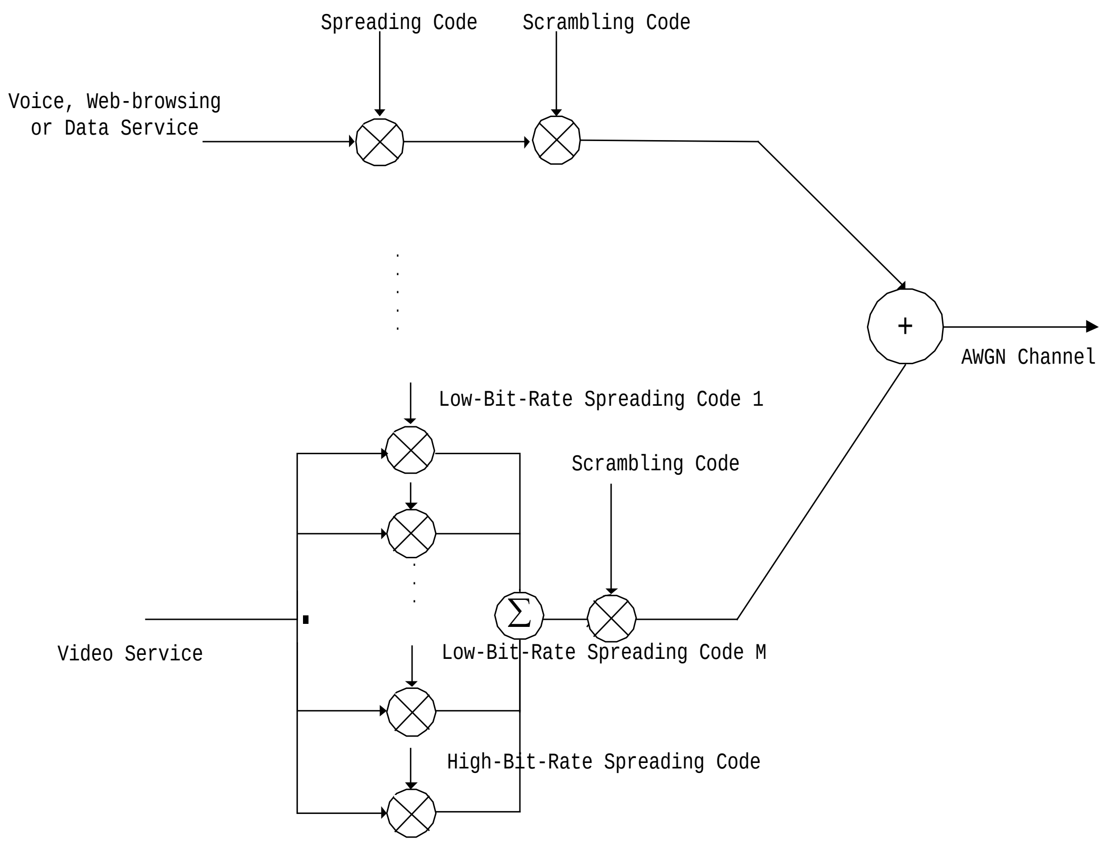

Figure 4.1 Spreading and Scrambling for the Single-Connection System Model

The following are system parameters for the single-connection system model.

System Parameters

1. G_k , BER_k^* , γ_k^* and S_k , $k \in \{1, 2l, 2h, 3, 4\}$, denote the spreading gains, BER requirements, SINR requirements and desired received power levels of voice, video using low-bit-rate spreading code, video using high-bit-rate spreading code, web-browsing and data services during the on period, respectively.

2. η denotes the average power of AWGN.

3. $I_{intercell}$ denotes the intercell interference in the system.

4. N denotes the number of mobile users in a cell. Within the N mobile users, N_1, N_2, N_3 and N_4 denote the number of voice, video, web-browsing and data services in a cell, respectively.

6. p_{on1}, p_{on2l}, p_{on2h}, p_{on3} and p_{on4} denote the activity factors of voice, video using low-bit-rate spreading code, video using high-bit-rate spreading code, web-browsing and data services, respectively in their source traffic. $p_{on3,c}$ and $p_{on4,c}$ denote the activity factors of web-browsing and data services, respectively in the WCDMA channel.

7. l_k, $k \in \{1, 2l, 2h, 3, 4\}$, denote the instantaneous number of active spreading codes used by all voice services, by all low-bit-rate video services, by all high-bit-rate video services, by all web-browsing services and by all data services, respectively. ($0 \le l_1 \le N_1$, $0 \le l_{2l} \le MN_2$, $0 \le l_2 \le N_2$, $0 \le l_3 \le N_3$, $0 \le l_4 \le N_4$)

8. $\psi_{j,1}$, $\psi_{j,2l}$, $\psi_{j,2h}$, $\psi_{j,3}$ and $\psi_{j,4}$ denote the instantaneous number of active spreading codes used by the *j*th voice service, active low-bit-rate spreading codes used by the *j*th video service, active high-bit-rate spreading codes used by the *j*th video service, active spreading codes used by the *j*th web-browsing service and active spreading codes used by the *j*th data service, respectively. ($0 \le \psi_{j,1} \le 1$, $0 \le \psi_{j,2l} \le M$, $0 \le \psi_{j,2h} \le 1$, $0 \le \psi_{j,3} \le 1$, $0 \le \psi_{j,4} \le 1$)

9. $P_{out,k}$, $k \in \{1, 2l, 2h, 3, 4\}$, denote the outage probabilities of voice, video using low-bit-rate spreading code, video using high-bit-rate spreading code, web-browsing and data services, respectively.

4.1.2 Multi-Connection System Model

In general, the multi-connection system model is similar to the single-connection system model described in section 4.1.1, but it makes improvement in some aspects. In contrast to the single-connection system model, the multi-connection system model is able to support multi-connection multiclass services within each mobile user. That is, each mobile user can have multiple connections to serve more than one traffic class. Obviously, since the transmissions within a mobile user are completely orthogonal, a service is not interfered by the other services of the same mobile user. Comparatively, a service in single-connection system model is interfered by all other services in cells, as one mobile user only supports one service. As each mobile user serves different combination of classes and experiences different amount of interference, the same class of services within different mobile users needs different power levels to satisfy the SINR requirements, depending on the other services that the individual mobile user serves.

Similar to the single-connection system model, before all services within the same mobile user are sent out, they are first spread with the corresponding spreading codes, then are scrambled with a common scrambling code and finally are transmitted over the AWGN channel, which is assumed in the multi-connection system model. The Figure 4.2 illustrates the procedures of transmissions within the mobile users.

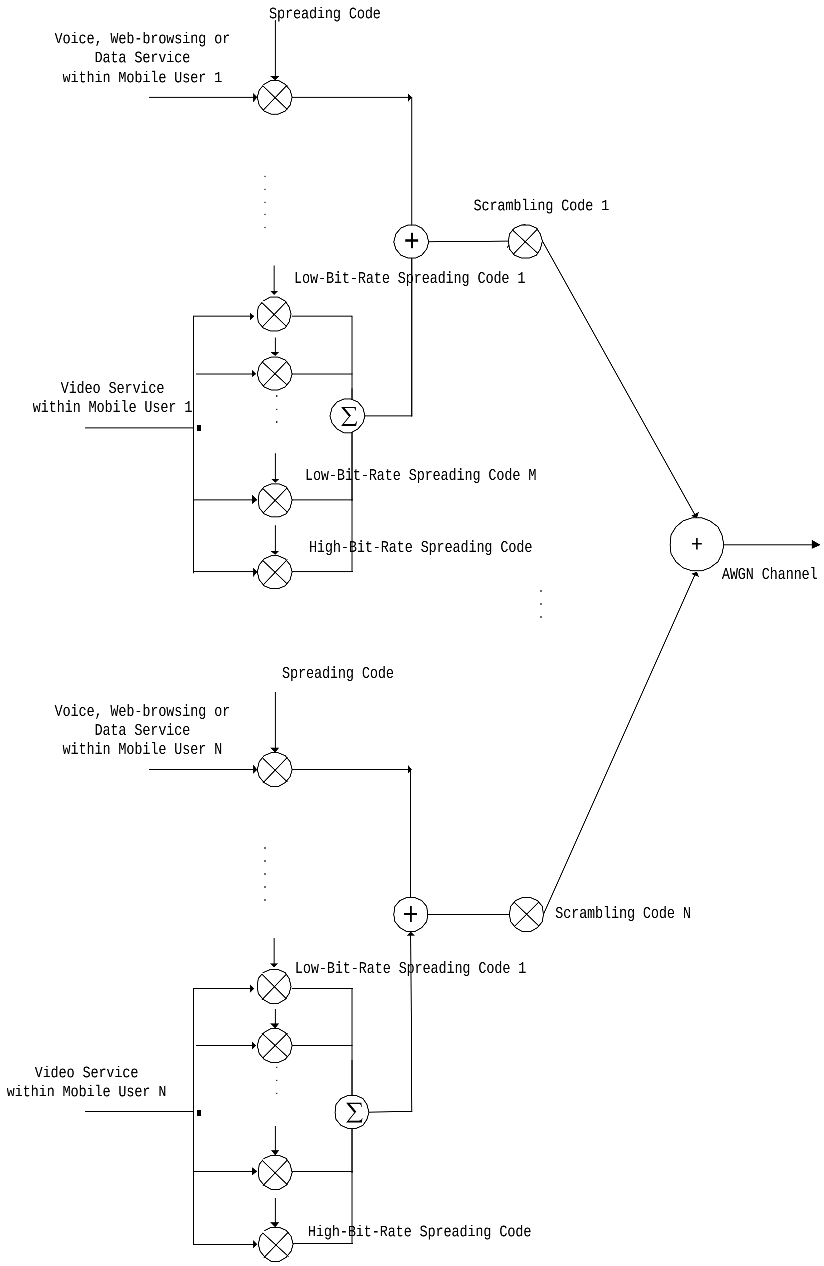

Figure 4.2 Spreading and Scrambling for the Multi-Connection System Model

The followings are the system parameters in the multi-connection system model.

System Parameters

1. Refer to the system parameters 1-5 in the single-connection system model.

2. N denotes the number of mobile users in a cell.

3. $n_{i,1}$, $n_{i,2}$, $n_{i,3}$ and $n_{i,4}$ denote the number of voice, video, web-browsing and data services within the *i*th mobile user in a cell.

4. p_{on1}, p_{on2l}, p_{on2h}, p_{on3} and p_{on4} denote the activity factors of voice, video using low-bit-rate spreading code, video using high-bit-rate spreading code, web-browsing and data services, respectively in their source traffic. $p_{on3,i,c}$ and $p_{on4,i,c}$ denote the activity factors of web-browsing and data services, respectively in the WCDMA channel within the *i*th mobile user.

5. $S_{i,k}$, $k \in \{1, 2l, 2h, 3, 4\}$, denote the desired received powers of voice, video using low-bit-rate spreading code, video using high-bit-rate spreading code, web-browsing and data services within the *i*th mobile user.

6. $l_{i,k}$, $k \in \{1, 2l, 2h, 3, 4\}$, denotes the instantaneous number of active spreading codes used by all voice services, low-bit-rate video services, high-bit-rate video services, web-browsing services and data services within the *i*th mobile user, respectively ($0 \le l_{i,1} \le n_{i,1}$, $0 \le l_{i,2l} \le Mn_{i,2}$, $0 \le l_{i,2h} \le n_{i,2}$, $0 \le l_{i,3} \le n_{i,3}$, $0 \le l_{i,4} \le n_{i,4}$).

5. $\psi_{i,j,1}$, $\psi_{i,j,2l}$, $\psi_{i,j,2h}$, $\psi_{i,j,3}$ and $\psi_{i,j,4}$ denote the instantaneous number of active spreading codes used by the *j*th ($0 \le j \le n_{i,1}$) voice service, active low-bit-rate spreading codes used by the *j*th ($1 \le j \le n_{i,2}$) video service, active high-bit-rate spreading codes used by the *j*th ($1 \le j \le n_{i,2}$) video service, active spreading codes used by the *j*th ($1 \le j \le n_{i,3}$) web-browsing service and active spreading codes used by the *j*th ($1 \le j \le n_{i,4}$) data service within the *i*th ($1 \le i \le N$) mobile user. ($0 \le \psi_{i,j,1} \le 1$, $0 \le \psi_{i,j,2l} \le M$, $0 \le \psi_{i,j,2h} \le 1$, $0 \le \psi_{i,j,3} \le 1$, $0 \le \psi_{i,j,4} \le 1$)

6. $P_{out,i,k}$, $k \in \{1, 2l, 2h, 3, 4\}$, denote the outage probabilities for voice, video using a low-bit-rate spreading code, video using a high-bit-rate spreading code, web-browsing and data services respectively within the *i*th mobile user.

4.2 MAC/RLC Method

Medium access control (MAC) and radio link control (RLC) are two sub-layers of data link layer. The functions of MAC include selection of appropriate spreading code to each service, handling of services with different priorities; service access procedures in the WCDMA system and supporting multiple connections within one mobile user. The task of RLC focuses on the reliable transfer of signals over the wireless interface. It is achieved through an automatic retransmission request (ARQ) method. There are many studies in the literature on concrete MAC/RLC protocol for WCDMA interface, such as [25-28, 51-53].

In this section, we introduce a simple MAC/RLC method implemented in our system. In Chapter 3, voice, video, web-browsing and data are representative applications of conversational class, streaming class, interactive class and background class, respectively. According to their traffic models, voice, web-browsing and data services are on/off processes. Since the transmission rate during the on period is a constant, the MAC layer assigns a single spreading code to each voice, web-browsing or data service. In this chapter, we assume all services are transmitted to the base station through Dedicated Channel (DCH), which is identified by a spreading code. In order to utilize spreading code resources in the system efficiently, a DCH is assigned to a voice, web-browsing or data service at the start of each on period and is released and withdrawn by the base station at the end of each on period. On the other hand, a video service is a kind of VBR traffic. Based on the proposed MAC protocol, at the beginning of a video service, a set of spreading codes, which is comprised of a high-bit-rate spreading code and M low-bit-rate spreading codes, is assigned to a video service. Since the instantaneous bit rate of the video service is varying, the MAC layer is in charge of selecting a particular combination of these spreading codes to satisfy the rate level. When the service ends, the set of spreading codes are released by the base station.

RLC layer is responsible for the reliable transmissions of signals. The quality of a wireless communication link is highly unstable and is influenced by the multiple access interference (MAI) of the WCDMA system. MAI causes transmission errors to the packets over the wireless channel from mobile users to the base station. In order to guarantee reliable transmissions and protect the information bits from errors, RLC layer provides two schemes to fulfill this function. The first is known as forward error correction (FEC). A FEC scheme inserts redundant bits into a packet and makes it possible to correct at least some of the detected erroneous information bits during the transmissions. Besides FEC, another error correction scheme is called automatic repeat request (ARQ). If the receiver at the base station detects the bit error rate of a received packet is above the required level, it asks for a retransmission of the corresponding packet from the mobile user. As we have discussed in Chapter 3, we choose Go-Back-N ARQ in our analysis as it is commonly implemented.

Obviously, ARQ results in a longer transfer delay of a service. Consequently, as delay-sensitive real-time services, voice and video only implement FEC mechanism. In comparison, as delay-insensitive non-real-time services, web-browsing and data implement both FEC and ARQ methods.

4.3 Power Distribution Algorithm

The system capacity in the WCDMA network and the QoS performances are directly associated with multiple access interference (MAI) which is contributed by interfering mobile users. Therefore, signal-to-interference-plus-noise ratio (SINR) is an important attribute at the data link layer. SINR is a function of the received powers at the base station, the spreading gains of all services and the number of active spreading codes. Thus, SINR is fluctuating over time. To attain good performance at the data link layer, it is necessary that the average SINR of each service should be maintained at a required level. Furthermore, we assume that perfect power control in the WCDMA system is adopted so that the desired received powers of each service are achieved at the receiver of the base station. In the following, the power distribution schemes are designed in the two system models described in sections 4.1.1 and 4.1.2, respectively.

4.3.1 Power Distribution for Single-Connection System Model

Each mobile user has only a single connection in the single-connection system model. The same class of services requires the same amount received power in the system. Therefore, power assignment is performed based on each class. As the WCDMA system needs to guarantee that the average received SINR of each class exceeds the SINR requirements of each class of service, the following equations have to be satisfied in the WCDMA channel.

$$\frac{S_1G_1}{p_{on1}(N_1-1)S_1+p_{on2l}MN_2S_{2l}+p_{on2h}N_2S_{2h}+p_{on3,c}N_3S_3+p_{on4,c}N_4S_4+E[I_{intercell}]+\eta}=\gamma_1^{\ *} \tag{4.1}$$

$$\frac{S_{2l}G_{2l}}{p_{on1}N_1S_1+p_{on2l}M(N_2-1)S_{2l}+p_{on2h}(N_2-1)S_{2h}+p_{on3,c}N_3S_3+p_{on4,c}N_4S_4+E[I_{intercell}]+\eta}=\gamma_{2l}^{\ *} \tag{4.2}$$

$$\frac{S_{2h}G_{2h}}{p_{on1}N_1S_1+p_{on2l}M(N_2-1)S_{2l}+p_{on2h}(N_2-1)S_{2h}+p_{on3,c}N_3S_3+p_{on4,c}N_4S_4+E[I_{intercell}]+\eta}=\gamma_{2h}^{\ *} \tag{4.3}$$

$$\frac{S_3G_3}{p_{on1}N_1S_1+p_{on2l}MN_2S_{2l}+p_{on2h}N_2S_{2h}+p_{on3,c}(N_3-1)S_3+p_{on4,c}N_4S_4+E[I_{intercell}]+\eta}=\gamma_3^{\ *} \tag{4.4}$$

$$\frac{S_4G_4}{p_{on1}N_1S_1+p_{on2l}MN_2S_{2l}+p_{on2h}N_2S_{2h}+p_{on3,c}N_3S_3+p_{on4,c}(N_4-1)S_4+E[I_{intercell}]+\eta}=\gamma_4^{\ *} \tag{4.5}$$

In the above equations, the first five terms in the denominator of each equation denote the intracell interference of voice service, video service using low-bit-rate spreading gain, video service using high-bit-rate spreading code, web-browsing service and data service, respectively. The sixth term denotes the total intercell interference of voice service, video service using low-bit-rate spreading gain, video service using high-bit-rate spreading code, web-browsing service and data services. The last term is the average background noise power of the AWGN.

From equations (4.1)-(4.5), the desired received power levels S_1, S_{2l}, S_{2h}, S_3 and S_4 can be obtained. After some algebraic rearrangements of equations (4.1)-(4.5), the following equations are derived.

$$\begin{aligned}&(p_{on1}+\frac{G_1}{\gamma_1^{\ *}})S_1=\\&p_{on1}N_1S_1+p_{on2l}MN_2S_{2l}+p_{on2h}N_2S_{2h}+p_{on3,c}N_3S_3+p_{on4,c}N_4S_4+E[I_{intercell}]+\eta\end{aligned} \tag{4.6}$$

$$\begin{aligned}&(\frac{G_{2l}}{\gamma_{2l}^{*}}+p_{on2l}M+\frac{p_{on2h}G_{2l}/\gamma_{2l}^{*}}{G_{2h}/\gamma_{2h}^{*}})S_{2l}=\\&p_{on1}N_1S_1+p_{on2l}MN_2S_{2l}+p_{on2h}N_2S_{2h}+p_{on3,c}N_3S_3+p_{on4,c}N_4S_4+E[I_{intercell}]+\eta\end{aligned}\tag{4.7}$$

$$\begin{aligned}&(\frac{G_{2h}}{\gamma_{2h}^{*}}+p_{on2h}+p_{on2l}M\frac{G_{2h}/\gamma_{2h}^{*}}{G_{2l}/\gamma_{2l}^{*}})S_{2h}=\\&p_{on1}N_1S_1+p_{on2l}MN_2S_{2l}+p_{on2h}N_2S_{2h}+p_{on3,c}N_3S_3+p_{on4,c}N_4S_4+E[I_{intercell}]+\eta\end{aligned}\tag{4.8}$$

$$\begin{aligned}&(p_{onk,c}+\frac{G_k}{\gamma_k^{*}})S_k=\\&p_{on1}N_1S_1+p_{on2l}MN_2S_{2l}+p_{on2h}N_2S_{2h}+p_{on3,c}N_3S_3+p_{on4,c}N_4S_4+E[I_{intercell}]+\eta\end{aligned}\tag{4.9}$$

where $k\in\{3,4\}$.

In the right hand side of equations (4.6)-(4.9), $I_{intercell}$ is the total intercell interference. $E[I_{intercell}]$ and $Var[I_{intercell}]$ are the mean and variance of the intercell interference, respectively. In the Appendix, $E[I_{intercell}]$ and $Var[I_{intercell}]$ can be shown to be given by the following.

$$\begin{aligned}&E[I_{intercell}]\\&\le(S_1p_{on1}N_1+S_{2l}Mp_{on2l}N_2+S_{2h}p_{on2h}N_2+S_3p_{on3,c}N_3+S_4p_{on4,c}N_4)\frac{\iint f(\frac{r_m}{r_d})dA}{A}\end{aligned}\tag{4.10}$$

$$\begin{aligned}&Var[I_{intercell}]\\&=N_1\frac{\iint[p_{on1}g(\frac{r_m}{r_d})-p_{on1}^{2}f^2(\frac{r_m}{r_d})]dA}{A}+N_2\frac{\iint\{Mp_{on2l}[1+(M-1)p_{on2l}]g(\frac{r_m}{r_d})-(Mp_{on1})^2f^2(\frac{r_m}{r_d})]\}dA}{A}\\&N_3\frac{\iint[p_{on3,c}g(\frac{r_m}{r_d})-p_{on3,c}^{2}f^2(\frac{r_m}{r_d})]dA}{A}+N_4\frac{\iint[p_{on4,c}g(\frac{r_m}{r_d})-p_{on4,c}^{2}f^2(\frac{r_m}{r_d})]dA}{A}\end{aligned}\tag{4.11}$$

where $f(\frac{r_m}{r_d})$ is given by

$$f(\frac{r_m}{r_d})=(\frac{r_m}{r_d})^4e^{(\frac{\sigma\ln10}{10})^2}[1-Q(\frac{40\log(r_m/r_d)}{\sqrt{2\sigma^2}}-\frac{\sqrt{2\sigma^2}\ln10}{10})],\tag{4.12}$$

and

$$g(\frac{r_m}{r_d}) = (\frac{r_m}{r_d})^8 e^{(\frac{\sigma \ln 10}{5})^2} [1 - Q(\frac{40\log(r_m / r_d)}{\sqrt{2\sigma^2}} - \frac{\sqrt{2\sigma^2}\ln 10}{5})]. \tag{4.13}$$

In equations (4.10) and (4.11), r_m denotes the distance between an intercell mobile user and its own base station; r_d denotes the distance between an intercell mobile user and the intracell base station; ε_m and ε_d are two independent Guassian random variables with σ^2 variance and zero mean; A denotes the area of a cell.

The power vector $\boldsymbol{S}$ is defined as $[S_1, S_{2l}, S_{2h}, S_3, S_4]$. Therefore, we can clearly see that the objective of the power distribution is to derive a positive solution for the vector $\boldsymbol{S}$.

Suppose ε is a real number. Based on equation (4.12), we define ε as follows.

$$\varepsilon = 1 - [\frac{p_{on1}N_1}{p_{on1} + \frac{G_1}{\gamma_1^*}} + \frac{p_{on2l}MN_2}{\frac{G_{2l}}{\gamma_{2l}^*} + p_{on2l}M + \frac{p_{on2h}G_{2l}/\gamma_{2l}^*}{G_{2h}/\gamma_{2h}^*}} + \frac{p_{on2h}N_2}{\frac{G_{2h}}{\gamma_{2h}^*} + p_{on2h} + p_{on2l}M\frac{G_{2h}/\gamma_{2h}^*}{G_{2l}/\gamma_{2l}^*}}$$
$$+ \frac{p_{on3,c}N_3}{p_{on3,c} + \frac{G_3}{\gamma_1^*}} + \frac{p_{on4,c}N_4}{p_{on4,c} + \frac{G_4}{\gamma_4^*}}] \times [1 + \frac{\iint f(\frac{r_m}{r_d})dA}{A}] \tag{4.14}$$

According to [29-31], for the linear equations (4.6)-(4.9), if and only if $0 \le \varepsilon \le 1$ is satisfied, $\boldsymbol{S}$ has a positive solution. The solution can be easily given by

$$S_1 = \frac{\eta}{(1-\varepsilon)(p_{on1} + \frac{G_1}{\gamma_1^*})}, \tag{4.15}$$

$$S_{2l} = \frac{\eta}{(1-\varepsilon)(\frac{G_{2l}}{\gamma_{2l}^*} + p_{on2l}M + \frac{p_{on2h}G_{2l}/\gamma_{2l}^*}{G_{2h}/\gamma_{2h}^*})}, \tag{4.16}$$

$$S_{2h} = \frac{\eta}{(1-\varepsilon)(\frac{G_{2h}}{\gamma_{2h}^{*}} + p_{on2h} + p_{on2l} M \frac{G_{2h}/\gamma_{2h}^{*}}{G_{2l}/\gamma_{2l}^{*}})}, \tag{4.17}$$

$$S_3 = \frac{\eta}{(1-\varepsilon)(p_{on3,c} + \frac{G_3}{\gamma_1^{*}})}, \tag{4.18}$$

and

$$S_4 = \frac{\eta}{(1-\varepsilon)(p_{on4,c} + \frac{G_4}{\gamma_4^{*}})}. \tag{4.19}$$

Accordingly, if the condition $0 \le \varepsilon \le 1$ holds, equations (4.15)-(4.19) satisfy the SINR requirements in equations (4.1)-(4.5). Similar to the single-connection system model, equations (4.15)-(4.19) gives the minimum power target of each service in the system. From equations (4.1)-(4.5), it is obvious that if all positive received powers are increased by the same ratio, the achieved SINR will exceed the corresponding SINR requirements as AWGN can be overcome, and the data link layer QoS of the system is improved. Therefore, in our calculation of outage probability, the positive power solutions of all services obtained from equations (4.15)-(4.19) are multiplied by a common factor, θ ($\theta > 1$), to attain better QoS performance.

4.3.2 Power Distribution for Multi-Connection System Model

Each mobile user may transmit multi-connection multiclass services in the multi-connection system model. The same class of services within different mobile users needs different power levels to satisfy the SINR requirements in the case of the multi-connection system model. The power distribution in the multi-connection system model assigns the received power targets to each connection of all mobile users. In [54], we propose a new power distribution scheme to allocate powers to different services of each mobile user. According to our method in [54], the average SINR of each class of services are expressed as follows based on the system parameters within the *i*th mobile user.

For the voice service within the *i*th mobile user, the average SINR is given by

$$\frac{S_{i,k}G_k}{\sum_{j=1;j\neq i}^{N}(S_{j,1}p_{on1}n_{j,1}+S_{j,2l}p_{on2l}Mn_{j,2}+S_{j,2h}p_{on2h}n_{j,2}+S_{j,3}p_{on3,i,c}n_{j,3}+S_{j,4}p_{on4,i,c}n_{j,4})+E[I_{intercell}]+\eta} = \gamma_1^*. \quad (4.20)$$

where

$$k \in \{1, 2l, 2h, 3, 4\}$$

In equation (4.20), the first term in the denominator refers to the average intracell interference; the second term in the denominator refers to average intercell interference; and the third term in the denominator refers to the background noise. Note that different connections of services for the same mobile user do not interfere each other since short orthogonal codes are used for spreading. Thus, different services in the same mobile user experience the same amount of interference. Equation (4.20) differs from the SINR equations (4.1)-(4.5), which only investigates the case that each mobile user supports a single connection. In the case of multi-connection multiclass services, the same class of services within different mobile users needs different power levels to satisfy the SINR requirements. In the following, we will present how the desired received powers can be solved.

In order to solve equation (4.20), we first define

$$\bar{S}_i = p_{on1}n_{i,1}S_{i,1}+p_{on2l}Mn_{i,2}S_{i,2l}+p_{on2h}n_{i,2}S_{i,2h}+p_{on3,i,c}n_{i,3}S_{i,3}+p_{on4,i,c}n_{i,4}S_{i,4}, \quad (4.21)$$

and $$\Gamma_i = p_{on1}n_{i,1}\frac{\gamma_1^*}{G_1}+p_{on2l}Mn_{i,2}\frac{\gamma_{2l}^*}{G_{2l}}+p_{on2h}n_{i,2}\frac{\gamma_{2h}^*}{G_{2h}}+p_{on3,i,c}n_{i,3}\frac{\gamma_3^*}{G_3}+p_{on4,i,c}n_{i,4}\frac{\gamma_4^*}{G_4}. \quad (4.22)$$

With the algebraic manipulations that we develop in [54], equation (4.20) is transformed into the following equation.

$$[1+\Gamma_i]\bar{S_i} = \Gamma_i(\sum_{j=1}^{N}\bar{S_j} + E[I_{intercell}] + \eta) \tag{4.23}$$

$I_{intercell}$ denotes the intercell interference. $E[I_{intercell}]$ denotes the mean of the intercell interference. The mean and variance of the total intercell interference can be given by (See Appendix)

$$E[I_{intercell}] \le \sum_{i=1}^{N}(S_{i,1}n_{i,1}p_{on1} + S_{i,2l}n_{i,2}p_{on2l}M + S_{i,2h}n_{i,2}p_{on2h} + S_{i,3}n_{i,3}p_{on3,i,c} + S_{i,4}n_{i,4}p_{on4,i,c})$$
$$\times\frac{\iint f(\frac{r_m}{r_d})dA}{A}, \tag{4.24}$$

and $Var[I_{intercell}]$

$$\le \sum_{j=1}^{N}\{S_{j,1}{}^2 n_{j,1}\frac{\iint[p_{on1}g(\frac{r_m}{r_d}) - p_{on1}{}^2 f^2(\frac{r_m}{r_d})]dA}{A}$$
$$+S_{j,2l}{}^2 n_{j,2}\frac{\iint[Mp_{on2l}[1+(M-1)p_{on2l}]g(\frac{r_m}{r_d}) - (Mp_{on2l})^2 f^2(\frac{r_m}{r_d})]dA}{A}$$
$$+S_{j,2h}{}^2 n_{j,2}\frac{\iint[p_{on2h}g(\frac{r_m}{r_d}) - (p_{on2h})^2 f^2(\frac{r_m}{r_d})]dA}{A}$$
$$+S_{j,3}{}^2 n_{j,3}\frac{\iint[p_{on3,i,c}g(\frac{r_m}{r_d}) - p_{on3,i,c}{}^2 f^2(\frac{r_m}{r_d})]dA}{A}$$
$$+S_{j,4}{}^2 n_{j,4}\frac{\iint[p_{on4,i,c}g(\frac{r_m}{r_d}) - p_{on4,i,c}{}^2 f^2(\frac{r_m}{r_d})]dA}{A}\}. \tag{4.25}$$

In equations (4.24) and (4.25), the definitions of $f(\frac{r_m}{r_d})$, $g(\frac{r_m}{r_d})$, r_m and r_d are as the same as those in the single-connection system model.

Thus, equation (4.23) is algebraically rearranged as follows based on equation (4.24).

$$\frac{(1+\Gamma_i)}{\Gamma_i}\bar{S}_i = \left(1+\frac{\iint f(\frac{r_m}{r_d})dA}{A}\right)\sum_{i=1}^{N}\bar{S}_i + \eta \qquad (4.26)$$

The power vector $\boldsymbol{S_i}$ is defined as $[S_{i,1}, S_{i,2l}, S_{i,2h}, S_{i,3}, S_{i,4}]$. Therefore, we can clearly see that the objective of the power distribution is to derive a positive solution for the vector $\boldsymbol{S_i}$, $0 \le i \le N$. Suppose

$$\varepsilon = 1 - \sum_{i=1}^{N} \frac{\left(1+\frac{\iint f(\frac{r_m}{r_d})dA}{A}\right)\Gamma_i}{1+\Gamma_i}. \qquad (4.27)$$

According to theorems given by [29-31], for the linear equation (4.26), if and only if $0 \le \varepsilon \le 1$ is satisfied, $\bar{S}_i$ has a positive solution. The solution can be easily given by

$$\bar{S}_i = \frac{\eta \Gamma_i}{\varepsilon(1+\Gamma_i)}. \qquad (4.28)$$

Otherwise, it is impossible to find a positive solution to satisfy equation (4.20). From the definition of $\bar{S}_i$, if a positive $\bar{S}_i$ is available, the positive power vector $\boldsymbol{S_i}$ exists and the desired received powers within the *i*th mobile user are formulated for each type of services. Thus, we have

$$S_{i,j} = \frac{\eta \gamma_j^*}{\varepsilon(1+\Gamma_i)G_j}, \; j = \{1, 2l, 2h, 3, 4\}. \qquad (4.29)$$

Equation (4.29) gives the minimum power target of each service in the system to satisfy the SINR requirements. In order to make power distribution, equation (4.27) is first checked. If the condition $0 \le \varepsilon \le 1$ holds, equation (4.29) is solved. According to equation (4.20), if all positive received powers are increased by the same ratio, the AWGN is less important, which obviously makes the achieved SINR exceed the corresponding SINR requirements and the data link layer QoS of the system better. Therefore, the positive power solutions obtained from equation (4.29) are usually multiplied by a common factor, θ ($\theta > 1$), in distributing received power targets to attain better QoS performance.

4.4 Outage Probability

Generally, in telecommunication systems, outage is a kind of service condition in which a system parameter is below a predefined threshold. This threshold is called the outage threshold. It indicates the minimum performance level that is needed in the system. BER requirement or SINR requirement of each class in the WCDMA system is regarded as the outage threshold. According to 3GPP Technical Specification, each QoS traffic class may tolerate a specific level of BER. If the BER is less than the BER requirement, the receiver considers a packet to be correctly received. Otherwise, the receiver will either discard the packets of voice and video services directly or require the mobile user to retransmit the packets of web-browsing and data services.

Accordingly, the outage probability in a WCDMA system is defined as the probability that the achieved SINR is below the SINR requirement or the achieved BER is above the BER requirement. In [15,21-22], Wong et al. analyze the outage probabilities for on/off multiclass services, variable bit rate multiclass services and video multiclass services in DS-CDMA systems, respectively. However, these analyses do not consider retransmissions of services. Thus, we will extend the formulation of the outage probabilities in the Go-Back-N ARQ system.

As defined in the system models in sections 4.1.1 and 4.1.2, BER_k^* and γ_k^*, $k \in \{1, 2l, 2h, 3, 4\}$, denote the BER and SINR requirements for voice service, video service using low-bit-rate spreading codes, video service using high rate spreading codes, web-browsing service and data service, respectively. According to sections 4.1.1 and 4.1.2, we will present the outage probability for the two system models.

4.4.1 Outage Probability for Single-Connection System Model

In our multi-cell cellular mobile networks, the total interference includes intracell interference and intercell interference. The intracell interference is a function of the number of active spreading codes in the intracell and the desired received powers of all classes. The intercell interference is influenced by the shadowing effect and mobile user distribution. According to our assumptions, all mobile users are assumed uniformly distributed in each cell. The fluctuation of the intercell interference is usually assumed to be a lognormal shadowing. Furthermore, due to the large number of mobile user in the system, the total intercell interference is approximated to be a Guassian random variable, with a mean and a variance given by $E[I_{intercell}]$ and $Var[I_{intercell}]$. Equations (4.1)–(4.5) express the average SINR for each class. Actually, the instantaneous SINR in networks is varying over the time. When the number of active spreading codes in the intracell is specified, the instantaneous outage probability of each class is determined by Guassian distributed intercell interference and thus is fluctuating. As the number of active spreading codes can be approximated by binomial distribution, we can sum up all instantaneous outage probability with the binomial state probabilities to obtain the average outage probability for each class. Let BER_k and γ_k, $k \in \{1, 2l, 2h, 3, 4\}$, be the actual achieved BER and SINR of voice, video using low-bit-rate spreading code, video using high-bit-rate spreading code, web-browsing and data services, respectively, at the receiver of the base station.

For voice, low-bit-rate video, high-bit-rate video, web-browsing and data services, the average outage probabilities are given by equation (4.30), for *k*=1, 2*l*, 2*h*, 3 and 4, respectively. For $k=\{1,3,4\}$, let $N_j^{'}=\begin{cases}N_j-1, & j=k\\ N_j, & j\neq k\end{cases}$, $j=\{1,2,3,4\}$ and for $k=\{2l,2h\}$,

let $N_j^{'}=\begin{cases}N_j-1, & j=2\\ N_j, & j\neq 2\end{cases}$, $j=\{1,2,3,4\}$.

$$\begin{aligned}&P_{out,k}=Pr(BER_k\geq BER_k^{*})=Pr(\gamma_k\leq\gamma_k^{*})\\&=\sum_{l_1=0}^{N_1^{'}}\sum_{l_{2h}=0}^{N_2^{'}}\sum_{l_{2l}=0}^{MN_2^{'}}\sum_{l_3=0}^{N_3^{'}}\sum_{l_4=0}^{N_4^{'}}\\&\{I_{intercell}\geq\frac{S_kG_k}{\gamma_k^{*}}-\eta-(l_1S_1+l_{2l}S_{2l}+l_{2h}S_{2h}+l_3S_3+l_4S_4)\,|\\&\sum_{j=1}^{N_1^{'}}\psi_{j,1}=l_1,\sum_{j=1}^{MN_2^{'}}\psi_{j,2l}=l_{2l},\sum_{j=1}^{N_2^{'}}\psi_{j,2h}=l_{2h},\sum_{j=1}^{N_3^{'}}\psi_{j,3}=l_3,\sum_{j=1}^{N_4^{'}}\psi_{j,4}=l_4\}.\end{aligned}\tag{4.30}$$

In equation (4.30), we can see that when the active spreading codes used by each service are given, the instantaneous outage probability of a service is the probability that the total intercell interference is above a specific level. Since the total intercell interference is approximately Guassian distributed, the instantaneous outage probabilities can be given by a *Q(x)* function. Then, equation (4.30) can be transformed into

$$\begin{aligned}P_{out,k}=&\sum_{l_1=0}^{N_1^{'}}\sum_{l_{2h}=0}^{N_2^{'}}\sum_{l_{2l}=0}^{MN_2^{'}}\sum_{l_3=0}^{N_3^{'}}\sum_{l_4=0}^{N_4^{'}}\{Q(\frac{\delta_k-\mu_k}{\sigma_k})\binom{N_1^{'}}{l_1}(p_{on1})^{l_1}(1-p_{on1})^{N_1^{'}-l_1}\binom{N_2^{'}}{l_{2h}}(p_{on2h})^{l_{2h}}(1-p_{on2h})^{N_2^{'}-l_{2h}}\\&\binom{MN_2^{'}}{l_{2l}}(p_{on2l})^{l_{2l}}(1-p_{on2l})^{MN_2^{'}-l_{2l}}\binom{N_3^{'}}{l_3}(p_{on3,c})^{l_3}(1-p_{on3,c})^{N_3^{'}-l_3}\binom{N_4^{'}}{l_4}(p_{on4,c})^{l_4}(1-p_{on4,c})^{N_4^{'}-l_4}\},\end{aligned}\tag{4.31}$$

where

$$\delta_k=\frac{S_kG_k}{\gamma_k^{*}}-\eta,\tag{4.32}$$

$$\mu_k = (l_1\eta + l_{2l}S_{2l} + l_{2h}S_{2h} + l_3S_3 + l_4S_4) + E[I_{intercell}], \tag{4.33}$$

$$\sigma_k^{\ 2} = Var[I_{intercell}], \tag{4.34}$$

$$Q(x) = \int_x^{\infty} \frac{1}{\sqrt{2\pi}} e^{-\frac{x^2}{2}} dx, \tag{4.35}$$

and

$$k \in \{1, 2l, 2h, 3, 4\}. \tag{4.36}$$

In the above expressions of the outage probabilities, $E[I_{intercell}]$ and $Var[I_{intercell}]$ are given in equations (4.10) and (4.11).

4.4.2 Outage Probability for Multi-Connection System Models

Each mobile user in the multi-connection system model can support multi-connection multiclass services simultaneously, which is different from the single-connection system model. That is, each mobile user can have multiple connections to serve more than one traffic class. Since the power distribution algorithms are deigned for both the two system models in section 4.3, the outage probabilities are different in the two models. Let $BER_{i,k}$ and $\gamma_{i,k}$, $k \in \{1, 2l, 2h, 3, 4\}$, be the actual achieved BER and SINR of voice, video using low-bit-rate spreading code, video using high-bit-rate spreading code, web-browsing and data services, respectively, within the ith mobile user at the receiver of the base station. Since the number of active spreading codes used by services is varying, the instantaneous outage probabilities are fluctuating over time. The average outage probability of each service can be obtained by calculating the probabilistic sum of all possible instantaneous outage probabilities.

Accordingly, we will generalize outage probabilities for each of the services within each mobile user. As assumed in the section 4.1, the outage probabilities within the *i*th mobile user for voice, video using a low-bit-rate spreading code, video using a high-bit-rate spreading code, web-browsing and data services are expressed as $P_{out,i,k}$, $k \in \{1, 2l, 2h, 3, 4\}$, respectively and given by

$$\begin{aligned}
&P_{out,i,k} = Pr\{\, BER_{i,k} \le BER_k^{\ *} \,\} = Pr\{\, \gamma_k \ge \gamma_k^{\ *} \,\} \\
&= \sum_{l_{1,1}=0}^{n_{1,1}} \sum_{l_{1,2l}=0}^{Mn_{1,2}} \sum_{l_{1,2h}=0}^{n_{1,2}} \sum_{l_{1,3}=0}^{n_{1,3}} \sum_{l_{1,4}=0}^{n_{1,4}} \cdots \sum_{\substack{l_{j,1}=0 \\ j\neq i}}^{n_{j,1}} \sum_{\substack{l_{j,2l}=0 \\ j\neq i}}^{Mn_{j,2}} \sum_{\substack{l_{j,2h}=0 \\ j\neq i}}^{n_{j,2}} \sum_{\substack{l_{j,3}=0 \\ j\neq i}}^{n_{j,3}} \sum_{\substack{l_{j,4}=0 \\ j\neq i}}^{n_{j,4}} \cdots \sum_{l_{N,1}=0}^{n_{N,1}} \sum_{l_{N,2l}=0}^{Mn_{N,2}} \sum_{l_{N,2h}=0}^{n_{N,2}} \sum_{l_{N,3}=0}^{n_{N,3}} \sum_{l_{N,4}=0}^{n_{N,4}} \\
&\{\, I_{intercell} \ge \frac{S_{i,k} G_k}{\gamma_k^{\ *}} - \eta - (\sum_{i=1}^{N} \sum_{l_{i,1}=0}^{n_{i,1}} l_{k,1} S_{i,1} + \sum_{i=1}^{N} \sum_{l_{i,2l}=0}^{Mn_{i,2}} l_{i,2l} S_{i,2l} + \sum_{i=1}^{N} \sum_{l_{i,2h}=0}^{n_{i,2h}} l_{i,2h} S_{i,2h} \\
&+ \sum_{i=1}^{N} \sum_{l_{i,3}=0}^{n_{i,3}} l_{i,3} S_{i,3} + \sum_{i=1}^{N} \sum_{l_{i,4}=0}^{n_{i,4}} l_{i,4} S_{i,4} \mid \sum_{j=1}^{n_{i,1}} \psi_{i,j,1} = l_{i,1}, \sum_{j=1}^{n_{i,2l}} \psi_{i,j,2l} = l_{i,2l}, \sum_{j=1}^{n_{i,2h}} \psi_{i,j,2h} = l_{i,2h}, \\
&\sum_{j=1}^{n_{i,3}} \psi_{i,j,3} = l_{i,3}, \sum_{j=1}^{n_{i,4}} \psi_{i,j,4} = l_{i,4}, i = 1, 2, \ldots, N \,\},
\end{aligned} \tag{4.37}$$

where

$$k \in \{1, 2l, 2h, 3, 4\} .$$

The difference in calculating the average outage probability in the multi-connection system model is that the possible combinations of the number of active spreading codes are more than those in the single-connection system model. The instantaneous outage probability is also denoted by the variation of the intercell interference. As $I_{intercell}$ is a Guassian random variable, the instantaneous outage probability can be formulated by a *Q(x)* function shown in equation (4.35). Thus, equation (4.37) can be given by

$$
\begin{aligned}
P_{out,i,k} = &\sum_{l_{1,1}=0}^{n_{1,1}} \sum_{l_{1,2l}=0}^{Mn_{1,2}} \sum_{l_{1,2h}=0}^{n_{1,2}} \sum_{l_{1,3}=0}^{n_{1,3}} \sum_{l_{1,4}=0}^{n_{1,4}} \cdots \sum_{\substack{l_{j,1}=0\\ j\neq i}}^{n_{j,1}} \sum_{\substack{l_{j,2l}=0\\ j\neq i}}^{Mn_{j,2}} \sum_{\substack{l_{j,2h}=0\\ j\neq i}}^{n_{j,2}} \sum_{\substack{l_{j,3}=0\\ j\neq i}}^{n_{j,3}} \sum_{\substack{l_{j,4}=0\\ j\neq i}}^{n_{j,4}} \cdots \sum_{l_{N,1}=0}^{n_{N,1}} \sum_{l_{N,2l}=0}^{Mn_{N,2}} \sum_{l_{N,2h}=0}^{n_{N,2}} \sum_{l_{N,3}=0}^{n_{N,3}} \sum_{l_{N,4}=0}^{n_{N,4}} \\
&\{Q(\frac{\delta_{i,k}-\mu_i}{\sigma_i}) \prod_{\substack{j=1\\ j\neq i}}^{N} \binom{n_{j,1}}{l_{j,1}} (p_{on1})^{l_{j,1}} (1-p_{on1})^{n_{j,1}-l_{j,1}} \binom{Mn_{j,2}}{l_{j,2l}} (p_{on2l})^{l_{j,2l}} (1-p_{on2l})^{Mn_{j,2}-l_{j,2l}} \\
&\times \binom{n_{j,2}}{l_{j,2h}} (p_{on2h})^{l_{j,2h}} (1-p_{on2h})^{n_{j,2}-l_{j,2h}} \binom{n_{j,3}}{l_{j,3}} (p_{on3,i,c})^{l_{j,3}} (1-p_{on3,i,c})^{n_{j,3}-l_{j,3}} \\
&\times \binom{n_{j,4}}{l_{j,4}} (p_{on4,i,c})^{l_{j,4}} (1-p_{on4,i,c})^{n_{j,4}-l_{j,4}} \}, \qquad (4.38)
\end{aligned}
$$

where

$$k \in \{1, 2l, 2h, 3, 4\},$$

$$\delta_{i,k} = \frac{S_{i,k}G_k}{\gamma_k^*} - \eta, \qquad (4.39)$$

$$\mu_i = \sum_{j=1;\, j\neq i}^{N} (l_{j,1}S_{j,1} + l_{j,2l}S_{j,2l} + l_{j,2h}S_{j,2h} + l_{j,3}S_{j,3} + l_{j,4}S_{j,4}) + E[I_{intercell}], \qquad (4.40)$$

and

$$\sigma_i^2 = Var[I_{intercell}], \qquad (4.41)$$

In the expressions of the above outage probabilities, $E[I_{intercell}]$ and $Var[I_{intercell}]$ are given by equations (4.24) and (4.25).

4.5 Lengthened Activity Factor

As stated in Chapter 3, the Go-Back-N ARQ is implemented for the transmissions of non-real-time services, such as web-browsing and data. Both of them are Pareto on/ Pareto off processes in their source model. As Go-Back-N ARQ is a kind of continuous ARQ, it guarantees their traffic to be still on/off process in the WCDMA channel. However, the packet retransmissions result in the lengthening of the on period in the WCDMA channel as well as the activity factors for both web-browsing and data services.

Thus, we define $p_{on3,c}$ and $p_{on4,c}$ as the lengthened activity factors of web-browsing and data services in the single-connection system model, while define $p_{on3,i,c}$ and $p_{on4,i,c}$ as the lengthened activity factors of web-browsing and data services within the ith $(1 \le i \le N)$ mobile user in the multi-connection system model. The objective of this section is to calculate the lengthened activity factors in both system models.

As explained in Chapter 3, the retransmission probability is required to address the lengthened activity factors in GBN ARQ system. In the WCDMA system, non-real-time services, including web-browsing and data, are retransmitted in case of outage. That is, when an outage occurs to these two services, the mobile user retransmits the web-browsing packet and the data packet with the Go-Back-N ARQ scheme. Thus, the retransmission probabilities of the two services are equal to the respective outage probabilities for them. Since the outage probability of each class is fluctuating over time, it is necessary to obtain the instantaneous lengthened activity factors of web-browsing and data services first, and then to sum up all possible instantaneous lengthened activity factors to derive the average lengthened activity factors for both system models. In order to obtain $p_{on3,c}$, $p_{on4,c}$, $p_{on3,i,c}$ and $p_{on4,i,c}$, the necessary parameters are listed in the following.

- Let B_3 and B_4 be the buffer sizes of web-browsing and data services expressed in terms of the number of packets, respectively.
- Let T_3 and T_4 be the packet duration of web-browsing and data services, respectively.
- Let T_{3a} and T_{4a} be the packet acknowledgement time of web-browsing and data services, respectively.

- Let M_{re3} and M_{re4} be the maximum number of retransmissions for web-browsing and data services, respectively.
- As given in Chapter 2, $f_3(t) = c_{3,on} a_{3,on}^{\ c_{3,on}} t^{-c_{3,on}-1},\ t \ge a_{3,on}$ and $f_4(t) = c_{4,on} a_{4,on}^{\ c_{4,on}} t^{-c_{4,on}-1},\ t \ge a_{4,on}$ are probability function of the Pareto on period of web-browsing service and data service, respectively, while $g_3(t) = c_{3,off} a_{3,off}^{\ c_{3,off}} t^{-c_{3,off}-1},\ t \ge a_{3,off}$ and $g_4(t) = c_{4,off} a_{4,off}^{\ c_{4,off}} t^{-c_{4,off}-1},\ t \ge a_{4,off}$ are probability density functions of the Pareto off period of web-browsing and data services, respectively.

4.5.1 Lengthened Activity Factor in Single-Connection System Model

The average lengthened activity factors of the web-browsing and data services in the single-connection system model are $p_{on3,c}$ and $p_{on4,c}$, respectively. In Chapter 3, we have generalized the expression of the lengthened activity factors in the WCDMA channel for a non-real-time service, as in equation (3.20). When the instantaneous outage probability is varying, we can formulate the average lengthened activity factors of web-browsing and data services in the WCDMA channel of the single-connection system model by equations (4.42) and (4.43). `

$$
\begin{aligned}
p_{on3,c} = &\sum_{l_1=0}^{N_1} \sum_{l_{2h}=0}^{N_2} \sum_{l_{2l}=0}^{MN_2} \sum_{l_3=0}^{N_3-1} \sum_{l_4=0}^{N_4} AfFun(T_3, T_{3a}, B_3, c_{3,on}, a_{3,on}, c_{3,off}, a_{3,off}, Q(\frac{\delta_3 - \mu_3}{\sigma_3}), M_{re3}) \\
&\binom{N_1}{l_1} (p_{on1})^{l_1} (1 - p_{on1})^{N_1 - l_1} \binom{N_2}{l_{2h}} (p_{on2h})^{l_{2h}} (1 - p_{on2h})^{N_2 - l_{2h}} \binom{MN_2}{l_{2l}} (p_{on2l})^{l_{2l}} (1 - p_{on2l})^{MN_2 - l_{2l}} \\
&\binom{N_3 - 1}{l_3} (p_{on3,c})^{l_3} (1 - p_{on3,c})^{(N_3-1)-l_3} \binom{N_4}{l_4} (p_{on4,c})^{l_4} (1 - p_{on4,c})^{N_4 - l_4} \}
\end{aligned}
\tag{4.42}
$$

$$p_{on4,c}=\sum_{l_1=0}^{N_1}\sum_{l_{2h}=0}^{N_2}\sum_{l_{2l}=0}^{MN_2}\sum_{l_3=0}^{N_3}\sum_{l_4=0}^{N_4-1}\{AfFun(T_4,T_{4a},B_4,c_{4,on},a_{4,on},c_{4,off},a_{4,off},Q(\frac{\delta_4-\mu_4}{\sigma_4}),M_{re4})$$
$$\binom{N_1}{l_1}(p_{on1})^{l_1}(1-p_{on1})^{N_1-l_1}\binom{N_2}{l_{2h}}(p_{on2h})^{l_{2h}}(1-p_{on2h})^{N_2-l_{2h}}\binom{MN_2}{l_{2l}}(p_{on2l})^{l_{2l}}(1-p_{on2l})^{MN_2-l_{2l}}$$
$$\binom{N_3}{l_3}(p_{on3,c})^{l_3}(1-p_{on3,c})^{N_3-l_3}\binom{N_4-1}{l_4}(p_{on4,c})^{l_4}(1-p_{on4,c})^{(N_4-1)-l_4}\}, \tag{4.43}$$

where the definitions of δ_3, μ_3, σ_3, δ_4, μ_4 and σ_4 are given by equations (4.32)-(4.35).

4.5.2 Lengthened Activity Factor in Multi-Connection System Model

The case in the multi-connection system model is different from that in the single-connection system model. The average lengthened activity factors of web-browsing and data services are different within different mobile users. We define the lengthened activity factors within the *i*th mobile user as $p_{on3,i,c}$ and $p_{on4,i,c}$, respectively. We have generalized the expression of the lengthened activity factors with equation (3.20). When the instantaneous outage probability is varying, we can formulate the average lengthened activity factors of the web-browsing and data services by equations (4.44) and (4.45).

$$p_{on3,i,c}=\sum_{l_{1,1}=0}^{n_{1,1}}\sum_{l_{1,2l}=0}^{Mn_{1,2}}\sum_{l_{1,2h}=0}^{n_{1,2}}\sum_{l_{1,3}=0}^{n_{1,3}}\sum_{l_{1,4}=0}^{n_{1,4}}\cdots\sum_{\substack{l_{j,1}=0\\ j\neq i}}^{n_{j,1}}\sum_{\substack{l_{j,2l}=0\\ j\neq i}}^{Mn_{j,2}}\sum_{\substack{l_{j,2h}=0\\ j\neq i}}^{n_{j,2}}\sum_{\substack{l_{j,3}=0\\ j\neq i}}^{n_{j,3}}\sum_{\substack{l_{j,4}=0\\ j\neq i}}^{n_{j,4}}\cdots\sum_{l_{N,1}=0}^{n_{N,1}}\sum_{l_{N,2l}=0}^{Mn_{N,2}}\sum_{l_{N,2h}=0}^{n_{N,2}}\sum_{l_{N,3}=0}^{n_{N,3}}\sum_{l_{N,4}=0}^{n_{N,4}}$$
$$\{AfFun(T_3,T_{3a},B_3,c_{3,on},a_{3,on},c_{3,off},a_{3,off},Q(\frac{\delta_{i,3}-\mu_i}{\sigma_i}),M_{re3})\times\prod_{\substack{j=1\\ j\neq i}}^{N}\binom{n_{j,1}}{l_{j,1}}(p_{on1})^{l_{j,1}}(1-p_{on1})^{n_{j,1}-l_{j,1}}$$
$$\times\binom{Mn_{j,2}}{l_{j,2l}}(p_{on2l})^{l_{j,2l}}(1-p_{on2l})^{Mn_{j,2}-l_{j,2l}}\binom{n_{j,2}}{l_{j,2h}}(p_{on2h})^{l_{j,2h}}(1-p_{on2h})^{n_{j,2}-l_{j,2h}}$$
$$\times\binom{n_{j,3}}{l_{j,3}}(p_{on3,i,c})^{l_{j,3}}(1-p_{on3,i,c})^{n_{j,3}-l_{j,3}}\binom{n_{j,4}}{l_{j,4}}(p_{on4,i,c})^{l_{j,4}}(1-p_{on4,i,c})^{n_{j,4}-l_{j,4}}\} \tag{4.44}$$

$$p_{on4,i,c} = \sum_{l_{1,1}=0}^{n_{1,1}} \sum_{l_{1,2l}=0}^{Mn_{1,2}} \sum_{l_{1,2h}=0}^{n_{1,2}} \sum_{l_{1,3}=0}^{n_{1,3}} \sum_{l_{1,4}=0}^{n_{1,4}} \cdots \sum_{\substack{l_{j,1}=0 \\ j\neq i}}^{n_{j,1}} \sum_{\substack{l_{j,2l}=0 \\ j\neq i}}^{Mn_{j,2}} \sum_{\substack{l_{j,2h}=0 \\ j\neq i}}^{n_{j,2}} \sum_{\substack{l_{j,3}=0 \\ j\neq i}}^{n_{j,3}} \sum_{\substack{l_{j,4}=0 \\ j\neq i}}^{n_{j,4}} \cdots \sum_{l_{N,1}=0}^{n_{N,1}} \sum_{l_{N,2l}=0}^{Mn_{N,2}} \sum_{l_{N,2h}=0}^{n_{N,2}} \sum_{l_{N,3}=0}^{n_{N,3}} \sum_{l_{N,4}=0}^{n_{N,4}}$$

$$\{AfFun(T_4, T_{4a}, B_4, c_{4,on}, a_{4,on}, c_{4,off}, a_{4,off}, Q(\frac{\delta_{i,4}-\mu_i}{\sigma_i}), M_{re4}) \times \prod_{\substack{j=1 \\ j\neq i}}^{N} \binom{n_{j,1}}{l_{j,1}} (p_{on1})^{l_{j,1}} (1-p_{on1})^{n_{j,1}-l_{j,1}}$$

$$\times \binom{Mn_{j,2}}{l_{j,2l}} (p_{on2l})^{l_{j,2l}} (1-p_{on2l})^{Mn_{j,2}-l_{j,2l}} \binom{n_{j,2}}{l_{j,2h}} (p_{on2h})^{l_{j,2h}} (1-p_{on2h})^{n_{j,2}-l_{j,2h}}$$

$$\times \binom{n_{j,3}}{l_{j,3}} (p_{on3,i,c})^{l_{j,3}} (1-p_{on3,i,c})^{n_{j,3}-l_{j,3}} \binom{n_{j,4}}{l_{j,4}} (p_{on4,i,c})^{l_{j,4}} (1-p_{on4,i,c})^{n_{j,4}-l_{j,4}}\} \tag{4.45}$$

where $\delta_{i,3}$, $\delta_{i,4}$, μ_i and σ_i^2 are given by equations (4.39)-(4.41).

4.5.3 Iteration Method

In equations (4.15-4.19) and (4.29), the desired received power levels are given, based on the proposed power distribution scheme. Additionally, we generalize the outage probabilities with equations (4.31) and (4.38), and calculate the lengthened activity factors of web-browsing and data in the WCDMA channel with equations (4.42-4.45). We can clearly see that the received power levels, lengthened activity factors and outage probabilities of all service are intertwined.

Therefore, an iteration method is developed to obtain the feasible received power levels (S_k , $k \in \{1, 2l, 2h, 3, 4\}$), outage probabilities ($P_{out,k}$, $k \in \{1, 2l, 2h, 3, 4\}$) and lengthened activity factors ($p_{on3,c}$, $p_{on4,c}$) in the single-connection system model, as well as the feasible received power levels ($S_{i,k}$, $1 \le i \le N$, $k \in \{1, 2l, 2h, 3, 4\}$), outage probabilities ($P_{out,i,k}$, $1 \le i \le N$, $k \in \{1, 2l, 2h, 3, 4\}$) and lengthened activity factors ($p_{on3,i,c}$, $p_{on4,i,c}$, $1 \le i \le N$) in the multi-connection system model, satisfying equations (4.15-4.19), (4.29), (4.31), (4.38) and (4.42-4.45) simultaneously.

The following steps are given to derive the feasible S_k, $P_{out,k}$, $p_{on3,c}$ and $p_{on4,c}$, $k \in \{1, 2l, 2h, 3, 4\}$, in the single-connection system model.

1. Set initial $p_{on3,c}$ and $p_{on4,c}$ to be $p_{on3,c} = p_{on3}$ and $p_{on4,c} = p_{on4}$. Compute the received power levels, S_k, with equation (4.15-4.19).
2. Calculate $P_{out,k}$ with equations (4.31) and calculate the lengthened $p_{on3,c}$ and $p_{on4,c}$ with equations (4.42-4.43).
3. The power levels S_k and outage probabilities $P_{out,k}$ are recalculated again using the lengthened $p_{on3,c}$ and $p_{on4,c}$ derived by Step 2.
4. Iterate Step 2-3 until $p_{on3,c}$, $p_{on4,c}$, S_k and $P_{out,k}$ converge. Then, the feasible S_k, $P_{out,k}$, $p_{on3,c}$ and $p_{on4,c}$ are derived.

Similarly, the following are steps derive $P_{out,i,k}$ and $p_{on3,i,c}$ and $p_{on4,i,c}$, $k \in \{1, 2l, 2h, 3, 4\}$, $1 \le i \le N$, in the multi-connection system model.

1. Set initial $p_{on3,i,c}$ and $p_{on4,i,c}$ to be $p_{on3,i,c} = p_{on3}$ and $p_{on4,i,c} = p_{on4}$. Compute the received power levels, $S_{i,k}$, with equations (4.29).
2. Calculate $P_{out,i,k}$ with the derived $S_{i,k}$ and equation (4.38), and calculate the lengthened $p_{on3,i,c}$ and $p_{on4,i,c}$ using equations (4.44-4.45).
3. Based on the lengthened $p_{on3,i,c}$ and $p_{on4,i,c}$, the power levels $S_{i,k}$ and outage probabilities $P_{out,i,k}$ are calculated again.
4. Repeat Step 2-3 until $S_{i,k}$, $p_{on3,i,c}$, $p_{on4,i,4}$ and $P_{out,i,k}$ converge.

The above iteration procedures are used to solve the received power levels, outage probabilities and lengthened activity factors in both single-connection and multi-connection system models.

4.6 Conclusion

This chapter studies the data link layer QoS performances in the WCDMA system. We first proposed two system models, which are the single-connection and multi-connection system models. Then, power distribution algorithms that satisfy the SINR requirements of all services in the WCDMA system are developed for both system models. Next, we formulate the expressions of the outage probabilities for all traffic classes in both system models. At the same time, the lengthened activity factors of web-browsing and data services are given. Because the outage probabilities of all traffic classes and the lengthened activity factors of web-browsing and data services are intertwined, an iteration method is presented to address the stable outage probabilities of all traffic classes and the average lengthened activity factors of web-browsing and data services in the two system models. In Chapter 5, the packet level QoS attributes, such as the packet loss rate and average delay, will be presented based on the results of Chapters 3 and 4.

Chapter 5

Analysis of Packet Level QoS

The packet level is part of the network layer in the WCDMA system. The issues of packet level QoS are mainly about the performances of the individual packets in a data flow and are usually evaluated in terms of average delay and packet loss rate.

As described in 3GPP, the conversational and streaming classes require real-time delivery of packets and a limited level of packet loss rate. In contrast, the interactive and background classes do not require a delay bound but need a much stringent packet loss rate. In the single-connection and multi-connection system models, we develop different mathematical formulas for theoretical predictions of average delay and packet loss rate for each traffic class in the WCDMA system. This chapter is organized as follows.

In section 5.1, the packet loss rate performance is analyzed for each traffic class in the two system models. In section 5.2, the average delay performance is analyzed for each traffic class in two the system models. Finally, section 5.3 concludes this chapter.

5.1 Packet Loss Rate Performance

Packet loss rate is one of the main concerns at the packet level. Each class is subject to a specific packet loss rate requirement given by [12].

In the WCDMA system, multiple access interference in the network may result in bit errors in packet transmissions. If the detected BER at the base station exceeds the predetermined level, outage occurs. Since outage is caused by the fluctuation of the interference, we assumed that the intracell interference and lognormal distributed intercell interference are both slow fading so that the interference level remains nearly unchanged over the duration of a packet length. Under this assumption, we can roughly approximate the packet error rate by the outage probability.

For voice and video services, since they are real-time traffic, the erroneous packets are directly discarded and no retransmission is needed. Hence, their packet loss rate is approximately equal to the outage probabilities.

For web-browsing and data services, they are non-real-time traffic and should be treated differently. When a packet received at the base station is erroneous due to outage, the base station sends a negative acknowledgement (NACK) back to the mobile user and requires a retransmission. As we have assumed in previous chapters, Go-Back-N ARQ scheme is implemented, which is studied in greater details in Chapter 3. As both web-browsing and data services have a finite buffer, retransmissions may result in buffer overflow. Besides, an erroneous packet that is unsuccessfully retransmitted for a predefined maximum times is also discarded as packet loss. Therefore, the packet loss rate of web-browsing and data is due to either outage or buffer overflow.

In the following, we will formulate the packet loss rate for all traffic classes in the two different system models.

5.1.1 Packet Loss Rate in the Single-Connection System Model

The packet loss rates for voice, video using low-bit-rate spreading code, video using high-bit-rate spreading code, web-browsing and data services in single-connection system model are denoted by $P_{loss,1}$, $P_{loss,2l}$, $P_{loss,2h}$, $P_{loss,3}$ and $P_{loss,4}$, respectively.

1. Packet loss rate of voice service

As we mentioned in section 5.1, the packet loss rate of voice is approximately equal to its outage probability given by equation (4.31). The $P_{loss,1}$ is given by

$$P_{loss,1} = P_{out,1}. \tag{5.1}$$

2. Packet loss rate of video service

Just like voice service, the packet loss rate of video is also roughly equal to its outage probability. Thus, the packet loss rates of a video service using low-bit-rate spreading code and a video service using high-bit-rate spreading coded are given by equation (4.31).

$$P_{loss,2l} = P_{out,2l}, \tag{5.2}$$

$$P_{loss,2l} = P_{out,2l}. \tag{5.3}$$

3. Packet loss rate of web-browsing service

For web browsing service, Go-Back-N ARQ scheme is used to retransmit erroneous packets. An erroneous packet is retransmitted until it is correctly received or the maximum number of retransmissions is reached. Thereby, its packet loss results from either finite buffer overflow or outage. In Chapter 3, the packet loss rate performance in the Go-Back-N ARQ system is presented. Additionally, Chapter 4 analyzes the outage probability of web-browsing service in the WCDMA channel, which is fluctuating over time. Thus, in order to calculate the average packet loss rate, it is necessary to compute the instantaneous packet loss rate and to sum up all instantaneous packet loss rates to obtain the average packet loss rate. Equation (3.25) gives the packet loss rate with an instantaneous outage probability. Based on equations (3.25) and (4.31), the average packet loss rate of web-browsing service is given by

$$P_{loss,3}=\sum_{l_1=0}^{N_1}\sum_{l_{2h}=0}^{N_2}\sum_{l_{2l}=0}^{MN_2}\sum_{l_3=0}^{N_3-1}\sum_{l_4=0}^{N_4}$$
$$\{PlossFun[T_3,T_{3a},B_3,c_{3,on},a_{3,on},c_{3,off},a_{3,off},Q(\frac{\delta_3-\mu_3}{\sigma_3}),M_{re3}]\binom{N_1}{l_1}(p_{on1})^{l_1}(1-p_{on1})^{N_1-l_1}$$
$$\binom{N_2}{l_{2h}}(p_{on2h})^{l_{2h}}(1-p_{on2h})^{N_2-l_{2h}}\binom{MN_2}{l_{2l}}(p_{on2l})^{l_{2l}}(1-p_{on2l})^{MN_2-l_{2l}} \tag{5.4}$$
$$\binom{N_3-1}{l_3}(p_{on3,c})^{l_3}(1-p_{on3,c})^{N_3-1-l_3}\binom{N_4}{l_4}(p_{on4,c})^{l_4}(1-p_{on4,c})^{N_4-l_4}\}.$$

where δ_3, μ_3 and σ_3 are given by equations (4.32)-(4.36).

4. Packet loss rate of data service

Just like web-browsing service, the packet loss rate of data is due to either outage or buffer overflow. Based on equations (3.25) and (4.31), the average packet loss rate of data service is given by

$$P_{loss,4}=\sum_{l_1=0}^{N_1}\sum_{l_{2h}=0}^{N_2}\sum_{l_{2l}=0}^{MN_2}\sum_{l_3=0}^{N_3}\sum_{l_4=0}^{N_4-1}$$
$$\{PlossFun[T_4,T_{4a},B_4,c_{4,on},a_{4,on},c_{4,off},a_{4,off},Q(\frac{\delta_4-\mu_4}{\sigma_4}),M_{re4}]\binom{N_1}{l_1}(p_{on1})^{l_1}(1-p_{on1})^{N_1-l_1}$$
$$\binom{N_2}{l_{2h}}(p_{on2h})^{l_{2h}}(1-p_{on2h})^{N_2-l_{2h}}\binom{MN_2}{l_{2l}}(p_{on2l})^{l_{2l}}(1-p_{on2l})^{MN_2-l_{2l}} \tag{5.5}$$
$$\binom{N_3}{l_3}(p_{on3,c})^{l_3}(1-p_{on3,c})^{N_3-l_3}\binom{N_4-1}{l_4}(p_{on4,c})^{l_4}(1-p_{on4,c})^{(N_4-1)-l_4}\},$$

where δ_4, μ_4 and σ_4 are given by equations (4.32)-(4.36).

5.1.2 Packet Loss Rate in the Multi-Connection System Model

We need calculate the packet loss rate for each traffic class within each mobile user in the multi-connection system model. Within the *i*th mobile user, let the packet loss rates for voice, video using low-bit-rate spreading code, video using high-bit-rate spreading code, web-browsing, and data services be denoted by $P_{loss,i,1}$, $k=\{1,2l,2h,3,4\}$, respectively.

1. Packet loss rate of voice service within the *i*th mobile user

As real-time service, the packet loss rate of voice service is equal to the outage probability given by equation (4.38). Thus, $P_{loss,i,1}$ is given by

$$P_{loss,i,1} = P_{out,i,1} . \tag{5.6}$$

2. Packet loss rate of video service within the *i*th mobiles user

The packet loss rate of video service is also equal to the outage probability given by equation (4.38).

$$P_{loss,i,2l} = P_{out,i,2l} , \tag{5.7}$$

$$P_{loss,i,2h} = P_{out,i,2h} . \tag{5.8}$$

3. Packet loss rate of web-browsing service within the *i*th mobile user

Similar to the analysis in the single-connection system model, the average packet loss rate of web-browsing service is due to either outage or buffer overflow, and can be calculated with equations (3.25) and (4.38). Hence, $P_{loss,i,3}$ is given by

$$P_{loss,i,3} = \sum_{l_{1,1}=0}^{n_{1,1}} \sum_{l_{1,2l}=0}^{Mn_{1,2}} \sum_{l_{1,2h}=0}^{n_{1,2}} \sum_{l_{1,3}=0}^{n_{1,3}} \sum_{l_{1,4}=0}^{n_{1,4}} \cdots \sum_{\substack{l_{j,1}=0\\ j\neq i}}^{n_{j,1}} \sum_{\substack{l_{j,2l}=0\\ j\neq i}}^{Mn_{j,2}} \sum_{\substack{l_{j,2h}=0\\ j\neq i}}^{n_{j,2}} \sum_{\substack{l_{j,3}=0\\ j\neq i}}^{n_{j,3}} \sum_{\substack{l_{j,4}=0\\ j\neq i}}^{n_{j,4}} \cdots \sum_{l_{N,1}=0}^{n_{N,1}} \sum_{l_{N,2l}=0}^{Mn_{N,2}} \sum_{l_{N,2h}=0}^{n_{N,2}} \sum_{l_{N,3}=0}^{n_{N,3}} \sum_{l_{N,4}=0}^{n_{N,4}}$$
$$\{PlossFun[T_3, T_{3a}, B_3, c_{3,on}, a_{3,on}, c_{3,off}, a_{3,off}, Q(\frac{\delta_{i,3}-\mu_{i,3}}{\sigma_{i,3}}), M_{re3}]$$
$$\prod_{\substack{j=1\\ j\neq i}}^{N} \binom{n_{j,1}}{l_{j,1}} (p_{on1})^{l_{j,1}} (1-p_{on1})^{n_{j,1}-l_{j,1}} \binom{Mn_{j,2}}{l_{j,2l}} (p_{on2l})^{l_{j,2l}} (1-p_{on2l})^{Mn_{j,2}-l_{j,2l}} \tag{5.9}$$
$$\times \binom{n_{j,2}}{l_{j,2h}} (p_{on2h})^{l_{j,2h}} (1-p_{on2h})^{n_{j,2}-l_{j,2h}} \binom{n_{j,3}}{l_{j,3}} (p_{on3,i,c})^{l_{j,3}} (1-p_{on3,i,c})^{n_{j,3}-l_{j,3}}$$
$$\times \binom{n_{j,4}}{l_{j,4}} (p_{on4,i,c})^{l_{j,4}} (1-p_{on4,i,c})^{n_{j,4}-l_{j,4}} \},$$

where $\delta_{i,3}$, $\mu_{i,3}$ and $\sigma_{i,3}$ are given by equation (4.39)-(4.41).

4. Packet loss rate of data service within the *i*th mobiles user

With a similar method, the packet loss rate of data service is based on equations (3.25) and (4.38). $P_{loss,i,4}$ is given by

$$P_{loss,i,4} = \sum_{l_{1,1}=0}^{n_{1,1}} \sum_{l_{1,2l}=0}^{Mn_{1,2}} \sum_{l_{1,2h}=0}^{n_{1,2}} \sum_{l_{1,3}=0}^{n_{1,3}} \sum_{l_{1,4}=0}^{n_{1,4}} \cdots \sum_{\substack{l_{j,1}=0\\ j\neq i}}^{n_{j,1}} \sum_{\substack{l_{j,2l}=0\\ j\neq i}}^{Mn_{j,2}} \sum_{\substack{l_{j,2h}=0\\ j\neq i}}^{n_{j,2}} \sum_{\substack{l_{j,3}=0\\ j\neq i}}^{n_{j,3}} \sum_{\substack{l_{j,4}=0\\ j\neq i}}^{n_{j,4}} \cdots \sum_{l_{N,1}=0}^{n_{N,1}} \sum_{l_{N,2l}=0}^{Mn_{N,2}} \sum_{l_{N,2h}=0}^{n_{N,2}} \sum_{l_{N,3}=0}^{n_{N,3}} \sum_{l_{N,4}=0}^{n_{N,4}}$$
$$\{PlossFun[T_4, T_{4a}, B_4, c_{4,on}, a_{4,on}, c_{4,off}, a_{4,off}, Q(\frac{\delta_4-\mu_4}{\sigma_4}), M_{re4}] \prod_{\substack{j=1\\ j\neq i}}^{N} \binom{n_{j,1}}{l_{j,1}} (p_{on1})^{l_{j,1}} (1-p_{on1})^{n_{j,1}-l_{j,1}}$$
$$\times \binom{Mn_{j,2}}{l_{j,2l}} (p_{on2l})^{l_{j,2l}} (1-p_{on2l})^{Mn_{j,2}-l_{j,2l}} \binom{n_{j,2}}{l_{j,2h}} (p_{on2h})^{l_{j,2h}} (1-p_{on2h})^{n_{j,2}-l_{j,2h}}$$
$$\times \binom{n_{j,3}}{l_{j,3}} (p_{on3,i,c})^{l_{j,3}} (1-p_{on3,i,c})^{n_{j,3}-l_{j,3}} \binom{n_{j,4}}{l_{j,4}} (p_{on4,i,c})^{l_{j,4}} (1-p_{on4,i,c})^{n_{j,4}-l_{j,4}} \}, \tag{5.10}$$

where $\delta_{i,4}$, $\mu_{i,4}$ and $\sigma_{i,4}$ are given by equation (4.39)-(4.41).

5.2 Delay Performance

Based on technical specifications of 3GPP, each QoS class has its own delay requirement. Since voice service and video service are delay-sensitive real-time traffic, they thus do not use ARQ mechanism and their delay are equal to their transmission times. Comparatively, because web-browsing and data services are delay-insensitive non-real-time traffic, they implement Go-Back-N ARQ mechanism and finite buffer. Thus, the delay of web-browsing and data include queuing delay and transmission delay and acknowledgement delay. In the following, we will present the delay performance for each traffic class in both system models. Let L_k, $k \in \{1,2,3,4\}$ denote the packet sizes of a voice packet, a video packet, a web-browsing packet, and a data packet respectively and T_k, $k \in \{1,2l,2h,3,4\}$ denote the packet duration of a voice packet, a video packet using low-bit-rate spreading code, a video packet using high-bit-rate spreading code, a web-browsing packet and a data packet, respectively.

5.2.1 Delay Performance in Single-Connection System Model

The average delay for voice, video using low-bit-rate spreading code, video using high-bit-rate spreading code, web-browsing, and data service in single-connection system model are denoted by D_k, $k = \{1, 2l, 2h, 3, 4\}$, respectively.

1. Delay of voice services

Because the voice delay is equal to its transmission time, D_1 is given by

$$D_1 = T_1 = \frac{L_1 G_1}{W}. \tag{5.11}$$

2. Delay of video services

The delay of a video packet is also equal to its transmission time. As a video service uses M low-bit-rate spreading codes and one high-bit-rate spreading code for transmissions, a video packet is either transmitted using a low-bit-rate spreading code or using a high-bit-rate spreading code. Therefore, the delay of a video packet is equal to the transmission time either at low-bit-rate code channel or at high-bit-rate code channel. Thus, let D_{2l} and D_{2h} denote the video delay using low- and high-bit-rate spreading code, respectively. They are given by

$$D_{2l} = T_{2l} = \frac{L_2 G_{2l}}{W}, \tag{5.12}$$

and

$$D_{2h} = T_{2h} = \frac{L_2 G_{2h}}{W}. \tag{5.13}$$

3. Delay of web-browsing services

Web-browsing services implement the finite buffer and Go-Back-N ARQ method. The delay of a web-browsing service includes queuing delay, retransmission delay and acknowledgement delay. Chapter 3 studies the delay performance in the Go-Back-N ARQ system. The retransmission probability of a web-browsing service is determined by the instantaneous outage probability and varies over time. Thus, the average delay of a web-browsing service can be calculated by summing up all possible instantaneous delay. Equation (3.32) formulates the delay with an instantaneous outage probability. Based on equations (3.32) and (4.31), the average delay of a web-browsing service, D_3, is given by

$$\begin{aligned} D_3 = & \sum_{l_1=0}^{N_1} \sum_{l_{2h}=0}^{N_2} \sum_{l_{2l}=0}^{MN_2} \sum_{l_3=0}^{N_3-1} \sum_{l_4=0}^{N_4} \\ & \{DelayFun[T_3, T_{3a}, B_3, c_{3,on}, a_{3,on}, c_{3,off}, a_{3,off}, Q(\frac{\delta_3 - \mu_3}{\sigma_3}), M_{re3}] \binom{N_1}{l_1} (p_{on1})^{l_1} (1-p_{on1})^{N_1-l_1} \\ & \times \binom{N_2}{l_{2h}} (p_{on2h})^{l_{2h}} (1-p_{on2h})^{N_2-l_{2h}} \binom{MN_2}{l_{2l}} (p_{on2l})^{l_{2l}} (1-p_{on2l})^{MN_2-l_{2l}} \\ & \times \binom{N_3-1}{l_3} (p_{on3,c})^{l_3} (1-p_{on3,c})^{N_3-1-l_3} \binom{N_4}{l_4} (p_{on4,c})^{l_4} (1-p_{on4,c})^{N_4-l_4} \}, \end{aligned} \tag{5.14}$$

where δ_3, μ_3 and σ_3 are given by equations (4.32)-(4.36).

4. Delay of data services

The delay analysis of the data service is similar to that of the web-browsing service. Thus, based on equations (3.32) and (4.31), the average delay of data, D_4, is given by

$$D_4 = \sum_{l_1=0}^{N_1} \sum_{l_{2h}=0}^{N_2} \sum_{l_{2l}=0}^{MN_2} \sum_{l_3=0}^{N_3} \sum_{l_4=0}^{N_4-1}$$

$$\{DelayFun[T_4, T_{4a}, B_4, c_{4,on}, a_{4,on}, c_{4,off}, a_{4,off}, Q(\frac{\delta_4 - \mu_4}{\sigma_4}), M_{re4}] \binom{N_1}{l_1} (p_{on1})^{l_1} (1 - p_{on1})^{N_1 - l_1}$$

$$\times \binom{N_2}{l_{2h}} (p_{on2h})^{l_{2h}} (1 - p_{on2h})^{N_2 - l_{2h}} \binom{MN_2}{l_{2l}} (p_{on2l})^{l_{2l}} (1 - p_{on2l})^{MN_2 - l_{2l}}$$

$$\times \binom{N_3}{l_3} (p_{on3,c})^{l_3} (1 - p_{on3,c})^{N_3 - l_3} \binom{N_4 - 1}{l_4} (p_{on4,c})^{l_4} (1 - p_{on4,c})^{(N_4 - 1) - l_4}\}, \quad (5.15)$$

where δ_4, μ_4 and σ_4 are given by equations (4.32)-(4.36).

5.2.2 Delay Performance in Multi-Connection System Model

The delay performances in the multi-connection system model should be analyzed for each class within each mobile user. Thus, the average delay for voice, video using low-bit-rate spreading code, video using high-bit-rate spreading code, web-browsing, and data service within the *i*th mobile user are denoted by $D_{i,k}$, $k = \{1, 2l, 2h, 3, 4\}$, respectively.

1. Delay of voice services

The voice delay in the multi-connection system model is the same as that in the single-connection system model, because voice service does not use ARQ mechanism. The delay of voice service, $D_{i,1}$, within the *i*th mobile station is given by

$$D_{i,1} = D_1. \quad (5.16)$$

2. Delay of video services

The analysis of the video delay in the multi-connection system model is the same as that in the single-connection system model, as video services do not use ARQ mechanism. The delay of video service using low- and high-bit-rate spreading code within the *i*th mobile user are denoted by $D_{i,2l}$ and $D_{i,2h}$, which are given by

$$D_{i,2l} = D_{2l}, \tag{5.17}$$

and

$$D_{i,2h} = D_{2h}. \tag{5.18}$$

3. Delay of web-browsing services

Similar to the single-connection system model, the delay of a web-browsing service in the multi-connection system model is based on the Go-back-N ARQ and the outage probability. As we know, the outage probability in the multi-connection system model is different from that in the single-connection system model. With equations (3.32) and (4.38), within the *i*th mobile station, the average delay of a web-browsing service, $D_{i,3}$, is given by

$$D_{i,3} = \sum_{l_{1,1}=0}^{n_{1,1}} \sum_{l_{1,2l}=0}^{Mn_{1,2}} \sum_{l_{1,2h}=0}^{n_{1,2}} \sum_{l_{1,3}=0}^{n_{1,3}} \sum_{l_{1,4}=0}^{n_{1,4}} \cdots \sum_{\substack{l_{j,1}=0 \\ j\neq i}}^{n_{j,1}} \sum_{\substack{l_{j,2l}=0 \\ j\neq i}}^{Mn_{j,2}} \sum_{\substack{l_{j,2h}=0 \\ j\neq i}}^{n_{j,2}} \sum_{\substack{l_{j,3}=0 \\ j\neq i}}^{n_{j,3}} \sum_{\substack{l_{j,4}=0 \\ j\neq i}}^{n_{j,4}} \cdots \sum_{l_{N,1}=0}^{n_{N,1}} \sum_{l_{N,2l}=0}^{Mn_{N,2}} \sum_{l_{N,2h}=0}^{n_{N,2}} \sum_{l_{N,3}=0}^{n_{N,3}} \sum_{l_{N,4}=0}^{n_{N,4}}$$

$$\{DelayFun[T_3, T_{3a}, B_3, c_{3,on}, a_{3,on}, c_{3,off}, a_{3,off}, Q(\frac{\delta_{i,3}-\mu_{i,3}}{\sigma_{i,3}}), M_{re3}] \prod_{\substack{j=1 \\ j\neq i}}^{N} \binom{n_{j,1}}{l_{j,1}} (p_{on1})^{l_{j,1}} (1-p_{on1})^{n_{j,1}-l_{j,1}}$$

$$\times \binom{Mn_{j,2}}{l_{j,2l}} (p_{on2l})^{l_{j,2l}} (1-p_{on2l})^{Mn_{j,2}-l_{j,2l}} \binom{n_{j,2}}{l_{j,2h}} (p_{on2h})^{l_{j,2h}} (1-p_{on2h})^{n_{j,2}-l_{j,2h}}$$

$$\times \binom{n_{j,3}}{l_{j,3}} (p_{on3,i,c})^{l_{j,3}} (1-p_{on3,i,c})^{n_{j,3}-l_{j,3}} \binom{n_{j,4}}{l_{j,4}} (p_{on4,i,c})^{l_{j,4}} (1-p_{on4,i,c})^{n_{j,4}-l_{j,4}} \}, \tag{5.19}$$

where $\delta_{i,3}$, $\mu_{i,3}$ and $\sigma_{i,3}$ are given by equations (4.39)-(4.41).

4. Delay of data services

Similarly, the average delay of a data service is based on the outage probability and the Go-back-N ARQ. From equations (3.32) and (4.38), the average delay of a data service, $D_{i,4}$, within the ith mobile user is given by

$$D_{i,4}=\sum_{l_{1,1}=0}^{n_{1,1}}\sum_{l_{1,2l}=0}^{Mn_{1,2}}\sum_{l_{1,2h}=0}^{n_{1,2}}\sum_{l_{1,3}=0}^{n_{1,3}}\sum_{l_{1,4}=0}^{n_{1,4}}\cdots\sum_{\substack{l_{j,1}=0\\ j\neq i}}^{n_{j,1}}\sum_{\substack{l_{j,2l}=0\\ j\neq i}}^{Mn_{j,2}}\sum_{\substack{l_{j,2h}=0\\ j\neq i}}^{n_{j,2}}\sum_{\substack{l_{j,3}=0\\ j\neq i}}^{n_{j,3}}\sum_{\substack{l_{j,4}=0\\ j\neq i}}^{n_{j,4}}\cdots\sum_{l_{N,1}=0}^{n_{N,1}}\sum_{l_{N,2l}=0}^{Mn_{N,2}}\sum_{l_{N,2h}=0}^{n_{N,2}}\sum_{l_{N,3}=0}^{n_{N,3}}\sum_{l_{N,4}=0}^{n_{N,4}}$$

$$\{DelayFun[T_4,T_{4a},B_4,c_{4,on},a_{4,on},c_{4,off},a_{4,off},Q(\frac{\delta_{i,4}-\mu_{i,4}}{\sigma_{i,4}}),M_{re4}]\prod_{\substack{j=1\\ j\neq i}}^{N}\binom{n_{j,1}}{l_{j,1}}(p_{on1})^{l_{j,1}}(1-p_{on1})^{n_{j,1}-l_{j,1}}$$

$$\times\binom{Mn_{j,2}}{l_{j,2l}}(p_{on2l})^{l_{j,2l}}(1-p_{on2l})^{Mn_{j,2}-l_{j,2l}}\binom{n_{j,2}}{l_{j,2h}}(p_{on2h})^{l_{j,2h}}(1-p_{on2h})^{n_{j,2}-l_{j,2h}}$$

$$\times\binom{n_{j,3}}{l_{j,3}}(p_{on3,i,c})^{l_{j,3}}(1-p_{on3,i,c})^{n_{j,3}-l_{j,3}}\binom{n_{j,4}}{l_{j,4}}(p_{on4,i,c})^{l_{j,4}}(1-p_{on4,i,c})^{n_{j,4}-l_{j,4}}\},\qquad(5.20)$$

where $\delta_{i,4}$, $\mu_{i,4}$ and $\sigma_{i,4}$ are given by equation (4.39)-(4.41).

5.3. Conclusion

This chapter analyzes the performances of the packet level QoS attributes, such as the packet loss rate and the average delay, at the network layer. Under our assumption of slow fading, the packet loss of voice and video are totally equal to their respective outage probabilities in the single-connection and multi-connection system models. The average delay of voice and video are just equal to their packet duration times. On the other hand, non-real-time web-browsing and data services use Go-back-N ARQ in their transmissions, which result in lower packet loss rates and longer delays. Based on the analytical results in Chapter 3 and 4, the packet loss rate and average delay performances are analyzed for web-browsing and data services in both system models.

Chapter 6

Numerical Results

In Chapters 4 and 5, the QoS performances at the data link layer and the network layer (packet level) are studied, respectively. The issues in these two chapters consist of the outage probability, the packet loss rate, the average delay and the lengthened activity factors in the WCDMA channel. Mathematical formulas are developed for all traffic classes in both the single-connection and the multi-connection system models. In order to examine the accuracy of the proposed analytical methods, computer simulations are performed in this chapter. The simulation results and analytical results will be presented and compared with graphs. The comparisons prove that our mathematical predictions can approximate the simulation results under light or medium load and the assumptions made in the analysis are reasonable.

In this chapter, the simulation model is presented in section 6.1. In the simulation model, the system parameters are specified and the assumptions are given. In section 6.2, we will present the simulation results to verify that the number of active spreading codes for a non-real-time service, such as web-browsing or data, can be approximated by the binomial distribution given a set of traffic parameters. In section 6.3 and section 6.4, we provide the simulation results for the packet loss rates, the delay, the outage probabilities and the lengthened activity factors for the traffic classes in the single-connection and multi-connection system models, respectively. In section 6.5, we discuss the characteristics of achieved numerical results in the WCDMA system. In section 6.6, a call admission control method is described on the

QoS analytical platform and the corresponding admission regions are obtained for both system models, satisfying the QoS requirements at the packet level of the network layer. Finally, section 6.7 concludes this chapter.

6.1 Simulation Model Specifications

A UMTS cellular mobile network including multiple cells is studied in our simulation model. Each cell is square-shaped and has the same area. Figure 6.1 illustrates the supposed cellular network. All the simulations and analyses are based on this network.

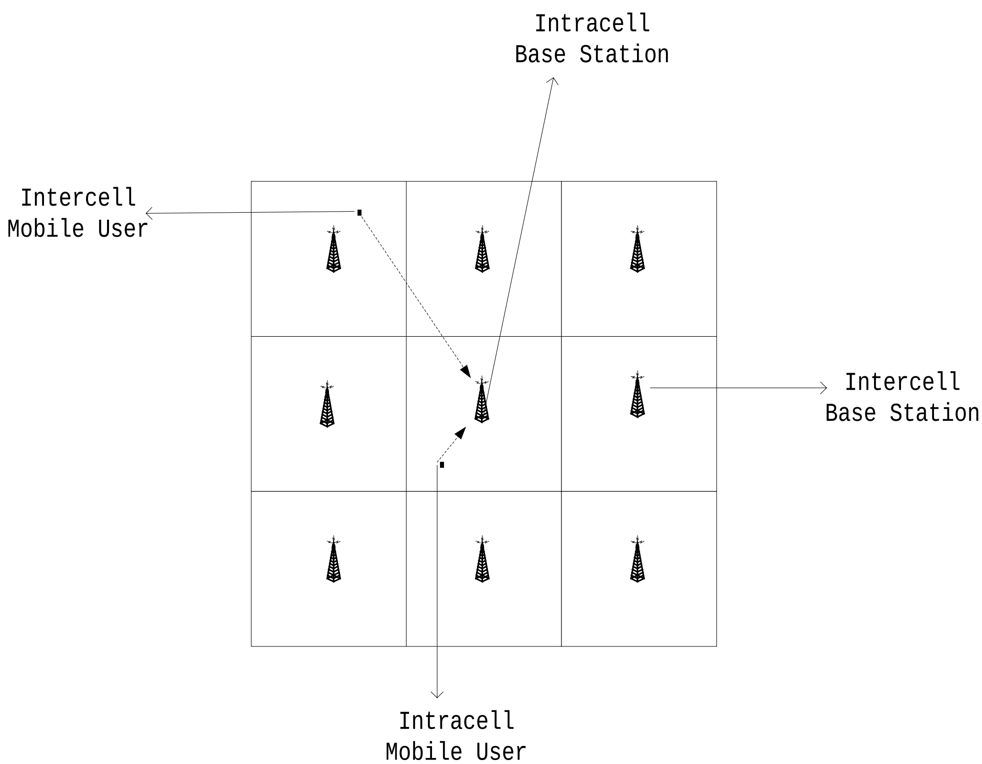

Figure 6.1 Cellular Mobile Network Model

As shown in Figure 6.1, our simulated cellular network consists of nine cells. One base station is located at the center of each cell. We assume that the number of mobile users is the same within each cell and all mobile users are uniformly located in each cell. We make the cell in the middle as the reference cell (intracell). The eight neighbouring cells around the reference cell are considered as intercells. The QoS performances of the mobile users in the intracell are studied. Each mobile user in the intracell is interfered by all other mobile users both in the same cell and in the intercells. We assume that perfect power control is used in each cell such that the desired power levels of all traffic classes in a cell are achieved at their own base stations. We assume that the system is simulated with a constant chip rate of 3.84 Mcps and a bandwidth of 5 MHz based on 3GPP, which are specified in [1]. The simulations are conducted on the SMPL platform [55].

It is assumed that each cell accommodates a number of mobile users, which can serve services such as voice, video, web-browsing and data. Each mobile user in the single-connection system model supports only one service, while each mobile user in the multi-connection system model supports multi-connection multiclass services simultaneously. In the section 4.1 of Chapter 4, the assumptions and system parameters are defined in greater details.

Before we perform the simulation and mathematical analysis, the QoS attributes and requirements of all traffic classes are specified in Table 6.1 based on [12].

Table 6.1 QoS Attributes

QoS Requirements	Real-Time Services		Non-Real-Time Service	
	Voice	Video	Web-browsing	Data
Delay Requirement (ms)	80	250	Nil	Nil
Packet Loss Rate Requirement	10^{-2}	10^{-2}	10^{-3}	10^{-3}
BER Requirement	10^{-2}	10^{-2}	10^{-3}	10^{-3}
SINR Requirement (dB)	2	2	3	3

Besides, the system parameters are configured in Table 6.2, based on [1, 36].

Table 6.2 System Parameters

Parameter Type	Voice	Video	Web-browsing	Data
Spreading Gain	64	128 (Low-bit-rate) 64 (High-bit-rate)	32	16
Number of Spreading codes	1	8 (Low-bit-rate) 1 (High-bit-rate)	1	1
Convolution Rate	1/2	1/2	1/2	1/2
Buffer Size	0	0	200	400
Shadowing Mean μ		0		
Shadowing Variance σ^2		$\sigma = 6\text{dB}$		
Path Loss Attenuation Constant		4		
Number of Cells, n		9		
Power of Thermal Noise Power η		-103.2dBm ($4.8\times10^{-14}Watt$)		
Increased Ratio of Received Powers θ		100		
Modulation Scheme		QPSK		

In Chapter 2, the traffic models are given to approximate all traffic classes. According to [23-24], a set of traffic parameters for each service is presented in Table 6.3.

Table 6.3 Traffic Parameters

Traffic Parameter Type	Real-Time Services		Non-Real-Time Service	
	Voice	Video	Web-browsing	Data
Ave. On Period (second)	1	0.418 (Low) 1.5 (High)	1.6	2.937
Ave. Off Period (second)	1.5	0.663 (Low) 1.5 (High)	12	25.643
Activity Factor	0.4	0.3867 (Low) 0.5 (High)	0.1176	0.1028
Ave. Rate (kbps)	24	122.3	14.1	22.8
Channel Rate (kbps)	60	30 (Low) 60 (High)	120	240
Packet Size (bits)	1200	1800	2400	4800

In generating the traffic for the four different classes, each voice service is generated as exponential on/exponential off process; each video service is generated as M low-bit-rate exponential on/exponential off processes and one high-bit-rate exponential on/exponential off process; each web-browsing or data service is generated as a Pareto on/Pareto off process. According to the traffic models in Chapter 2, the parameters used in the traffic generation are given by the following.

For the traffic model of a voice service given by equation (2.1), α_1 is 0.667 and β_1 is 1.0.

For the traffic model of a video service given by equations (2.2) and (2.3), α is 1.5 and β is 2.4. λ is 1.5 and μ is 1.5.

For the traffic model of a web-browsing service given by equations (2.5) and (2.7), $c_{3,on}$ is 1.1 and $a_{3,on}$ is 0.1455. $c_{3,off}$ is 1.1 and $a_{3,off}$ is 1.1.

For the traffic model of a data service given by equations (2.10) and (2.12), $c_{4,on}$ is 1.1 and $a_{4,on}$ is 0.268. $c_{4,off}$ is 1.1 and $a_{4,off}$ is 2.33.

6.2 Statistical Characteristics of Pareto on/ Pareto off Process

In equations (4.31) and (4.38), the number of active sources for each traffic class is assumed to follow binomial distribution. Generally, this assumption is valid for exponential on/exponential off services, such as voice and video, while it is not valid for Pareto on/Pareto off services, such as web-browsing and data. Therefore, the binomial distribution is only an approximation to simplify the calculation of the outage probabilities. In this section, simulation is performed to examine the accuracy of this approximation under a set of given traffic parameters in Table 6.4.

Table 6.4 Traffic Parameters in Binomial Assumption

Service Type	$c_{3,on}$ ($c_{4,on}$)	$a_{3,on}$ ($a_{4,on}$)	$c_{3,off}$ ($c_{4,off}$)	$a_{3,off}$ ($a_{4,off}$)	$p_{on,3}$ ($p_{on,4}$)
Web-browsing (1)	1.1	0.10	1.1	1.1	0.084
Web-browsing (2)	1.1	0.15	1.1	1.1	0.117
Web-browsing (3)	1.1	0.25	1.1	1.1	0.186
Web-browsing (4)	1.1	0.5	1.1	1.1	0.314
Web-browsing (5)	1.1	0.8	1.1	1.1	0.423
Data (1)	1.1	0.15	1.1	2.33	0.052
Data (2)	1.1	0.27	1.1	2.33	0.108
Data (3)	1.1	0.50	1.1	2.33	0.155
Data (4)	1.1	1.00	1.1	2.33	0.268
Data (5)	1.1	1.80	1.1	2.33	0.398

According to the binomial distribution, if there are N_{user} independent Pareto on/Pareto off services and activity factor is p_{on}, the probability that *i* services are in active state is given by $Pr(i)$.

$$Pr(i) = \binom{N_{user}}{i} p_{on}{}^{i} (1 - p_{on})^{N_{user} - i}$$

We assume there are ten (N_{user} =10) web-browsing (data) services in the system. p_{on} is p_{on3} (p_{on4}) for the web-browsing (data) services. For web-browsing (data) services, the number of active services is denoted by $i\,(0 \le i \le 10)$. The theoretical probabilities that there are i $(0 \le i \le 10)$ active web-browsing (data) services in the system are calculated with the above equation. Strictly, the activity factor and on/off traffic in the WCDMA channel are different from those in the source traffic and thus the following figures for on/off traffic in the source traffic cannot describe the binomial distribution of the active services in the WCDMA channel very accurately. However, because the difference between the source traffic and the traffic in the WCDMA channel is actually not significant, the binomial distribution can approximately represent the number of active web-browsing (data) services in the channel.

In our simulation, we independently generate 10 Pareto on/Pareto off traffic for a given set of traffic parameters (see Table 6.4) for a very long period of time. Statistics on the number of active sources, i $(0 \le i \le 10)$, are collected. The theoretical and simulation results are compared and illustrated in Figure 6.2 and Figure 6.3.

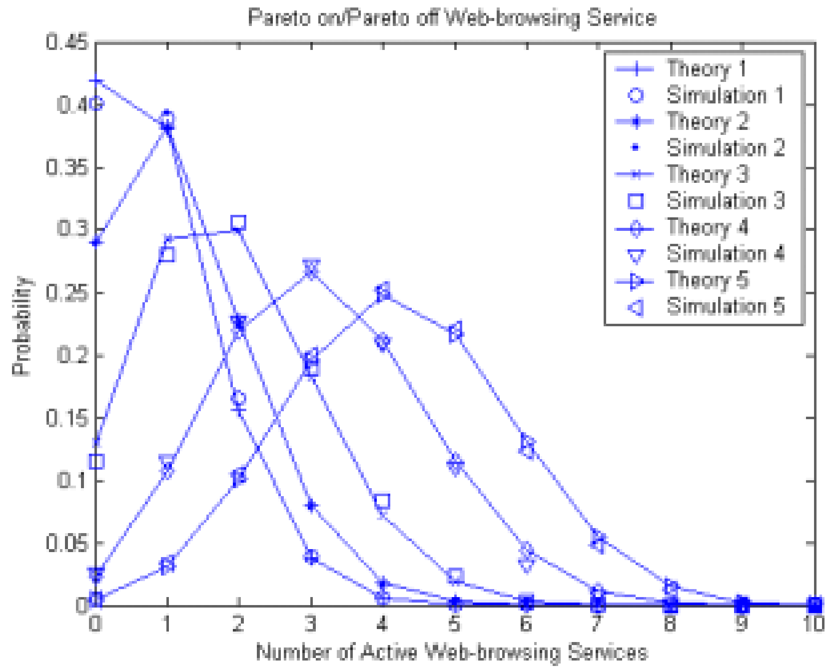

Figure 6.2 Probability Distribution for the Number of Active Spreading Codes Used by Web-browsing Services

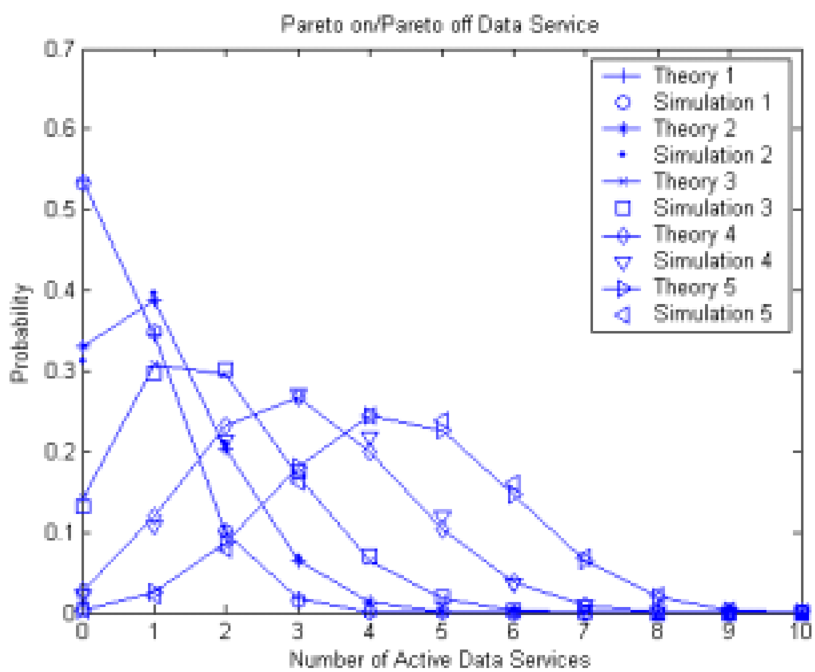

Figure 6.3 Probability Distribution of the Number of Active Spreading Codes Used by Data Services

In Figure 6.2 and Figure 6.3, the distribution of the active web-browsing and data sources are presented, respectively. According to Table 6.4, we fix $c_{3,on}$ ($c_{4,on}$), $c_{3,off}$

($c_{4,off}$) and $a_{3,off}$ ($a_{4,off}$) values, and change $a_{3,on}$ ($a_{4,on}$) to be five different values so that the activity factors of the web-browsing and data services vary accordingly. Thus, five curves corresponding to the five sets of parameters given in Table 6.4 are plotted for web-browsing and data services. These simulations prove that binomial assumption is suitable to approximate the number of active Pareto on/Pareto off sources for these sets of traffic parameters. The subsequent numerical computation in this chapter will be using these sets of values.

6.3 Numerical Results in the Single-Connection System Model

Each mobile user only transmits one service in the single-connection system model. Among the N mobile users in a cell, we assume there are N_1 voice services, N_2 video services, N_3 web browsing services and N_4 data services. All the parameters are defined by the system model in Section 4.1.1. The following Table 6.5 gives the number of services in a cell.

Table 6.5 Number of Services in the Single-Connection System Model

N_1	N_2	N_3	N_4
11~30	4	4	4

We vary the number of voice services from 11 to 30 in this simulation, and obtain the QoS performances, such as the packet loss rate, the average delay, the outage probability and the lengthened activity factors, of the services in the system.

6.3.1 Quality of Service for Voice Services

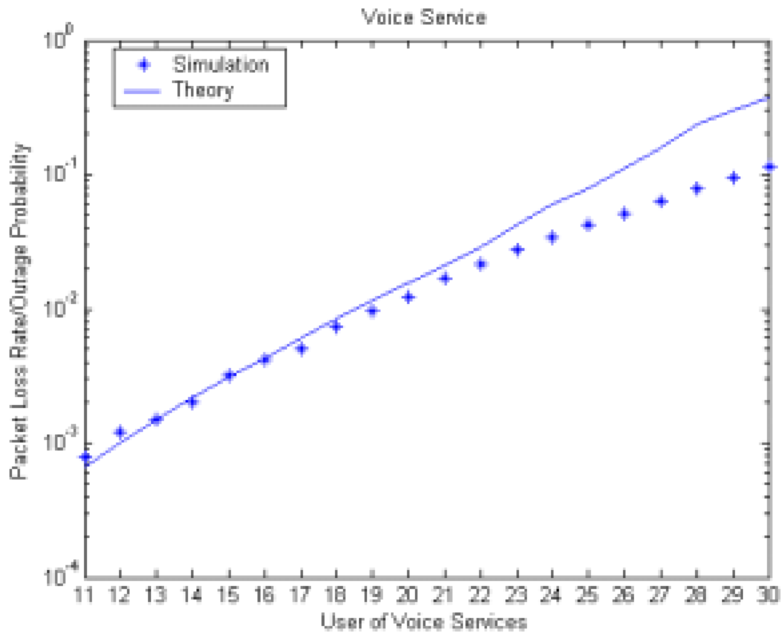

Figure 6.4 Packet Loss Rate/Outage Probability of Voice Services (in the Single-Connection System Model)

As given by Figure 6.4, when the number of voice services is varying from eleven to thirty, packet loss rate and outage probability of voice increases.

6.3.2 Quality of Service for Video Services

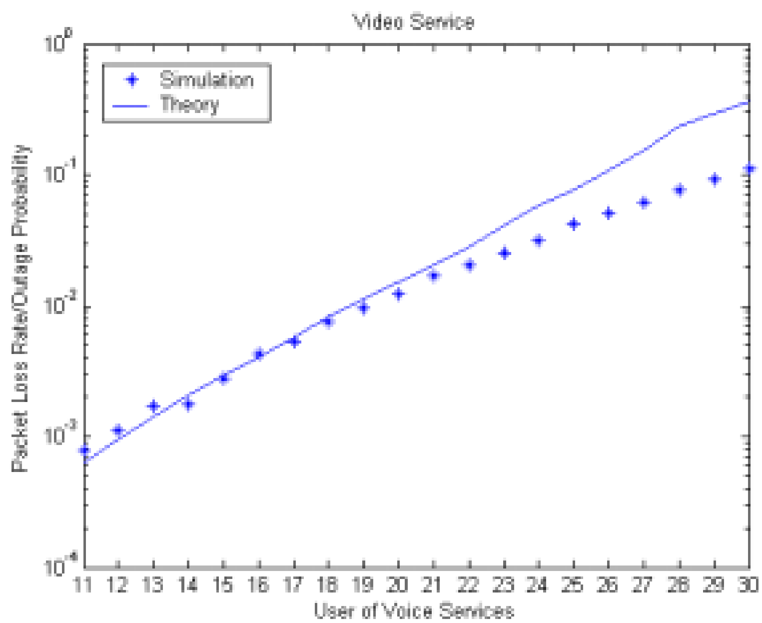

Figure 6.5 Packet Loss Rate / Outage Probability of Video Services (in the Single-Connection System Model)

In Figure 6.5, as we have assumed SINR requirements, BER requirements and packet loss rate requirements are the same for both video services using both low- and high-bit-rate spreading codes, the achieved outage probability and packet loss rate are thus the same for low-bit-rate and high-bit-rate video services.

6.3.3 Quality of Service for Web-browsing Services

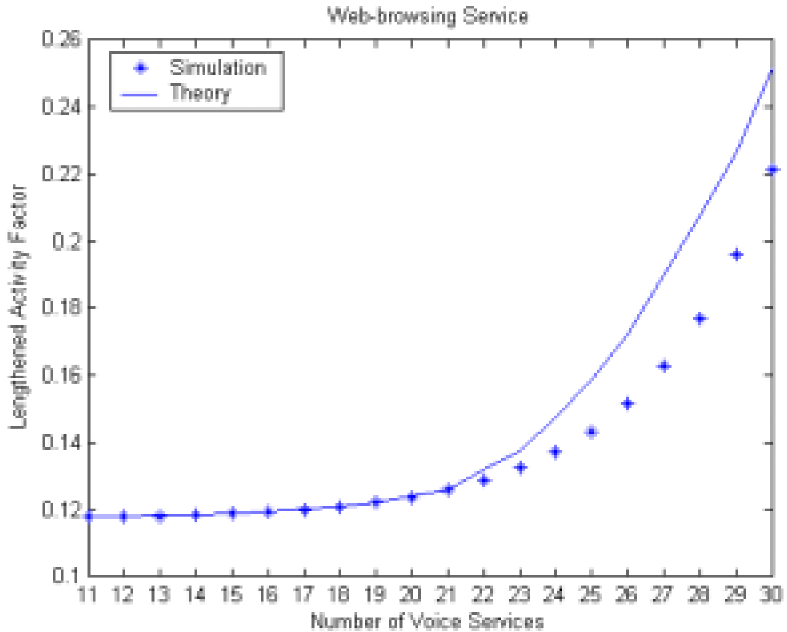

Figure 6.6 Lengthened Activity Factor of Web-browsing Services (in the Single-Connection System Model)

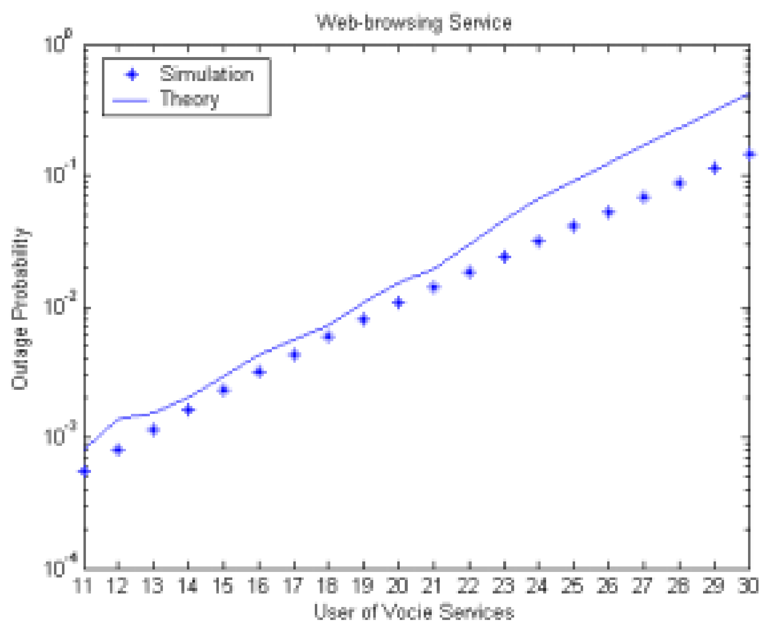

Figure 6.7 Outage Probability of Web-browsing Services (in the Single-Connection System Model)

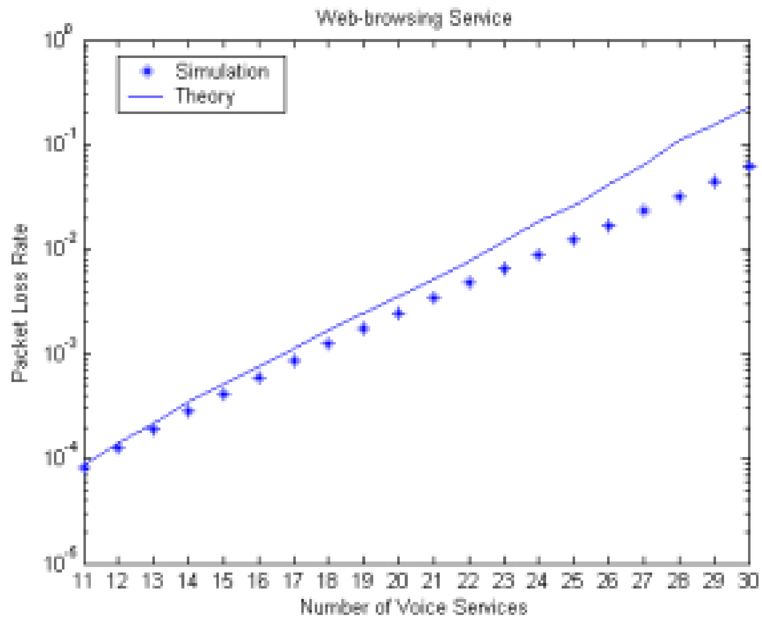

Figure 6.8 Packet Loss Rate of Web-browsing Services (in the Single-Connection System Model)

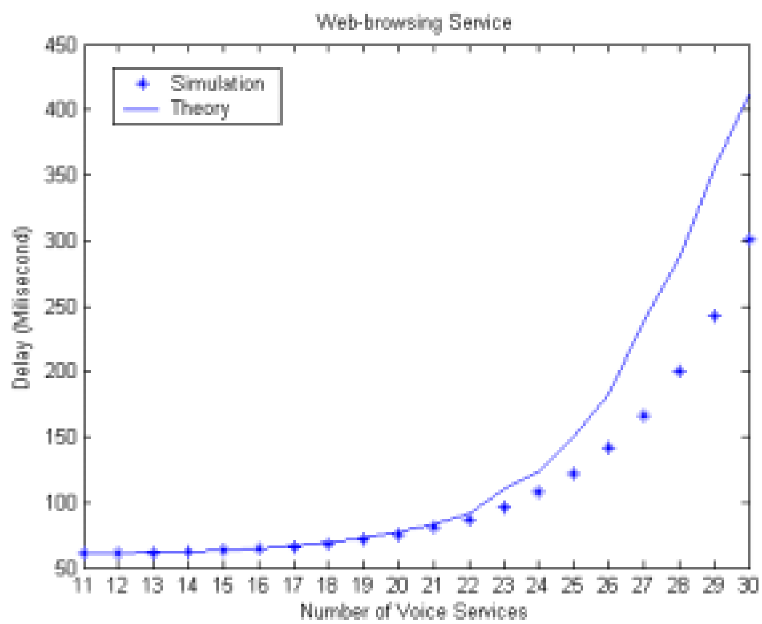

Figure 6.9 Average Delay of Web-browsing Services (in the Single-Connection System Model)

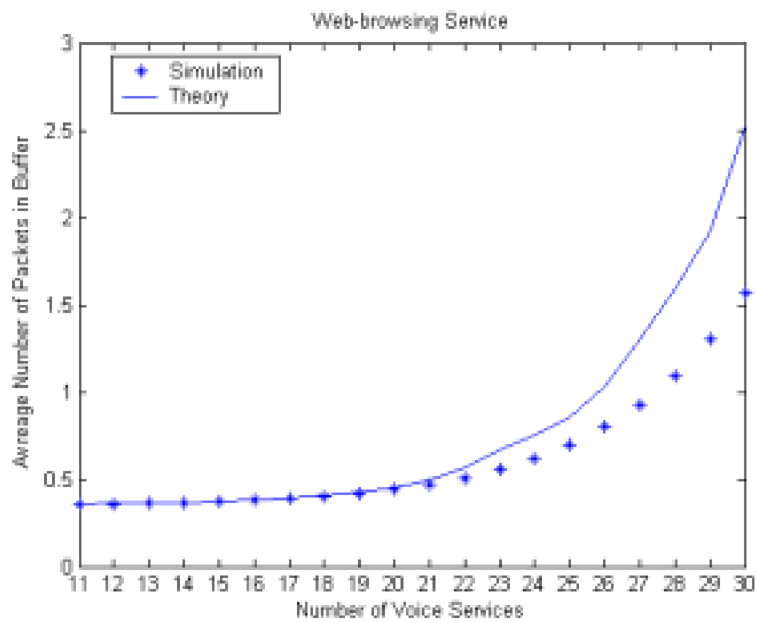

Figure 6.10 Average Number of Web-browsing Packets in the Buffer (in the Single-Connection System Model)

From Figures 6.6 to 6.10, QoS performances of web-browsing services are shown. As web-browsing services implement Go-Back-N ARQ mechanism, the activity factor of a web-browsing service is lengthened shown in Figure 6.6. At the same time, the packet loss rate can be reduced and is lower than its outage probability, as illustrated in Figure 6.7 and 6.8. However, Go-Back-N ARQ will result in a longer delay in Figure 6.9.

6.3.4 Quality of Service for Data Services

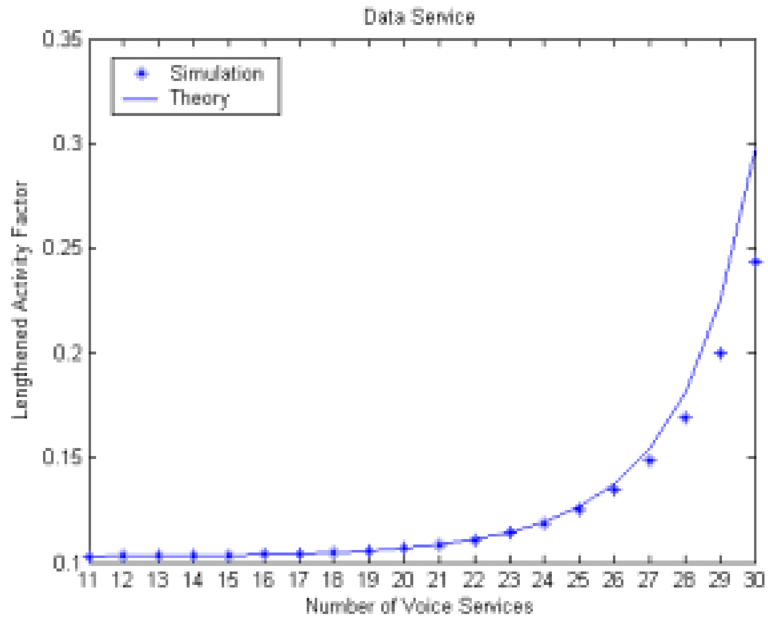

Figure 6.11 Lengthened Activity Factor of Data Services (in the Single-Connection System Model)

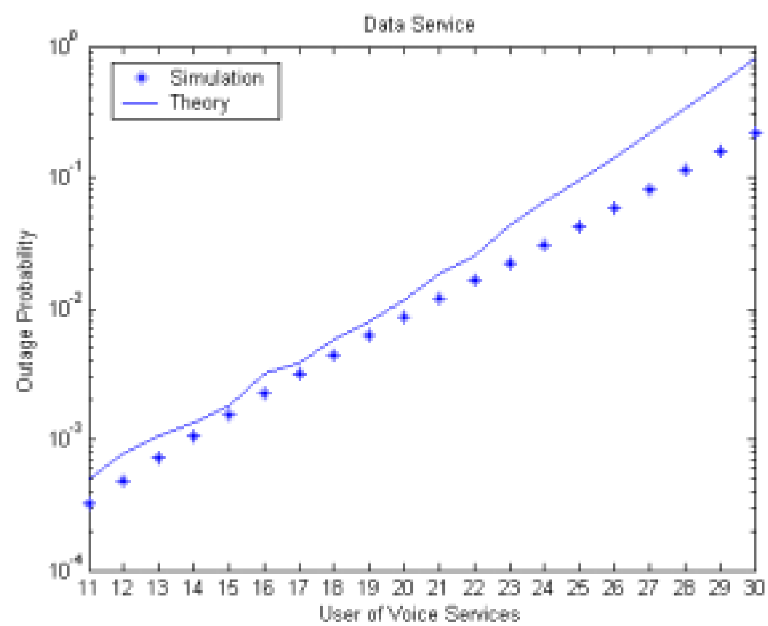

Figure 6.12 Outage Probability of Data Services (in the Single-Connection System Model)

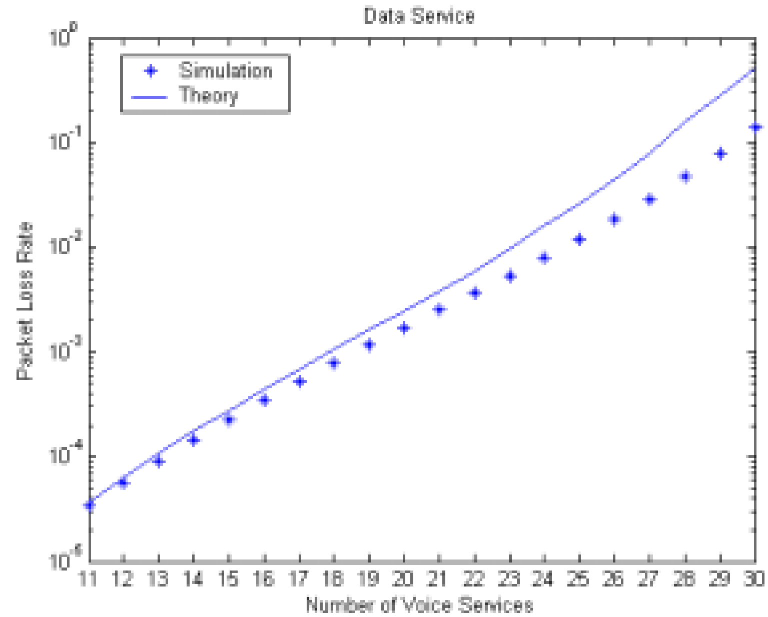

Figure 6.13 Packet Loss Rate of Data Services (in the Single-Connection System Model)

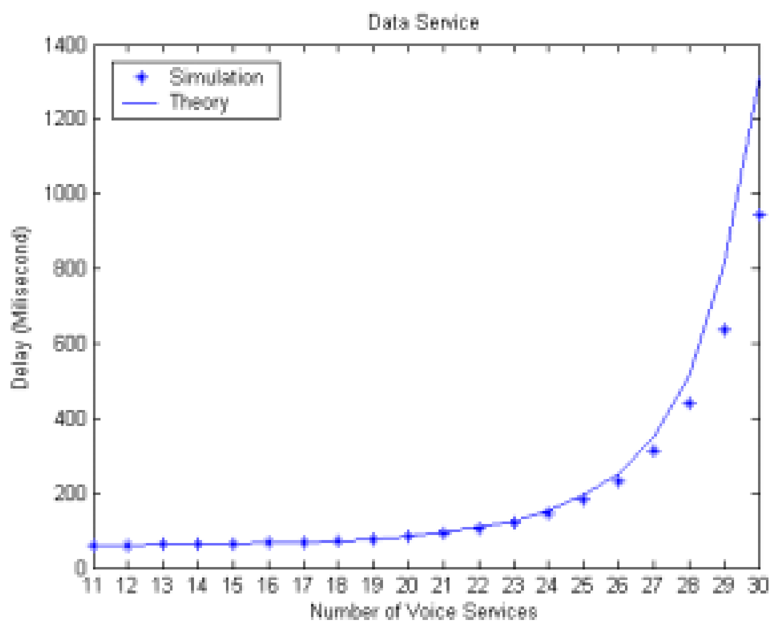

Figure 6.14 Average Delay of Data Services (in Single-Connection System Model)

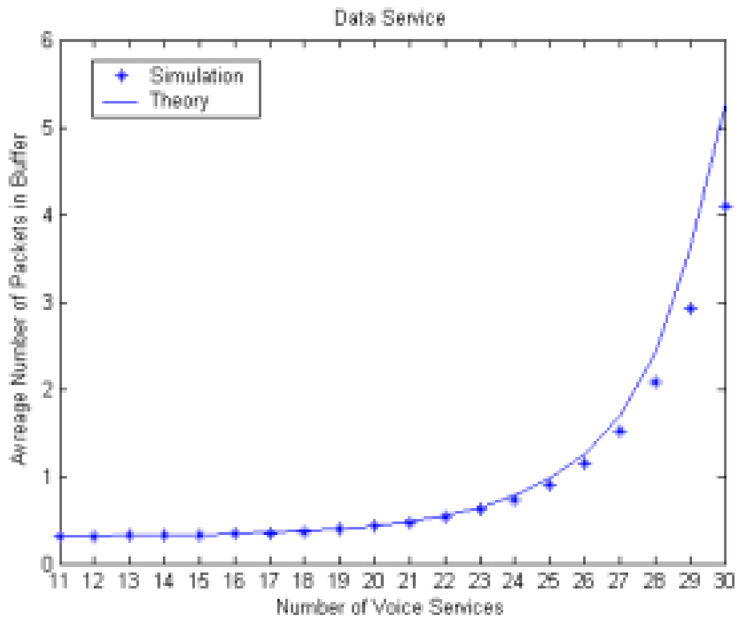

Figure 6.15 Average Number of Data packets in the Buffer (in the Single-Connection System Model)

From Figures 6.11 to 6.15, the QoS performances of data services are shown. Data services are generally similar to web-browsing services. Go-Back-N ARQ is implemented and this leads to a lengthened activity factor, a lower packet loss rate and a longer delay.

6.4 Numerical Results in the Multi-Connection System Model

Multi-connection multiclass services in the multi-connection system model can be transmitted within each mobile station. Among the *N* mobile users in a cell, each user can support different combination of services. In order to simplify the analysis, we can divide all mobile users into four groups. In the following, Table 6.6 lists the service combination and the number of mobile users within each group.

Table 6.6 Number of Mobile Users and Services in the Multi-Connection System Model

Group Index	Group 1	Group 2	Group 3	Group 4
Num. of Mobile Users	5 ~ 23	2	2	5
Num. of Voice Services / Mobile User	1	0	1	0
Num. of Video Services / Mobile User	0	1	1	0
Num. of Web Services / Mobile User	0	0	0	1
Num. of Data Services / Mobile User	0	0	0	1

6.4.1 Quality of Service Performances in Group One

Since each mobile user only serves one voice service, the packet loss rate and outage probability of voice services in group one are same and shown in Figure 6.16.

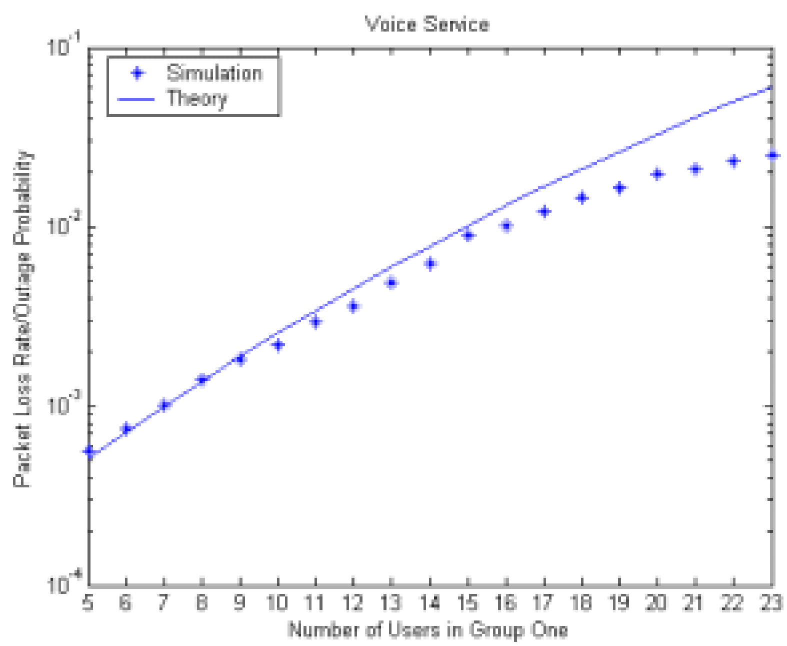

Figure 6.16 Packet Loss Rate/Outage Probability of Voice Services (Group 1, in the Multi-Connection System Model)

6.4.2 Quality of Service Performances in Group Two

Each mobile user only serves one video service. Thus, the packet loss rate and outage probability of video services in group two are same and are given by Figure 6.17.

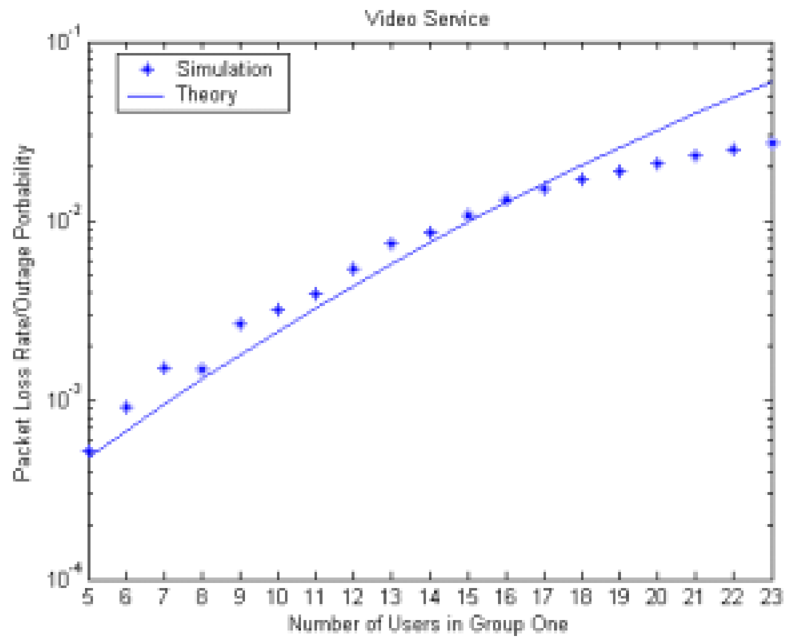

Figure 6.17 Packet Loss Rate/Outage Probability of Video Services (Group 2, in the Multi-Connection System Model)

6.4.3 Quality of Service Performances in Group Three

Each mobile user in group three contains one voice and one video service. Packet loss rate/outage probability of voice and video services are given by Figures 6.18 and 6.19, respectively.

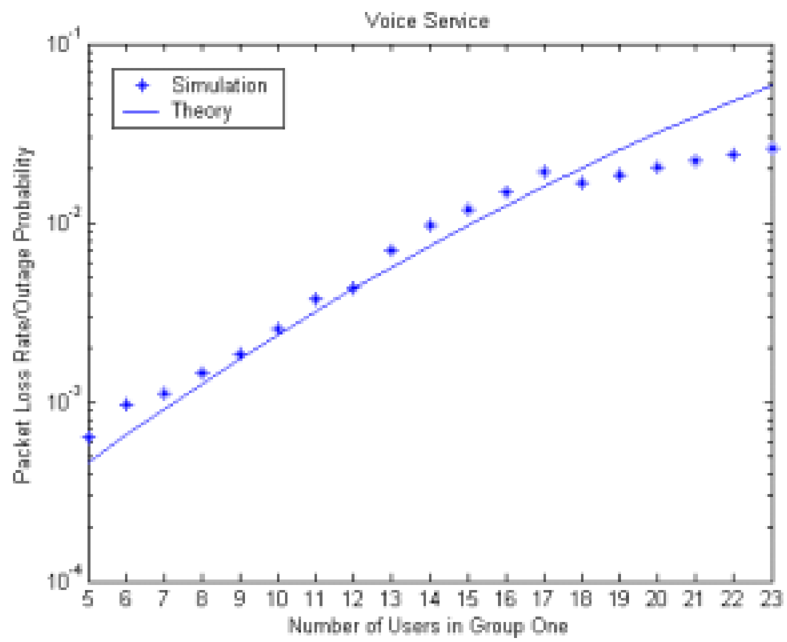

Figure 6.18 Packet Loss Rate/Outage Probability of Voice Services (Group 3, in the Multi-Connection System Model)

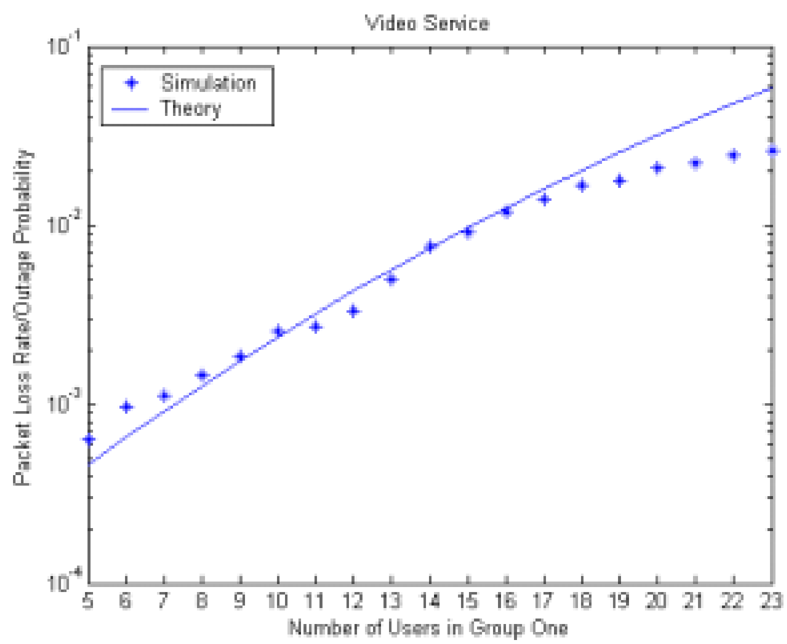

Figure 6.19 Packet Loss Rate/Outage Probability of Video Services (Group 3, in the Multi-Connection System Model)

6.4.4 Quality of Service Performances in Group Four

Each mobile user in group four serves one web-browsing and one data service. Thus, their lengthened activity factors, outage probabilities, packet loss rates, delays, average number of packets in the buffer are presented by the following figures. From Figures 6.21, 6.22, 6.26 and 6.27, we can see that the achieved packet loss rate of a web-browsing or data service is much lower than its corresponding outage probability. This advantage is due to the Go-Back-N ARQ mechanism and the finite buffer used in their transmissions. However, the buffer results in a longer average delay for these two services, which is shown in Figures 6.23 and 6.28.

1. Web-browsing Services

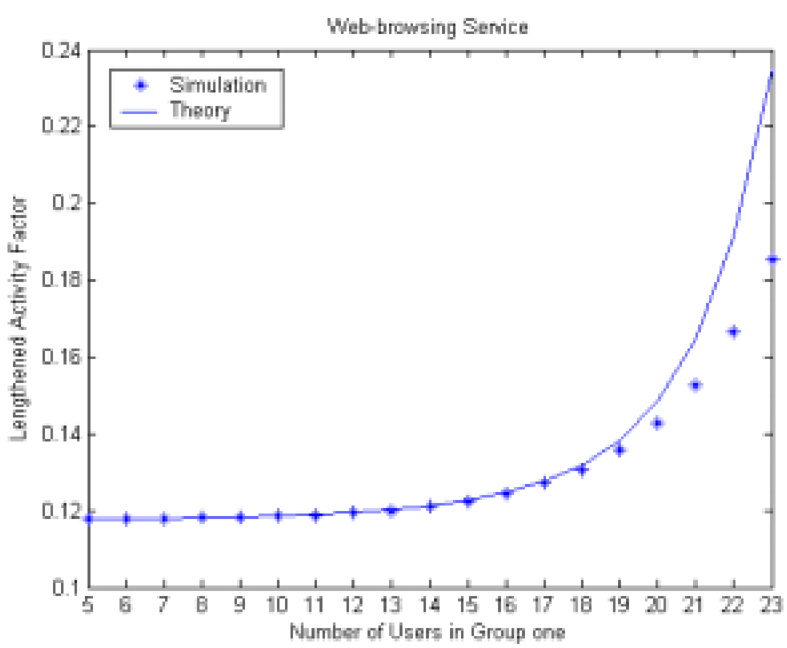

Figure 6.20 Lengthened Activity Factor of Web-browsing Services (Group 4, in the Multi-Connection System Model)

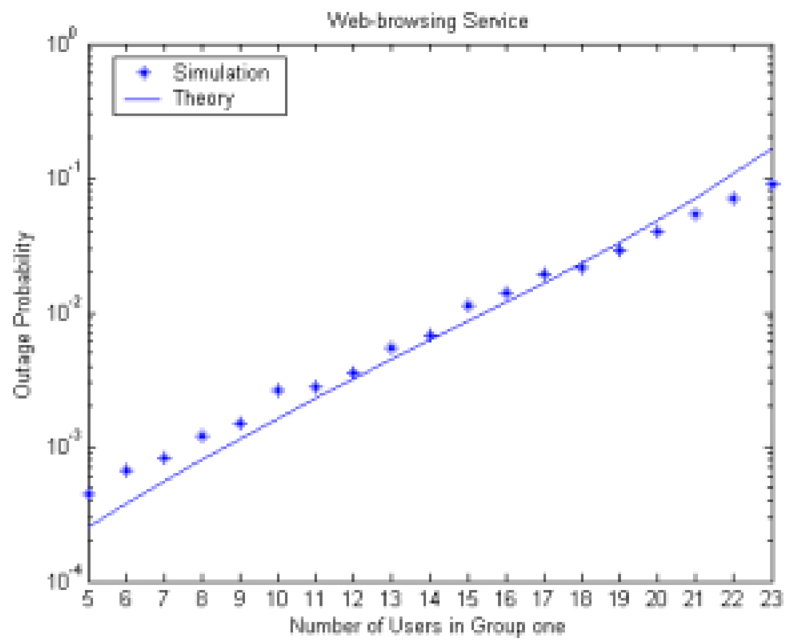

Figure 6.21 Outage Probability of Web-browsing Services (Group 4, in the Multi-Connection System Model)

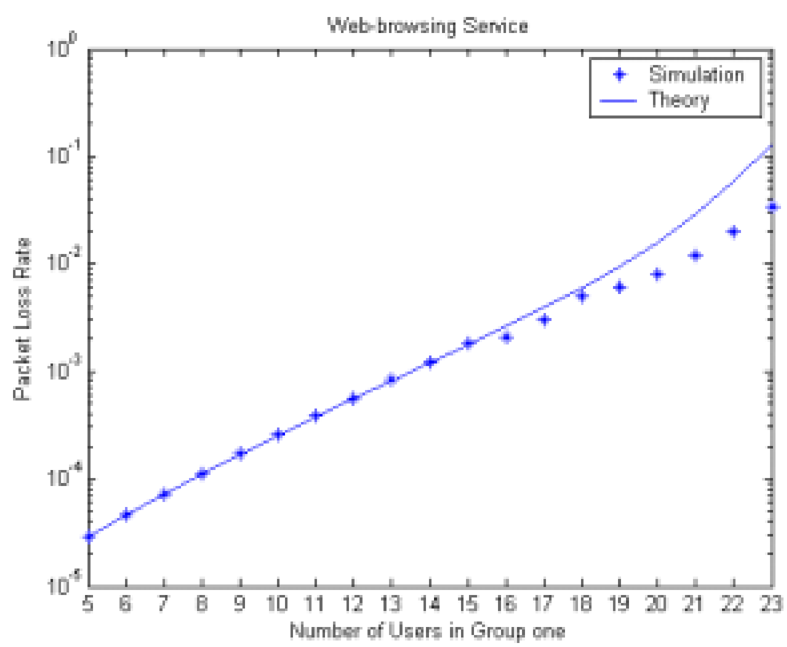

Figure 6.22 Packet Loss Rate of Web-browsing Services (Group 4, in the Multi-Connection System Model)

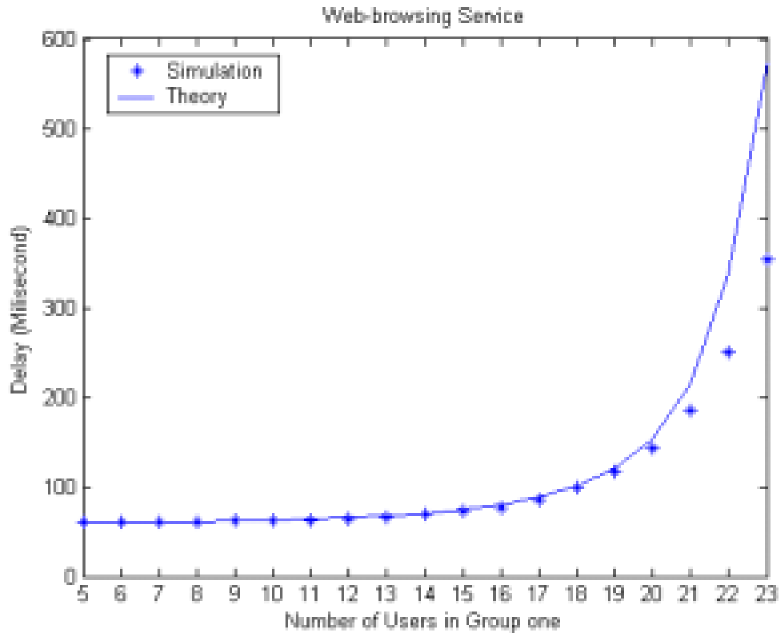

Figure 6.23 Average Delay of Web-browsing Services (Group 4, in the Multi-Connection System Model)

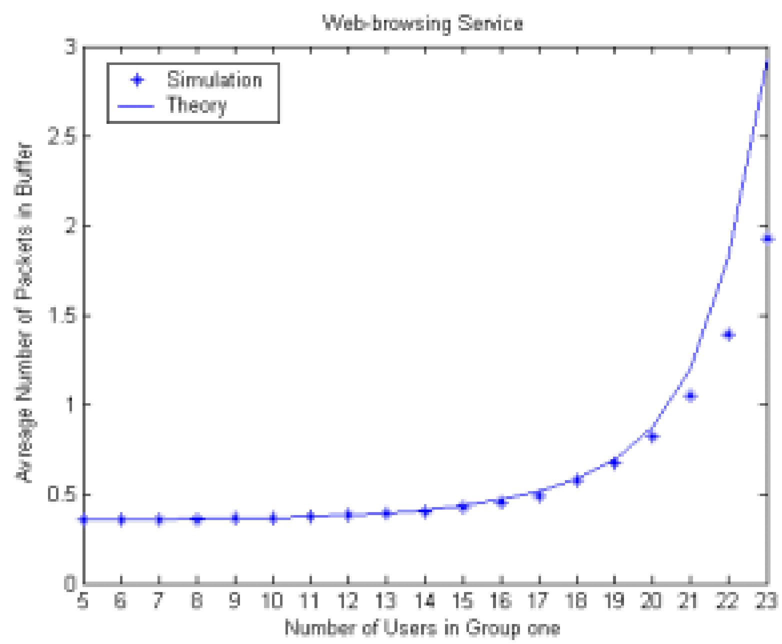

Figure 6.24 Average Number of Web-browsing Packets in the Buffer (Group 4, Multi-Connection System Model)

2. Data Services

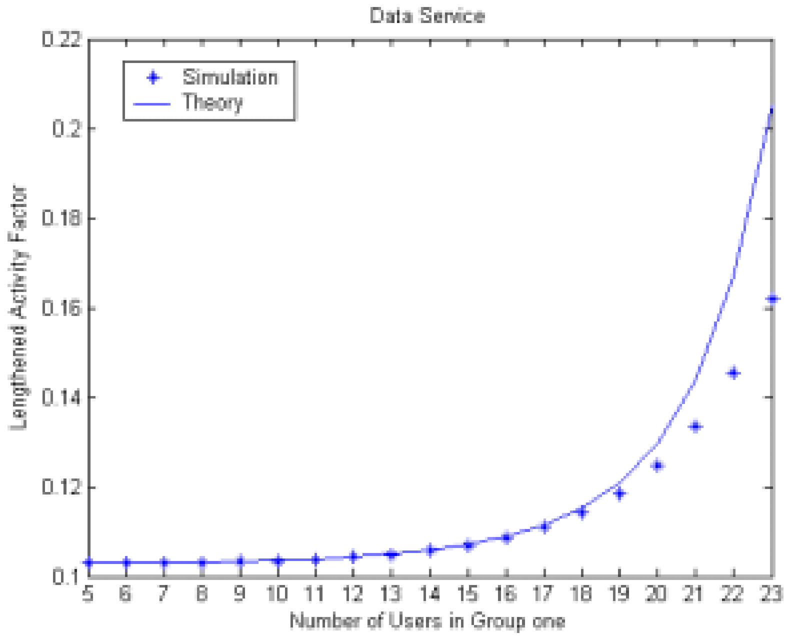

Figure 6.25 Lengthened Activity Factor of Data Services (Group 4, in the Multi-Connection System Model)

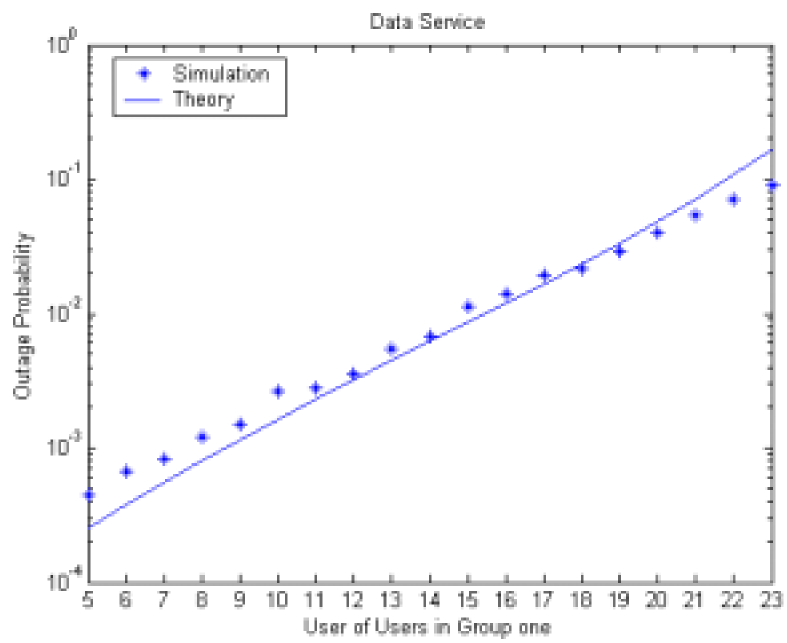

Figure 6.26 Outage Probability of Data Services (Group 4, Multi-Connection System Model)

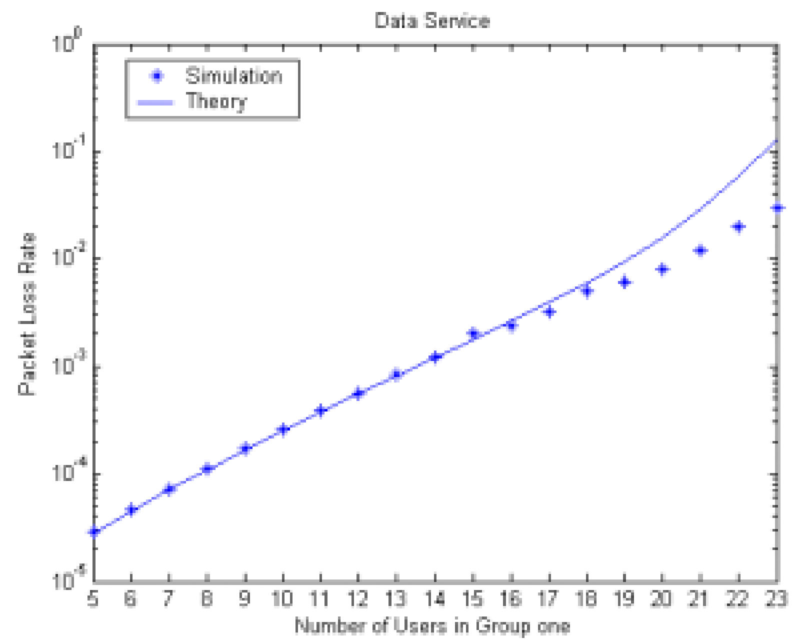

Figure 6.27 Packet Loss Rate of Data Services (Group 4, in the Multi-Connection System Model)

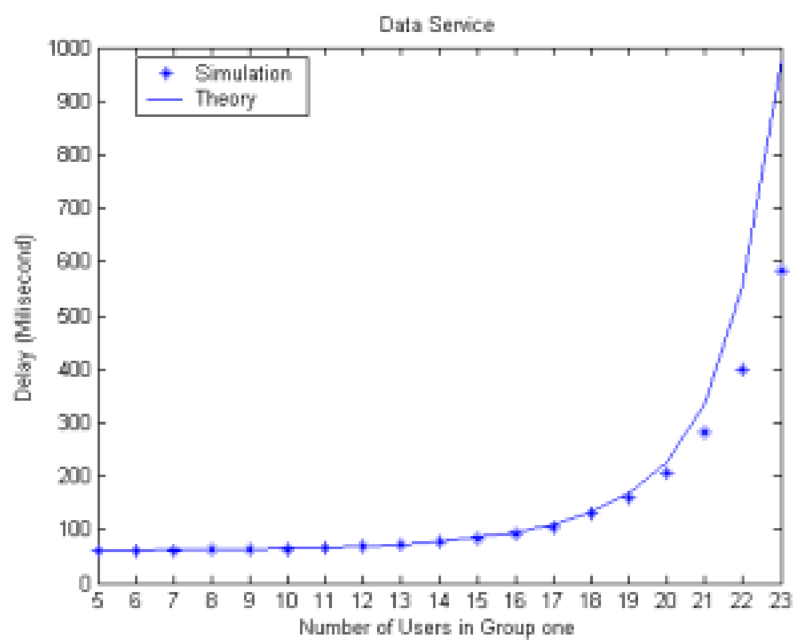

Figure 6.28 Average Delay of Data Services (Group 4, in the Multi-Connection System Model)

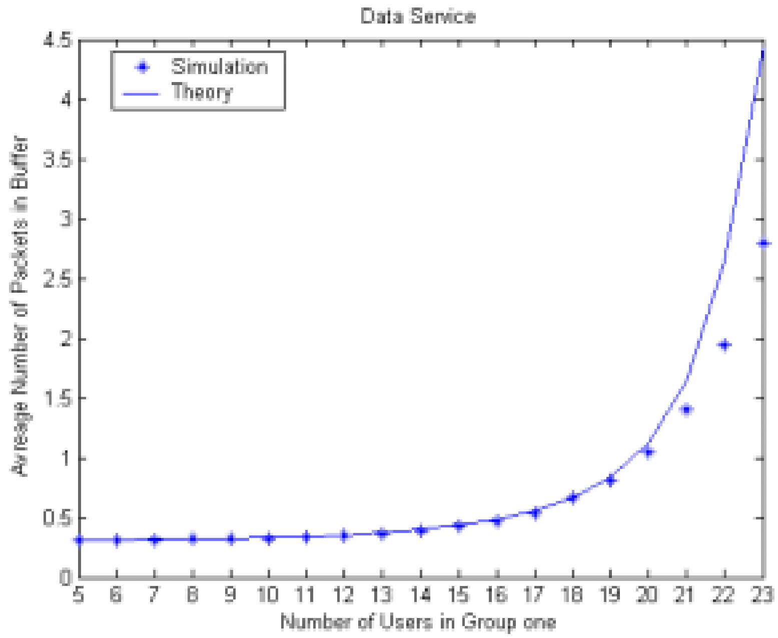

Figure 6.29 Average Number of Data Packets in the Buffer (Group 4, in the Multi-Connection System Model)

6.5 Discussion of Numerical Results

From the numerical results obtained for single- and multi-connection systems, we are able to generalize the following points.

Firstly, we can clearly observe that all analytical results, for examples, the packet loss rates given in Figures 6.4, 6.5, 6.8 and 6.13, and the delays given in Figures 6.9 and 6.14 in the single-connection system, show a better agreement when the systems are in light and medium load than when they are in heavy load. The deviation during heavy load, i.e., when there are more users in the system, can be explained as follows. The outage becomes more severe and thus retransmissions occur more frequently during heavy load. As we mentioned in Chapter 3, our Go-Back-N ARQ analysis is accurate only when the retransmissions occur less frequently and the packet error rate is low. Thus, the average outage probabilities of the web-browsing and data services should be no more than around 5% as shown in Figures 6.7 and 6.12, which is within the range of light and medium load. If a lot of retransmissions happen, the on periods of web-browsing or data services in the WCDMA channel may overlap, which influences the computation of their lengthened activity factors, outage probabilities, packet loss rates and delays. As all classes in a WCDMA system are intertwined to each other, the QoS performances, such as packet loss rates, delays and outage probabilities, of all classes, are therefore affected and deviate from simulation results. Thus, our analytical formulation is only suitable for light and medium load when the throughput of the system is below or around 1.2 Mbps. At higher load, 50%-150% deviation is observed for the packet loss rates shown in Figures 6.8 and 6.13, etc.

Secondly, we are able to achieve a desired feasible region for a given set of QoS requirements. For example, Figures 6.4, 6.5, 6.8, 6.9, 6.13 and 6.14 give the packet loss rate and average delay performances in the single-connection system model. For a given combination of the four groups of users ($N_1 = 18$, $N_2 = 4$, $N_3 = 4$ and $N_4 = 4$), we can see that the QoS requirement for group 1 user can be satisfied from Figure. 6.4. Similarly, from Figure 6.5, 6.8, 6.9 and 6.13 and 6.14, the packet loss rate of delay requirements for group 2, 3, 4, respectively, can be satisfied also. Hence we know that this combination of users ($N_1 = 18$, $N_2 = 4$, $N_3 = 4$ and $N_4 = 4$) is within the admission region.

Thirdly, we also have some comments on the complexity of the analysis. Our final analytical expressions are rather complex since all the equations consist of multiple summations. This is due to the fact that we jointly consider more realistic traffic models, Go-Back-N ARQ, multi-cell network and four traffic classes. This therefore complicates the analysis, especially in a large system. Despite this, the analysis still takes much shorter time to work out the results than using simulation. For example, it takes about 12 hours to obtain the simulation results in the multi-connection system, while the analytical results can be computed in less than one hour. The analysis and simulation programs are both written in C language.

6.6. QoS-Based Call Admission Control and Admission Regions

Based on the results given in sections 6.3 and 6.4, it is demonstrated that our analytical results in Chapter 3 to Chapter 5 are able to approximate the predictions of the QoS performances in the WCDMA system under low and medium load conditions. Here, we propose an analytical platform for a QoS-based call admission control (CAC) scheme, which is extended from [67]. The admission region obtained using this CAC scheme can satisfy the QoS requirements of all the four classes of admitted mobile users. The procedure to obtain the admission region is shown in Figure 6.30. Let $P_{loss,k}{}^{*}$ and $D_{loss,k}{}^{*}$ denote the packet loss rate and delay requirements of voice, video, web-browsing and data services, respectively when $k \in \{1,2,3,4\}$. The conditions of the CAC are $P_{loss,k} \le P_{loss,k}{}^{*}$ and $D_{loss,k} \le D_{loss,k}{}^{*}$, $k \in \{1,2,3,4\}$ for the single-connection model and $P_{loss,k,i} \le P_{loss,k}{}^{*}$ and $D_{loss,k,i} \le D_{loss,k}{}^{*}$, $i \in \{1,2,...,N\}$, $k \in \{1,2,3,4\}$ for the multi-connection model.

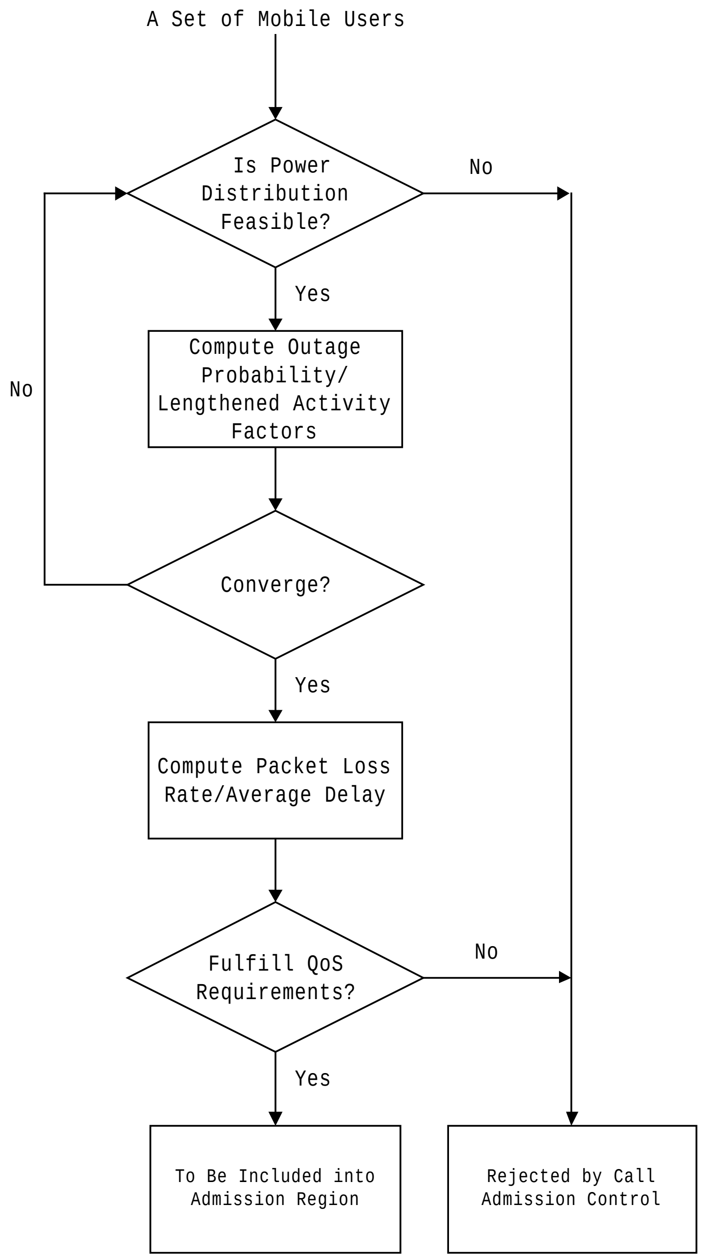

Figure 6.30 Call Admission Control Procedures

The power distribution scheme described in Chapter 4 is used in this CAC process. As shown in the Figure 6.30, the power distribution in Chapter 4 is used with the following steps in the CAC scheme.

[1]. The activity factors of web-browsing and data services in their sources are first used to calculate the received power levels of all classes in the system. This step can be done with equations (4.15)-(4.19) and (4.29).

[2]. With the obtained power levels, the outage probabilities of all classes are computed with equations (4.31) and (4.38).

[3]. With the obtained outage probabilities, the lengthened activity factors of web-browsing and data services are calculated with (4.42-4.45). Based on the lengthened activity factors, the power distribution scheme is performed again to calculate the received power levels, as stated in step [1]. From equations (4.15-19), (4.29), (4.31), (4.38) and (4.42)-(4.45), the power distribution scheme, outage probabilities and lengthened activity factors are intertwined. Therefore, step [1]-[3] are iterated until the derived values converge.

[4]. When the converged power levels, outage probabilities and activity factors are obtained, the delays and packet loss rates of all classes can be calculated, based on equations (5.1-5.20).

[5]. According to the derived delays and packet loss rates, CAC is able to determine whether these mobile users should be admitted or rejected. Admission regions can be achieved by computing all possible combination of mobile users with the satisfied QoS requirements and be presented in tables.

In the following, we give examples of admission regions for the single-connection and multi-connection system models, respectively.

6.6.1 Admission Region for the Single-Connection System Model

The single-connection system model only enables one connection within each mobile user. Using the parameters in Table 6.5, the AR is given by Figure 6.31. Note that the number of video services is set to be zero so that a 3-dimensinal AR can be presented. The admission region is the space on or under the shown surface.

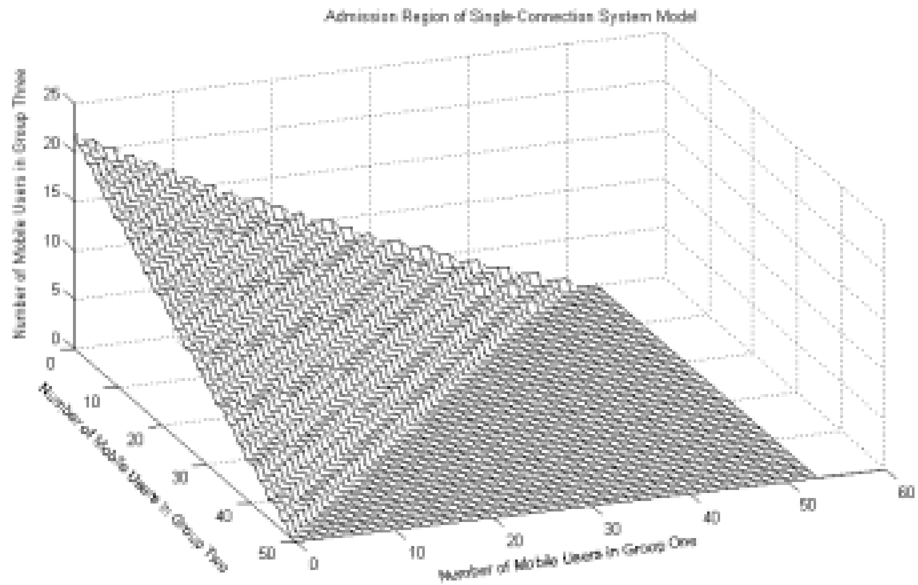

Figure 6.31 Admission Region of Single-Connection System Model (Number of Video Services = 0)

6.6.2 Admission Region for the Multi-Connection System Model

The multi-connection system model only enables multi-connection multiclass services within each mobile user. Using the parameters in Table 6.6, the AR is given by Figure 6.32. The number of users in Group two is fixed to be zero so as to present a 3-dimensional admission region. The space on or under the surface refers to the admission region. Any set of mobile users in the admission region can be guaranteed the satisfactory QoS performances.

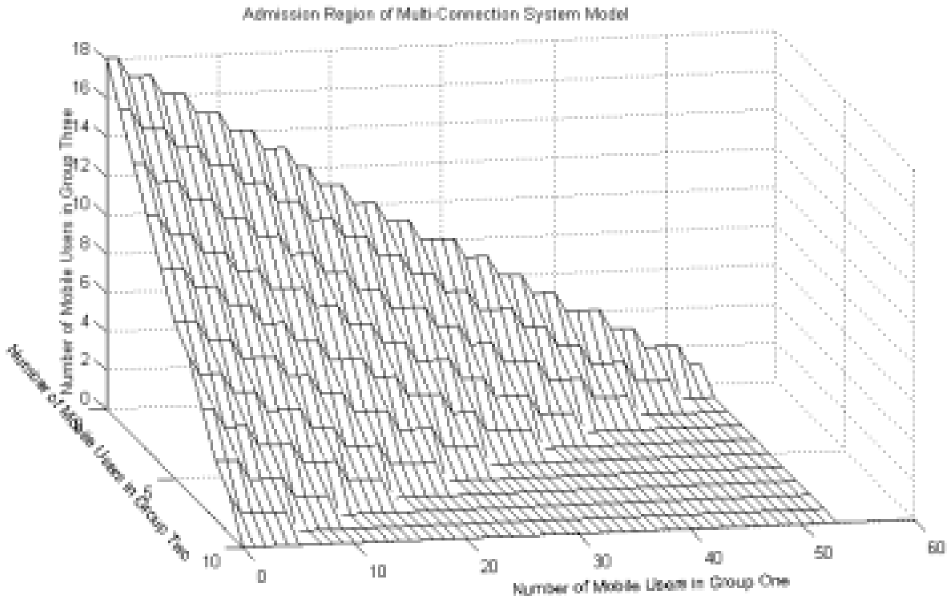

Figure 6.32 Admission Region for Multi-Connection System Model (Number of Mobile Users in Group 2 = 0)

From the above Figure 6.31 and Figure 6.32, we can see clearly that the system capacity in the single-connection system model is larger than that in the multi-connection system model in terms of the number of admitted mobile users. This is because each mobile user in the multi-connection system model supports more connections of services than each mobile user in the single-connection system model. However, the multi-connection system model is more realistic, since the WCDMA system enables a mobile user to provide multiservice capability. The system capacity of a multi-connection system model should be larger in terms of the total data rate, since the services within the same mobile user do not interfere with each other. Please note that we suppose the number of mobile users in group two is equal to zero, by which three-dimensional figures can be visually displayed. The call admission region for all the four groups of mobile users can also be shown in a table.

6.7 Conclusions

In this chapter, computer simulations are performed to verify the mathematical formulations of the QoS attributes in the WCDMA system, including outage probabilities, packet loss rates and delay, etc. The simulation and analytical results prove to be very close. Thus, the mathematical methods can be considered as reliable predictions of the QoS performances in the WCDMA system for the parameters chosen. In addition, we give a QoS-based call admission control scheme in this chapter. From this CAC scheme, 3-dimensional admission regions are derived in the two different system models.

Chapter 7

Conclusion and Future Works

7.1 Conclusion

In this thesis, we present an analytical framework to investigate the Quality of Service issue of the uplink of wideband CDMA cellular mobile networks. Our analysis is based on four QoS traffic classes, including voice, video, web-browsing and data services. These four classes differ in their traffic characteristics and their QoS requirements, such as packet loss rate and delay. Previous literatures have not provided a complete analytical solution to the QoS performances of the four classes in a WCDMA system. Our work focuses on providing an approximate method to calculate the QoS attributes of all these classes.

In order to proceed with the analytical work, we first define appropriate traffic models for these classes. Compared to existing works in the literature, our traffic models are more realistic. For example, we adopt a two-dimensional, continuous-time, discrete-state Sen's Markov model to approximate the VBR video sources. At the same time, we use heavy-tailed Pareto on/Pareto off process for both the web-browsing and data services. These models are definitely more appropriate than other models such as Poisson process used in some existing literatures.

Based on the Pareto on/Pareto off model, we investigate a Go-Back-N ARQ scheme with limited number of retransmissions and with a finite buffer size. These assumptions are more realistic as compared to some existing works which commonly assume unlimited number of retransmissions and with an infinite buffer size. This Go-Back-N scheme is analyzed in terms of the packet loss rates, delays and lengthened activity factors. The results obtained are new and have not been addressed before in the literature. We obtain an expression to estimate these attributes for the Pareto on/Pareto off process in the Go-Back-N channel. The shortcoming of our analytical model is that light or medium load is assumed.

We also present two different types of system models, including the single-connection and multi-connection models, of WCDMA cellular mobile networks. These two system models can serve a single service and multiclass services, respectively, to each user. A power distribution scheme is developed to allocate the required received power levels to all classes if the perfect power control is assumed, while satisfying the required SINR levels at the data link layer. This power distribution scheme is useful in evaluating the system capacity and calculating QoS performances.

Furthermore, we generalize an analytical expression of the outage probabilities for all traffic classes. At the same time, we study the lengthening of the activity factors of web-browsing and data services in the WCDMA channel. The lengthened activity factors, received power levels and outage probabilities are intertwined to each other. A simple iteration method is suggested to compute the convergent outage probabilities, lengthened activity factors and received power levels. Based on the outage probabilities and Go-Back-N ARQ analysis, we analyze the packet loss rate and delay for each class. From the numerical results obtained, our analysis can predict the QoS performances under light or medium traffic load rather well. Hence, our analytical approach can be used to determine the admission region.

7.2 Future Work

In this thesis, we only investigate the QoS performances in the uplink of the WCDMA system. Actually, the QoS issues in the downlink are also very critical. In the WCDMA system, since the uplink communication is asynchronous and experiences more interference, the QoS analysis of the downlink differs much from the work in this thesis. Additionally, the assignment of spreading codes in the downlink is another problem. That is because the code resources in the downlink are relatively scarce and thus an efficient allocation scheme is required for the spreading codes. These issues leave much room for further analyses.

Besides, the CAC at the base station of the WCDMA system is supposed to be performed based on satisfying the QoS requirements at different layers of the system. Therefore, the admission regions should be provided by the physical layer, the data link layer and network layer jointly. Therefore, our analytical work in this thesis has to be extended to the QoS provisioning/optimizing with the other layers. Thus, the intersection of the admission regions at all these layers is the final desired feasible admission region in the WCDMA system.

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Appendix

Intercell Interference Analysis

In a multi-cell UMTS cellular mobile network, a type of service in the WCDMA channel experiences interference both from intracell services within the same cell and from the intercell services within the neighbouring cells. In this thesis, we assume that our cellular system is made up of a number of square cells. Mobile users are assumed to be uniformly distributed in each cell. In the following, we will analyze the statistical characteristics of the intercell interference in both the single-connection system model and the multi-connection system model.

1. Intercell Interference for the Single-Connection System Model

Based on the single-connection system model given in section 4.1.1 and the analytical work given in [21, 22, 32], the intercell interference-to-signal ratio of a service can be formulated by equation (A.1), taking the path loss and lognormal shadowing into consideration.

$$\frac{I_i}{S_i} = \left(\frac{r_m}{r_d}\right)^4 10^{(\varepsilon_d - \varepsilon_m)/10}, \; i = 1, 2l, 2h, 3, 4, \tag{A.1}$$

where I_i and S_i are the intercell interference and the received power of a voice service, a video service using a low-bit-rate spreading code, a video service using a high-bit-rate spreading code, a web-browsing service and a data service, when $i \in \{1, 2l, 2h, 3, 4\}$, respectively. In equation (A.1), ε_m and ε_d are two independent Guassian random variables with zero mean and σ^2 variance. Let us suppose that r_m denotes the distance between an intercell service and the intercell base station suppose r_d denotes the distance between an intercell service and the intracell base station. The mean and variance of the intercell interference-to-signal ratio are given by

$$E[\frac{I_i}{S_i}] \le \begin{cases} N_1 p_{on1} \dfrac{\iint f(\frac{r_m}{r_d})dA}{A}, i=1 \\ N_2 p_{on2h} \dfrac{\iint f(\frac{r_m}{r_d})dA}{A}, i=2h \\ MN_2 p_{on2l} \dfrac{\iint f(\frac{r_m}{r_d})dA}{A}, i=2l, \\ N_3 p_{on3,c} \dfrac{\iint f(\frac{r_m}{r_d})dA}{A}, i=3, \\ N_4 p_{on4,c} \dfrac{\iint f(\frac{r_m}{r_d})dA}{A}, i=4, \end{cases} \tag{A.2}$$

and

$$Var[\frac{I_i}{S_i}] \le \begin{cases} N_1 \dfrac{\iint [p_{on1} g(\frac{r_m}{r_d}) - p_{on1}{}^2 f^2(\frac{r_m}{r_d})]dA}{A}, i=1 \\ N_2 \dfrac{\iint [p_{on2h} g(\frac{r_m}{r_d}) - p_{on2h}{}^2 f^2(\frac{r_m}{r_d})]dA}{A}, i=2h \\ N_2 \dfrac{\iint \{Mp_{on2l}[1+(M-1)p_{on2l}]g(\frac{r_m}{r_d}) - (Mp_{on2l})^2 f^2(\frac{r_m}{r_d})]\}dA}{A}, i=2l, \\ N_i \dfrac{\iint [p_{oni,c} g(\frac{r_m}{r_d}) - p_{oni,c}{}^2 f^2(\frac{r_m}{r_d})]dA}{A}, i=3,4, \end{cases} \tag{A.3}$$

where $f(\frac{r_m}{r_d})$ and $g(\frac{r_m}{r_d})$ are given by [21-22], [33] and [54].

$$f(\frac{r_m}{r_d}) = (\frac{r_m}{r_d})^4 e^{(\frac{\sigma \ln 10}{10})^2} [1 - Q(\frac{40 \log(r_m / r_d)}{\sqrt{2\sigma^2}} - \frac{\sqrt{2\sigma^2} \ln 10}{10})], \tag{A.4}$$

and

$$g(\frac{r_m}{r_d}) = (\frac{r_m}{r_d})^8 e^{(\frac{\sigma \ln 10}{5})^2} [1 - Q(\frac{40 \log(r_m / r_d)}{\sqrt{2\sigma^2}} - \frac{\sqrt{2\sigma^2} \ln 10}{5})]. \tag{A.5}$$

Then, the mean and variance of the total intercell interference can be expressed as

$$\begin{aligned} E[I_{intercell}] &= E[I_1 + I_{2l} + I_{2h} + I_3 + I_4] \\ &\le (S_1 N_1 p_{on1} + S_{2l} N_2 p_{on2l} M + S_{2h} N_2 p_{on2h} + S_3 N_3 p_{on3,c} + S_4 N_4 p_{on4,c}) \frac{\iint f(\frac{r_m}{r_d}) dA}{A}, \end{aligned} \tag{A.6}$$

and

$$\begin{aligned} &Var[I_{intercell}] = Var[I_1 + I_{2l} + I_{2h} + I_3 + I_4] \\ &\le S_1{}^2 N_1 \frac{\iint [p_{on1} g(\frac{r_m}{r_d}) - p_{on1}{}^2 f^2(\frac{r_m}{r_d})] dA}{A} \\ &+ S_{2l}{}^2 N_2 \frac{\iint [M p_{on2l} [1 + (M-1) p_{on2l}] g(\frac{r_m}{r_d}) - (M p_{on2l})^2 f^2(\frac{r_m}{r_d})] dA}{A} \\ &+ S_{2h}{}^2 N_2 \frac{\iint [p_{on2h} g(\frac{r_m}{r_d}) - (p_{on2h})^2 f^2(\frac{r_m}{r_d})] dA}{A} + S_3{}^2 N_3 \frac{\iint [p_{on3,c} g(\frac{r_m}{r_d}) - p_{on3,c}{}^2 f^2(\frac{r_m}{r_d})] dA}{A} \\ &+ S_4{}^2 N_4 \frac{\iint [p_{on4,c} g(\frac{r_m}{r_d}) - p_{on4,c}{}^2 f^2(\frac{r_m}{r_d})] dA}{A}. \end{aligned} \tag{A.7}$$

2. Intercell Interference for the Multi-Connection System Model

Based on the multi-connection system model given in section 4.1.2 and the analytical work given in [21, 22, 32], the intercell interference-to-signal ratio of a service within the *i*th mobile user can be formulated by equation (A.8), taking the path loss and lognormal shadowing into consideration.

$$\frac{I_{i,j}}{S_{i,j}} = \left(\frac{r_m}{r_d}\right)^4 10^{(\varepsilon_d - \varepsilon_m)/10}, \ \ j = 1, 2l, 2h, 3, 4, \tag{A.8}$$

where $I_{i,j}$ and $S_{i,j}$ are the intercell interference and the received power of a voice, a video service using a low-bit-rate spreading code, a video service using a high-bit-rate spreading code, a web-browsing service and a data service within the *i*th mobile user when $j \in \{1, 2l, 2h, 3, 4\}$, respectively.

In equation (A.8), ε_m and ε_d are two independent Guassian random variables with zero mean and σ^2 variance. Let us suppose that r_m denotes the distance between an intercell service and the intercell base station. Furthermore, let us suppose that r_d denotes the distance between an intercell service and the intracell base station. With $f(\frac{r_m}{r_d})$ and $g(\frac{r_m}{r_d})$ defined by (A.4) and (A.5). The mean and variance of the intercell interference-to-signal of the *i*th mobile user are given by

$$E[\frac{I_{i,j}}{S_{i,j}}] \le \begin{cases} n_{i,1} p_{on1} \dfrac{\iint f(\frac{r_m}{r_d}) dA}{A}, \ j = 1 \\ n_{i,2} p_{on2h} \dfrac{\iint f(\frac{r_m}{r_d}) dA}{A}, \ j = 2h \\ Mn_{i,2} p_{on2l} \dfrac{\iint f(\frac{r_m}{r_d}) dA}{A}, \ j = 2l, \\ n_{i,j} p_{onj,i,c} \dfrac{\iint f(\frac{r_m}{r_d}) dA}{A}, \ j = 3, 4, \end{cases} \tag{A.9}$$

and

$$Var[\frac{I_{i,j}}{S_{i,j}}] \le \begin{cases} n_{i,1}\dfrac{\iint [p_{on1}g(\frac{r_m}{r_d}) - p_{on1}{}^2 f^2(\frac{r_m}{r_d})]dA}{A}, & j=1 \\ n_{i,2}\dfrac{\iint [p_{2h}g(\frac{r_m}{r_d}) - p_{2h}{}^2 f^2(\frac{r_m}{r_d})]dA}{A}, & j=2h \\ n_{i,2}\dfrac{\iint \{Mp_{2l}[1+(M-1)p_{2l}]g(\frac{r_m}{r_d}) - (Mp_{2l})^2 f^2(\frac{r_m}{r_d})]\}dA}{A}, & j=2l, \\ n_{i,j}\dfrac{\iint [p_{onj,i,c}g(\frac{r_m}{r_d}) - p_{onj,i,c}{}^2 f^2(\frac{r_m}{r_d})]dA}{A}, & j=3,4. \end{cases} \tag{A.10}$$

Then, the mean and variance of the total intercell interference can be expressed as

$$\begin{aligned} E[I_{intercell}] &= \sum_{i=1}^{N} E[I_{i,1} + I_{i,2l} + I_{i,2h} + I_{i,3} + I_{i,4}] \le \\ &\sum_{i=1}^{N} (S_{i,1}n_{i,1}p_{on1} + S_{i,2l}n_{i,2}p_{on2l}M + S_{i,2h}n_{i,2}p_{on2h} + S_{i,3}n_{i,3}p_{on3,i,c} + S_{i,4}n_{i,4}p_{on4,i,c})\frac{\iint f(\frac{r_m}{r_d})dA}{A}, \end{aligned} \tag{A.11}$$

and

$$\begin{aligned} Var[I_{intercell}] &= \sum_{i=1}^{N} Var[I_{i,1} + I_{i,2l} + I_{i,2h} + I_{i,3} + I_{i,4}] \\ &\le \sum_{i=1}^{N} \{S_{i,1}{}^2 n_{i,1}\frac{\iint [p_{on1}g(\frac{r_m}{r_d}) - p_{on1}{}^2 f^2(\frac{r_m}{r_d})]dA}{A} \\ &+ S_{i,2l}{}^2 n_{i,2}\frac{\iint [Mp_{on2l}[1+(M-1)p_{on2l}]g(\frac{r_m}{r_d}) - (Mp_{on2l})^2 f^2(\frac{r_m}{r_d})]dA}{A} \\ &+ S_{i,2h}{}^2 n_{i,2}\frac{\iint [p_{on2h}g(\frac{r_m}{r_d}) - (p_{on2h})^2 f^2(\frac{r_m}{r_d})]dA}{A} \\ &+ S_{j,3}{}^2 n_{i,3}\frac{\iint [p_{on3,i,c}g(\frac{r_m}{r_d}) - p_{on3,i,c}{}^2 f^2(\frac{r_m}{r_d})]dA}{A} \\ &+ S_{i,4}{}^2 n_{i,4}\frac{\iint [p_{on4,i,c}g(\frac{r_m}{r_d}) - p_{on4,i,c}{}^2 f^2(\frac{r_m}{r_d})]dA}{A}\}. \end{aligned} \tag{A.12}$$