Dynamic Resource Allocation

Under Different Traffic Models in GPRS/EGPRS

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ABSTRACT

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With the increasing demand for information anywhere, any time, wireless data service is becoming more important than before. In 2G mobile cellular systems, emphasis is put on circuit-switched voice services that can only supply limited data services, which cannot meet nowadays data requirement. To meet the new challenge, 2G mobile cellular systems have taken an evolutionary approach in developing data based services, which is referred to 2.5G or 2.75G. GPRS, EGPRS and EDGE are standardized by European Telecommunications Standards Institute (ETSI) and belong to 2.5G-2.75G standard. GPRS is only used for packet data service instead of voice traffic. In fact in a cell that supports GPRS, both GSM and GPRS systems operate in parallel to provide service for packet data and voice traffic. EDGE Phase one focuses on the non-real-time packet data services. This is also called EGPRS. The EDGE Phase Two aims at providing both real-time services such as voice and video delivering and non-real-time services in an end-to-end packet mode.

To provide various services and to achieve more efficient utilization of the scarce frequency spectrum, research on radio resource allocation is becoming a hot topic. To evaluate the advantage of dynamic resource allocation, the performances of fixed and dynamic resource allocation are compared in this thesis. To simulate various services, four different data rate classes, and four different traffic models are assumed. The result

of the simulations show that: the Dynamic Resource Allocation Scheme can bring much better performance improvement than the Fixed Resource Allocation Scheme. At the same average data rate and the same variance, the performance is the same and does not change with the traffic model. The resource allocation schemes show robustness to different traffic models.

ACKNOWLEDGEMENTS

It is with the utmost sincerity that I express my thanks to my thesis supervisor Dr. A. K. Elhakeem, for originally proposing the research area and subsequently guiding me through the various stages of the thesis development.

I am also grateful for the constant support provided by my husband, daughter and friends.

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List of Acronyms and Abbreviations

ACK	Acknowledgment
AGCH	Access Grant Channel
ATM	Asynchronous Transfer Mode
AuC	Authentication Center
BCH	Broadcast Channel
BCCH	Broadcast Control Channel
BEC	Backward Error Correction
BSC	Base Station Controller
BSS	Base Station Subsystem
BSSGP	Base Station System GPRS Protocol
BTS	Base Transceiver Station
CCCH	Common Control Channel
DCCH	Dedicated Control Channel
EDGE	Enhanced Data rates for GSM Evolution
EGPRS	Enhanced General packet Radio Services
EIR	Equipment Identity Register
ETSI	European Telecommunications Standards Institute
FCCH	Frequency Correction Channel
FDMA	Frequency- Division Multiple Access
F-PAGCH	Fast Packet Access Grant Channel
F-PPCH	Fast Packet Polling Channel
GGSN	Gateway GPRS Support Node
GMSK	Gaussian Minimum Shift Keying
GPRS	General Packet Radio Services

GSM	Global System Mobile
GSN	GPRS Support Node
GTP	GPRS Tunnel Protocol
HLR	Home Location Register
HSCSD	High Speed Circuit Switched Data
IP	Internet Protocol
ITU	International Telecommunication Union
ISDN	Integrated Service Digital Network
ISO	International Organization for Standardization
IWF	Inter-Working Function
LLC	Logical Link Control
MAC	Medium Access Control
MS	Mobile Station
MSC	Mobile service Switching Center
NACK	Negative Acknowledgment
NSS	Network and Switching Subsystem
OMC	Operation and Maintenance Center
OSI	Open System Interconnections
PAGCH	Packet Access Grant Channel
PBCCH	Packet Broadcast Control Channel
PCCCH	Packet Common Control Channel
PCH	Paging Channel
PDCH	Packet Data Channel
PDTCH	Packet Data Transfer Channel
PDU	Packet Data Unit
PLL	Physical Link Sublayer
PLMN	Public Land Mobile Network

PNCH	Packet Notification Channel
PPCH	Packet Paging Channel
PRACH	Packet Random Access Channel
PSK	Phase Shift Keying
PSTN	Public Switched Telephone Network
PTCH	Packet Traffic Channels
QoS	Quality of Service
RACH	Random Access Channel
RLC	Radio Link Control
RFL	Physical RF Sublayer
SACCH	Slow Associated Control Channel
SCH	Synchronization Channel
SDCCH	Stand-alone Dedicated Control Channels
SGSN	Serving GPRS Support Node
SIM	Subscriber Identity Module
SNDCP	Subnetwork Dependent Convergence Protocol
SCCP	Signaling Correction Control Part
TBF	Temporary Block Flow
TCH	Traffic Channel
TCP	Transmission Control Protocol
TDMA	Time- Division Multiple Access
TFI	Temporary Flow Identifier
UDP	User Datagram Protocol
USF	Uplink State Flag
UMTS	Universal Mobile Telecommunications System
VLR	Visitor Location Register

Chapter 1

Introduction

1.1 Issues in Resource Allocation Schemes in GPRS/EGPRS

In the past few years, a tremendous growth of cellular voice services and an explosive subscription rate of Internet services were witnessed. These new changes made the operator consider offering diverse applications instead of voice service only through the systems. The diverse applications involve voice, web browsing, videoconference, e-mail etc. So, the wireless network has evolved from circuit switched networks to packet switched ones.

Users of packet switched networks can get benefit from shorter access times and higher data rates. In addition, a packet switched network offers a more user-friendly billing system than that offered by circuit switched services. In circuit switched mobile cellular systems, the user must pay for the duration of the call even for idle period when no messages are sent. Whereas, in packet switched mobile cellular system, billing can be based on the amount of transmitted data. So, packet switched mobile cellular networks are more suitable for data transmission.

GPRS, EGPRS, EDGE are evolution of GSM. They use a TDMA based radio technology with 200 kHz channels, packet switched core network. In GPRS or EDGE phase one, high data rates are provided for non-real-time services using high-level modulation. In EDGE phase two, both real-time (e.g., voice, video conferencing, real-time image transfer, etc.) and non-real-time (e.g., database applications, web browsing, email, streaming video and sound, etc.) services are provided over an all IP core network. The services are categorized to four classes: Background, Conversational, Interactive and Streaming.

The air interface of GPRS is based on Gaussian Minimum Shift Keying (GMSK) modulation that is applied to GSM. Four different coding schemes in GPRS can provide data rates from 9.05-171.2 kbps (theoretical). In contrast to the GSM system, GPRS allows multiple users to occupy one single traffic channel and also allows one single user to occupy several traffic channels. So, GPRS is much more flexible and efficient than the GSM system.

The air interface of EGPRS/EDGE is based on Gaussian Minimum Shift Keying (GMSK) modulation and 8-Phase Shift Keying (8-PSK) modulation. Nine different coding schemes in EGPRS/EDGE can provide data rates ranging from 8.8kbps to 59.2 kbps per channel. The same as GPRS, EGPRS/EDGE can provide flexible and efficient resource allocation.

In recent years, there has been a considerable amount of work done in the area of resource allocation in GPRS/EGPRS. There are three channel allocation schemes in GPRS. These are:

1. Fixed Channel Allocation
2. Dynamic Channel Allocation
3. Extended Dynamic Channel Allocation

In Fixed Allocation Scheme, the resource allocated to a particular MS is fixed for the duration of the call, whereas in Dynamic Allocation Scheme, the resource allocated to a particular MS is dynamic for the duration of the call. The efficiency of radio resource allocation in the dynamic scheme is much better than in the fixed resource allocation scheme.

1.2 Objectives of the thesis

The effect of different traffic models and different resource allocation schemes has not been analyzed thoroughly. In this thesis, a detailed simulation model is established and performance criterion such as Packet Efficiency, Average packet loss rate, Average buffer overflow and Average queuing delay, Call Block Probability, etc is analyzed.

The objects of this thesis are as follow:

1. Build a simulation model of GPRS/EGPRS resource allocation procedure.
2. To investigate the influence of various parameters, such as buffer size, Assignment Parameter and traffic load.
3. To investigate the effect of different combination of users' data rate (most of the users are high data rate users or low data rate users)
4. To investigate the effect of different traffic models.
5. To investigate the advantages of dynamic resource allocation scheme.

1.3 Thesis Organizations

This Thesis is organized as follows. In Chapter 2, GSM, GPRS and EGPRS/EDGE systems are introduced. In Chapter 3, emphasis is put on the resource allocation schemes of GPRS and EGPRS. In Chapter 4, the simulation model of GPRS/EGPRS resource allocation is developed in C++. The performance results are shown in this chapter. In Chapter 5, conclusions, thesis contributions and suggestions for future work are presented.

Chapter 2

Review of GSM, GPRS, and EGPRS

Most of today's ubiquitous cellular networks follow the second -generation or called 2G cellular standards. In general, the 2G mobile cellular systems have been primarily designed to provide circuit-switched voice services with limited data capability. But, with the explosive growth of the Internet and the subsequent demand for wireless data, data service became a major component of many mobile cellular systems around the world.

To meet the data service requirement, the 3G mobile cellular systems had been designed for packet based wireless service. However, 3G systems will require expensive new equipment, the installation of 3G systems will be slow and gradual throughout the world.

Compared with the 3G mobile cellular systems, the 2G mobile cellular systems have taken an evolutionary approach in developing data based services. The evolved systems are usually referred to as 2.5G-2.75G, means such systems bridge the gap between 2G and 3G systems.

GSM (Global System Mobile), standardized by European Telecommunications Standards Institute (ETSI) is undoubtedly the most successful 2G mobile cellular systems in the world. The main service design focus of GSM was voice, so, the circuit-switched connections are not ideally suited to packet data based applications and the data speeds supported are very slow. The GSM community reacted quickly and took the necessary steps to improve GSM's capabilities, while preserving it's fundamental design principles. Two major standards have emerged: GPRS (General Packet Radio Service) and EDGE (Enhanced Data rates for GSM Evolution).

GPRS is standardized by European Telecommunications Standards Institute (ETSI), is "designed for best-effort packet data services" [1], which specifies a packet switched-core network and an air interface based on Gaussian Minimum Shift Keying (GMSK) modulation. In contrast to time-oriented charging applied for circuit-switched connections, packet-switched data services will allow charging depending on the amount of data transmitted and the quality of service (QoS) negotiated. GPRS is an on-demand based bearer data service, sharing the same GSM air interface, that offers QoS to the users by dynamic allocation of different amounts of resources depending on the user requirements (throughput, delay, reliability, priority, etc.).

EGPRS (Enhanced General packet Radio Services) is a subset of the EDGE (Enhanced Data Rates for GSM Evolution). This is the standard for the European Telecommunication Standard Institute (ETSI) and Third Generation Partnership Project (3GPP) compliant standard for 2.5G protocols and 3G-evolution path. The main benefit of EGPRS is its provision of a cost-effective way to ease into next-generation services and applications. EGPRS uses a TDMA-based packet-switched radio technology with 200kHz channels, a time frame structure similar to GSM, and an evolved, packet-switched GPRS core network and introduces a new air interface-EDGE, to support higher data rates. This is accomplished mainly by using 8-phase shift keying (8-PSK), in addition to the traditional GMSK.

This chapter presents the basic system introduction of GSM, GPRS and EGPRS.

2.1 Overview of GSM system

Global System for Mobile (GSM) is a second-generation cellular system standard which was initially developed to replace the non-compatibility analog cellular systems used in all European countries with a unique digital cellular standard. It has rapidly gained acceptance and market share worldwide, and became the world's most popular 2G technology. In addition to digital transmission, GSM incorporates many advanced services and features, including ISDN compatibility and worldwide roaming in other GSM networks.

2.1.1 GSM System Architecture

The GSM system architecture consists of three major interconnected subsystems that interact between themselves and with the users through certain network interfaces. The subsystems are the Base Station Subsystem (BSS), Network and Switching Subsystem (NSS), and the Operation and Maintenance Centre (OMC). The Mobile Station (MS) is also a subsystem, but is usually considered as part of the BSS for architecture purposes. Figure 2.1 shows the GSM system architecture and interfaces.

Mobile Station (MS)

The MS consists of the mobile equipment (the terminal) and a smart card called the Subscriber Identity Module (SIM). The SIM card provides personal mobility, so that the user can access to subscribed services. By inserting the SIM card into another GSM terminal, the user is able to receive calls, make calls and receive other subscribed services with that terminal.

Base Station Subsystem (BSS)

The BSS, also known as the radio subsystem, provides and manages radio transmission paths between the mobile stations and all other subsystems of GSM. The BSS is composed of two parts, the Base Transceiver Station (BTS) and the Base Station Controller (BSC). BSC and BTS communicate across the standardized Abis interface.

The BSC manages the radio resources for many BTSs. It handles radio-channel set-up, frequency hopping, and handovers. The BSC is the connection between the mobile station and the Mobile Service Switching Centre (MSC).

The BTS handles the radio-link protocols with the Mobile Station and manages the signal processing related to the air interface. In a large urban area, there will be a large number of BTSs with small power deployed. Whereas in rural area, there will be small number of BTSs with large power deployed.

Network and Switching Subsystem (NSS)

Several functional entities work together to form the NSS and to provide services like registration, authentication, location updating, handovers, and call routing to a roaming subscriber.

The central component of the Network and Switching Subsystem is the Mobile services Switching Centre (MSC). It acts like a normal switching node of the PSTN or ISDN, and additionally provides all the functionality needed to handle a mobile subscriber. The MSC provides the connection to the fixed networks (such as the PSTN or ISDN).

The Home Location Register (HLR) and Visitor Location Register (VLR), together with the MSC, provide the call-routing and roaming capabilities of GSM. The HLR contains all the administrative information of each subscriber registered in the corresponding GSM network, along with the current location of the mobile. The location of the mobile is typically in the form of the signalling address of the VLR associated with the mobile station. There is logically one HLR per GSM network, although it may be implemented as a distributed database.

The Visitor Location Register (VLR) contains selected administrative information from the HLR, necessary for call control and provision of the subscribed services, for each mobile currently located in the geographical area controlled by the VLR.

The Authentication Centre (AuC) is a protected database that holds a copy of the secret key stored in each subscriber's SIM card, which is used for authentication and encryption over the radio channel. The AuC provides additional security against fraud. It is normally located close to each HLR within a GSM network.

The Equipment Identity Register (EIR) is a database that contains a list of all valid mobile station equipment within the network.

Operation and Maintenance Center (OMC)

The OMC is a management system that oversees the GSM functional blocks. The OMC allows system engineers to monitor, diagnose, and troubleshoot all aspects of the GSM system. Hardware redundancy and intelligent error detection mechanisms help prevent network downtime. The OMC is responsible for controlling and maintaining the MSC, BSC, and BTS. The OMC has three main functions:

- Maintain all telecommunication hardware and network operations with a particular market.
- Manage all charging and billing procedures.
- Manage all mobile equipment in the system.

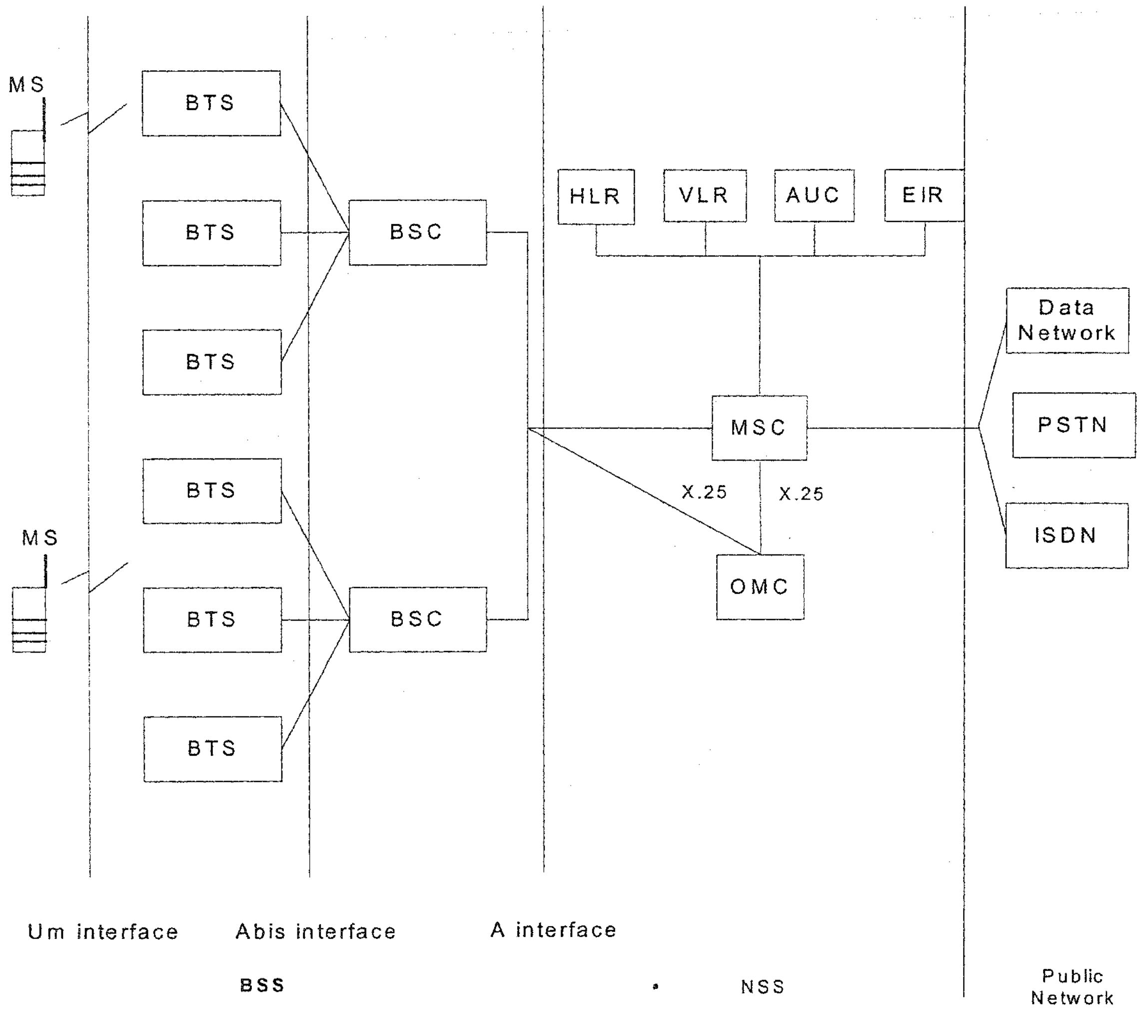


Figure 2.1 General architecture of a GSM network

Interfaces:

- **Um:** The MS and the BSS communicate across the Um interface, also known as the air interface or radio link. Um interface is the most important interface of the GSM system [2]
- **Abis:** The BTS and the BSC communicate across the Abis interface.
- **A:** The BSS communicates with the MSC across the A interface.
- **SS7:** SS7 protocol, called the Signalling Correction Control Part (SCCP), supports communication between the MSC and the other public network.

2.1.2 GSM Channel Coding

The speech is analog, so in order to be transmitted over digital communication systems it should be digitized and coded. Speech is divided into 20 millisecond samples (frame) and current sample is predicted from previous samples due to the slow change of voice patterns. The predicted and real information are compared and the difference is saved. Each 20-millisecond sample is encoded using 260 bits to form a 13 kbps data stream. (GSM Phase 2 introduced a half-rate speech code operating at around 7 kbps, effectively doubling the capacity of a network). The digital data stream is forward-error-correction coded by a convolution encoder. The gross bit rate after channel coding is 22.8 kbps (or 456 bits every 20 ms). These 456 bits are divided into eight 57-bit blocks, and the result is interleaved amongst eight successive time slot bursts for protection against burst transmission errors.

Each time slot burst is 156.25 bits and contains two 57-bit blocks, and a 26-bit training sequence used for equalization. A burst is transmitted in 0.577 ms for a total bit rate of 270.8 kbps, and is modulated using Gaussian Minimum Shift Keying (GMSK) onto the 200 kHz carrier frequency. Forward error control and equalization contribute to the robustness of GSM radio signals against interference and multi-path fading.

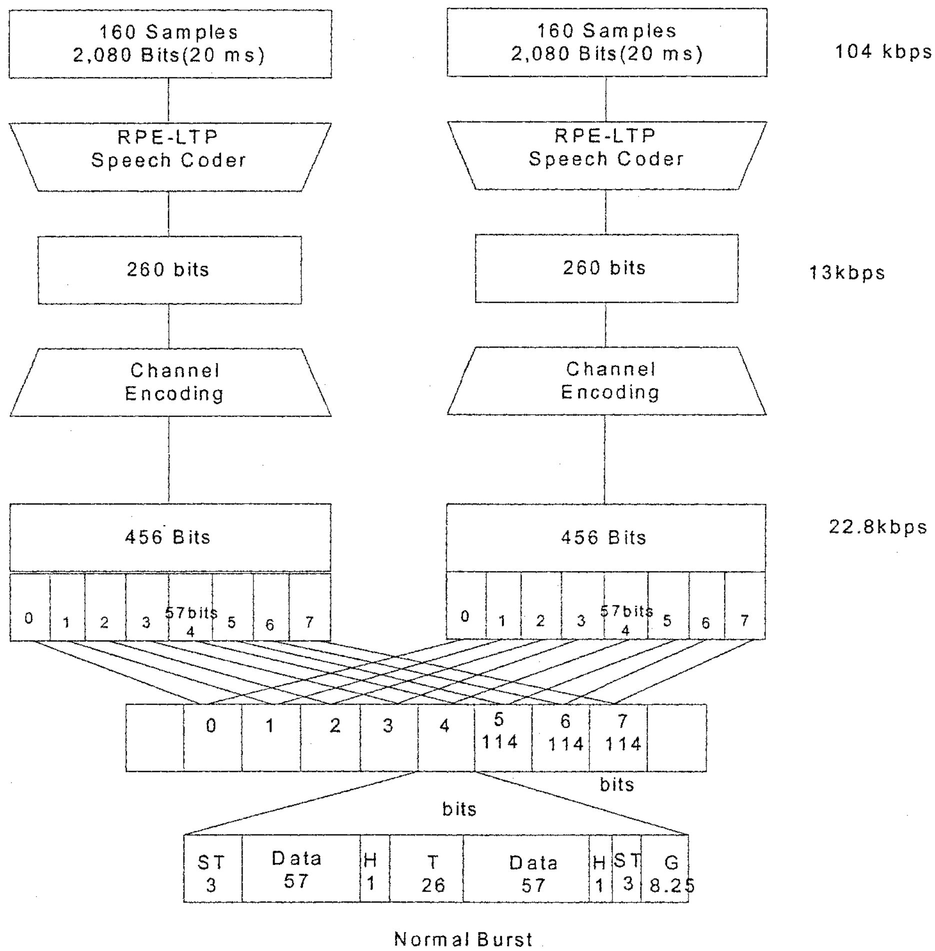


Figure 2.2 Interleaving speech frames to TDMA frames

2.1.3 GSM Radio Subsystem

The International Telecommunication Union (ITU), which manages the international allocation of radio spectrum, allocated the following bands:

- GSM900:

890-915 MHz for the uplink (mobile station to base station)

935-960 MHz for the downlink (base station to mobile station)

- GSM1800:

  1710-1785 MHz for the uplink (mobile station to base station)

  1805-1880 MHz for the downlink (base station to mobile station)

- GSM1900:

  1850-1910 MHz for the uplink (mobile station to base station)

  1930-1990 MHz for the downlink (base station to mobile station)

GSM uses a combination of Time- and Frequency- Division Multiple Access (TDMA/FDMA) to provide multiple accesses to mobile users. The FDMA part involves the division by frequency of the (maximum) 25 MHz bandwidth into 124 carrier frequencies spaced 200 kHz apart. One or more carrier frequencies are assigned to each base station. Each of these carrier frequencies is then divided in time, using a TDMA scheme. The fundamental unit of time in this TDMA scheme is called a *burst period* and it lasts 15/26 ms (or approx. 0.577 ms). Eight burst periods are grouped into a *TDMA frame* (120/26 ms, or approx. 4.615 ms), which forms the basic unit for the definition of logical channels. One physical channel is one burst period per TDMA frame. The number and position of their corresponding burst periods define channels. All these definitions are cyclic, and the entire pattern repeats approximately every 3 hours.

2.1.4 GSM Channels

There are two types of GSM logical channels, called *traffic channels* (TCH) and *control channels* (CCHs). Traffic channels carry digitally encoded user speech or user data and have identical functions and formats on both the forward and reverse link. Control channels carry signalling and synchronizing commands between the base station and the mobile station. Certain types of control channels are defined for just the forward or reverse link. The definition of GSM channels is in [2].

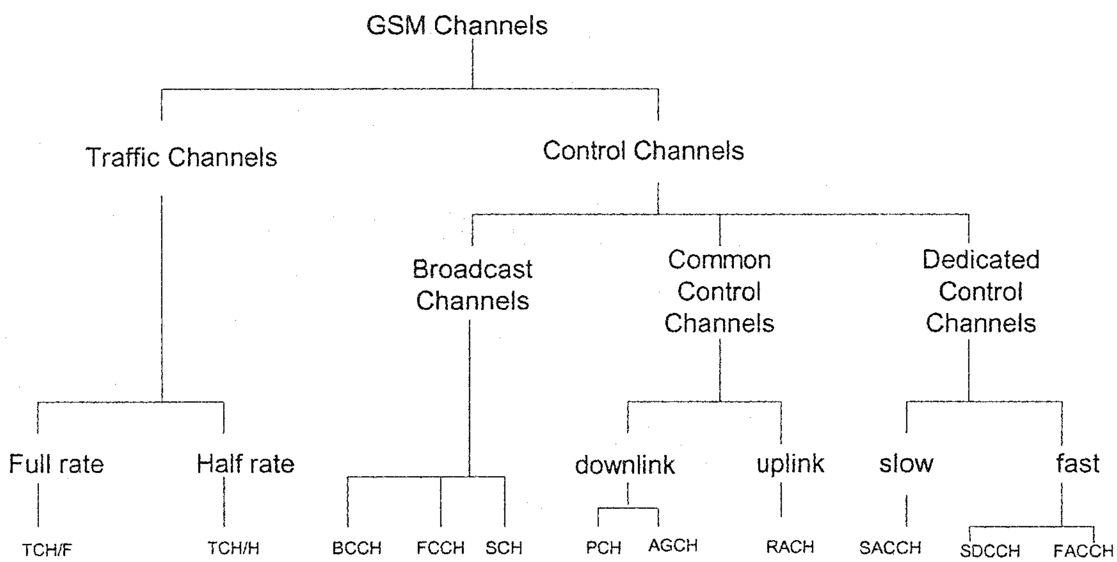


Figure 2.3 GSM Channels

Figure 2.4 is the GSM frame structure and time slot normal data bursts in GSM. There are 148 bits in one time slot, 114 bits are information-bearing bits that are transmitted as two 57 bits sequences close to the beginning and end of the burst. The 26 bit training sequence in the middle allows the adaptive equalizer in the mobile or base station receiver to analyze the radio channel characteristics before decoding the user data. The stealing bits are used to distinguish whether the TS contains voice or control data.

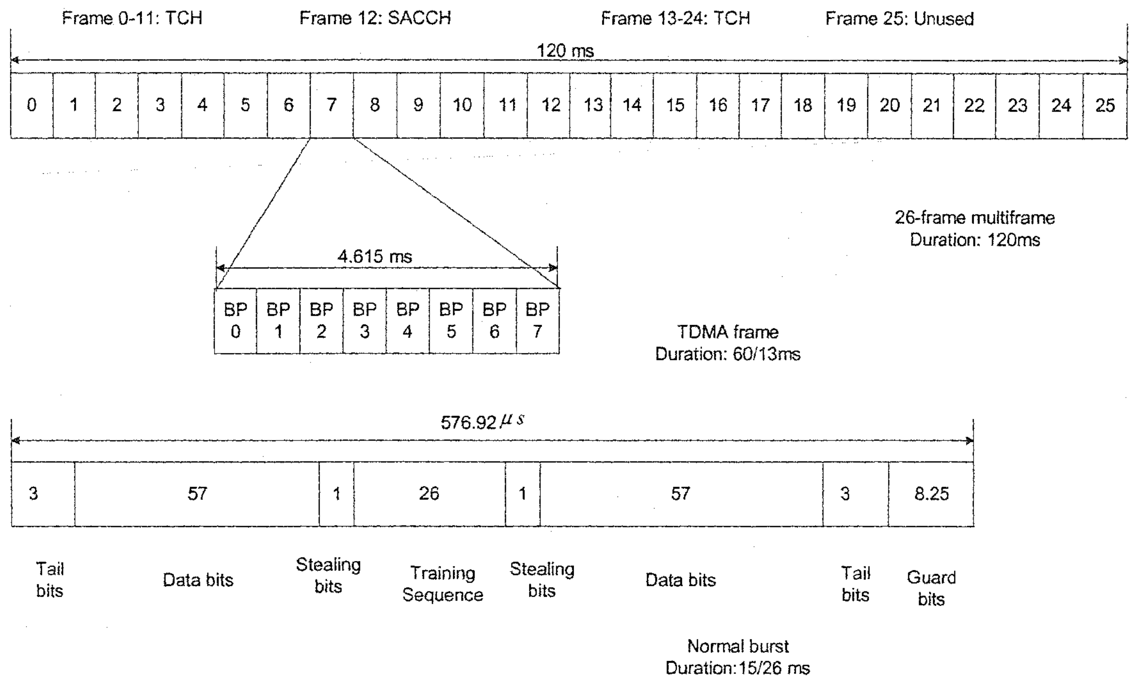


Figure 2.4 Organization of bursts, TDMA frames, and multi-frames for speech and data

FCCH burst

3 start bits	142 fixed bits of all zeroes	3 stop bits	8.25 bits guard period

SCH burst

3 start bits	39 bits of encrypted data	64 bits of training	39 bits of encrypted data	3 stop bits	8.25 bits guard period

RACH burst

8 start bits	41 bits of synchronization	36 bits of encrypted data	3 stop bits	68.25 bits extended guard period

Dummy burst

3 start bits	58 mixed bits	26 training bits	58 mixed bits	3 stop bits	8.25 bits guard period

Figure 2.5 Time slot data bursts in GSM

Normal bursts are used for TCH and DCCH transmissions on both the up and down link. Other types of data bursts used for various control and traffic bursts are shown in figure 2.5. FCCH and SCH bursts are used in TS0 of specific frames to broadcast the frequency and time synchronization control messages on the downlink. The RACH burst is used by all mobiles to access service from any base station, and the dummy burst is used as filler information for unused time slots on the downlink.[23]

2.1.4.1 GSM Traffic Channels (TCHs)

A traffic channel (TCH) is used to carry speech and data traffic. Traffic channels are defined using a 26-frame multiframe, or a group of 26 TDMA frames. The length of a 26-frame multiframe is 120 ms, which is how the length of a burst period is defined (120 ms divided by 26 frames divided by 8 burst periods per frame). Out of the 26 frames, 24 are used for traffic, 1 is used for the Slow Associated Control Channel (SACCH) and 1 is currently unused).

In full-rate TCH (TCH/F, 22.8kbps), data is mapped in one time slot per frame. In half-rate TCH (TCH/H, 11.4kbps), user data is mapped in the same time slot, but in alternate fames. That is, two half-rate channel users would share the same time slot, but would alternately transmit during every other frame. If a TCH/F is used for data communications, the usable data rate drops to 9.6kbps (in TCH/H: max.4.8kbps) due to the enhanced security algorithms. Half-rate TCH will effectively double the capacity of a system once half-rate speech coders are specified. Eighth-rate TCHs are also specified, and are used for signalling. In the recommendations, they are called Stand-alone Dedicated Control Channels (SDCCH).

2.1.4.2 GSM Control channels (CCHs)

There are three main control channels in the GSM system. These are the *broadcast channel* (BCH), the *common control channel* (CCCH), and the *dedicated control channel* (DCCH). Each control channel consists of several logical channels, which are distributed in time to provide the necessary GSM control functions. The control channels are defined within a 51-frame multi-frame, so that dedicated mobiles using the 26-frame multi-frame TCH structure can still monitor control channels. The control channels include:

BCH- Broadcast Channel

The broadcast channel operates on the forward link and transmits data only in the first time slot (TS0) of certain GSM frames. Three separate channels that are given access to TS0 during various frames of the 51-frame sequence. The three types of BCH are:

(a) BCCH- Broadcast Control Channel

The BCCH is a forward control channel that is used to broadcast information such as cell and network identity, and operating characteristics of the cell.

(b) FCCH- Frequency Correction Channel

The FCCH occupies TS0 for the very first GSM frame (frame 0) and allows each subscriber unit to synchronize its internal frequency standard to the exact frequency of the base station.

(c) SCH- Synchronization Channel

The SCH is broadcast in TS0 of the frame immediately following the FCCH frame and is used to identify the serving BTS while allowing each MS to frame synchronize with the BTS.

CCCH- Common Control Channels

There are three types of common control channels in GSM

(a) PCH- Paging Channel

The PCH provides paging signals from the base station to all mobiles in the cell, and notifies a specific mobile of an incoming call which originates from the PSTN.

(b) RACH- *Random Access Channel*

The RACH is a reverse link channel used by a subscriber unit to acknowledge a page from the PCH, and is also used by mobiles to originate a call. The RACH uses a slotted ALOHA access scheme.

(c) AGCH- *Access Grant Channel*

The AGCH is used by the base station to provide forward link communication to the mobile, and carries data that instructs the mobile to operate in a particular physical channel with a particular dedicated control channel.

DCCH- Dedicated Control Channel

There are three types of dedicated control channels in GSM.

(a) SDCCH- Stand-alone Dedicated Control Channels

SDCCH works like an intermediate and temporary channel that accepts a newly completed call from the BCH and holds the traffic while waiting for the base station to allocate a TCH channel.

(b) SACCH- Slow Associated Control Channel

The SACCH is always associated with a traffic channel or a SDCCH and maps onto the same physical channel.

*(c) **FACCH-*** Fast Associated Control Channel

The FACCH carries urgent messages, and contains essentially the same type of information as the SDCCH.

2.2 Overview of the GPRS System

General Packet Radio Service is a packet-based data network, which is well suited for non-real time Internet usage, including the retrieval of email, faxes, and asymmetric web browsing, where the user downloads much more data than it uploads on the Internet. GPRS support multi-user network sharing of individual radio channels. Thus, GPRS can support many more users than GSM, but in a bursty manner.

GPRS standard provides a packet network on dedicated GSM or IS-136 radio channels. Implementation of GPRS merely requires the GSM operator to install new routers and Internet gateways at the base station, along with new software that redefines the base station air interface standard for GPRS channels and time slots, no new base station or RF hardware is required.

Notice that GPRS is only used for packet data service instead of voice traffic. The GSM techniques are used for voice calls. In fact in a cell that supports GPRS, both GSM and GPRS systems operates in parallel to provide service for packet data and voice traffic. Voice and GPRS can co-exist on the same site, same carrier and same TDMA time slot.

2.2.1 GPRS System Architecture

GPRS specifies a packet switched-core network. GPRS core network was primarily designed to support best-effort data services. This packet-switched core network is IP-based network. This core network illustrated is in figure 2.5.

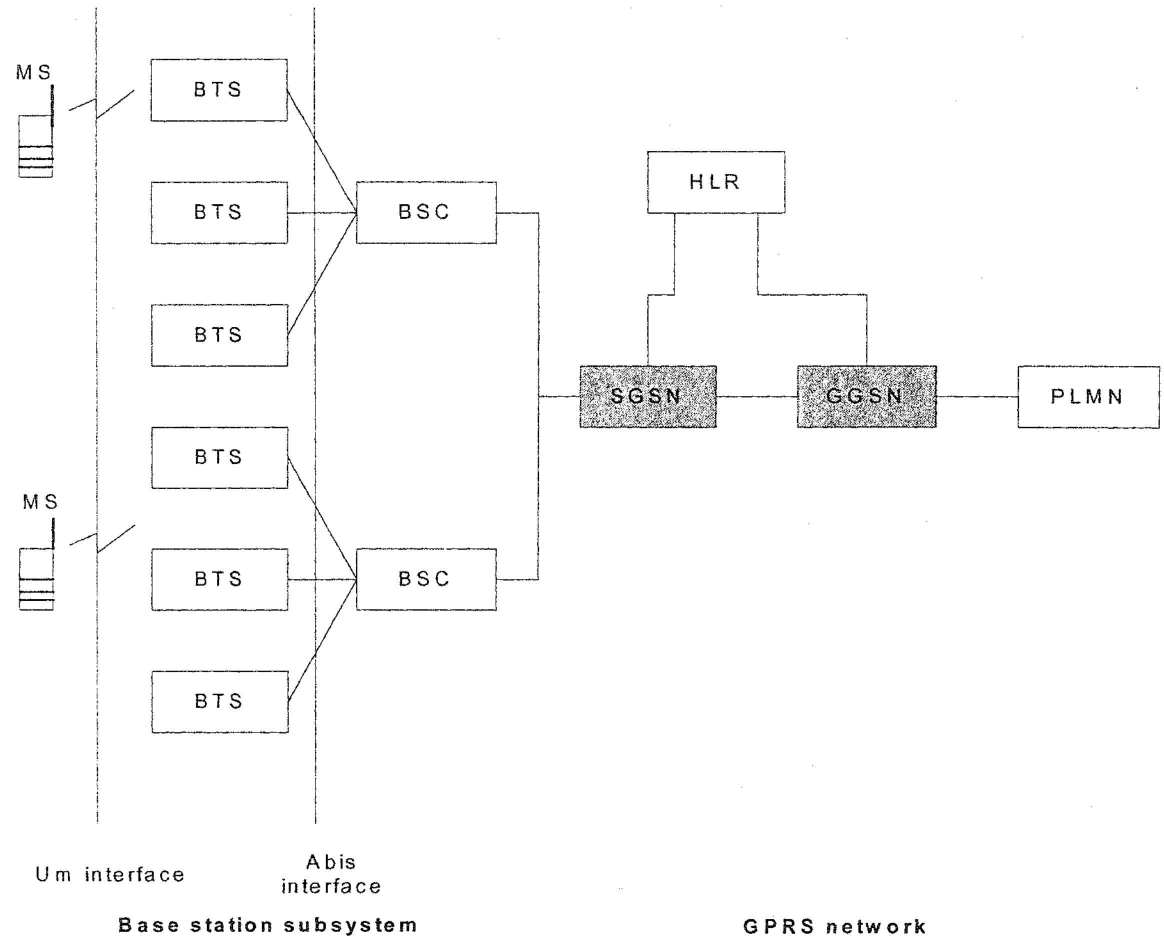


Figure 2.6 The GPRS core network architecture

GPRS ' air interface based on Gaussian Minimum Shift Keying (GMSK) modulation and supports transmission rates from 9.05kbps to 21.4kbps per channel. When all eight-time slots of a GSM radio channel are dedicated to GPRS, an individual user is able to achieve as much as 171.2 kbps (eight time slots multiplied by 21.4 kbps of raw uncoded data throughput).

GPRS allows the subscriber to send and receive data in an end-to-end packet transfer mode, without using any network resources in circuit-switched mode. To support GPRS, a GSM network needs to be enhanced with new functional network element: A new logical network node called GPRS support node (GSN) to provide independent packet routing and transfer within the public land mobile network (PLMN). There are two kinds GSN: the Gateway GPRS Support Node (GGSN) and the Serving GPRS Support Node (SGSN) [3] [10].

GGSN:

The Gateway GPRS Support Node (GGSN) acts as a logical interface to external packet data networks that serve as the gateway router connecting the GPRS core network to other packet data networks. The GGSN is the GPRS equivalent of the shared Inter-Working Function (IWF).

SGSN:

The Serving GPRS Support Node (SGSN) is responsible for the delivery of packets to the MSs within its service area that serves as the access router to the GPRS core network. The SGSN is the GPRS equivalent of the Mobile Switching Centre (MSC) and Visitor Location Register (VLR). The SGSN is at the core of the GPRS network. It connects the radio subsystem with the backbone network through the Base Station System GPRS Protocol (BSSGP). BSSGP supports the connectionless transfer of logical link control (LLC) frames between the BSS and SGSN, providing a signalling and user data interface between the radio interface's RLC/MAC and the SGSN. To transfer LLC frames between the mobile station and the base station system, the BSS relies on the RLC/MAC function.

The GPRS network subsystem is designed to be independent from the radio subsystem. This allows a variety of radio interface technologies to be used, including GSM 900/1800/1900, EDGE and UMTS.

2.2.2 GPRS Coding

GPRS uses four different coding schemes, CS-1 to CS-4 to carry RLC data blocks. These four schemes are defined in [4] are summarized in table 2.1:

Table 2.1 Four different coding schemes in GPRS

Channel Coding Scheme	Data bits and radio block	Data rate per time slot kbps on radio layer	Maximum data rate per 8 timeslots kbps
CS-1	181	9.05	72.4
CS-2	268	13.4	107.2
CS-3	312	15.6	124.8
CS-4	428	21.4	171.2

The main difference between these four coding schemes is the level of protection from transmission errors that they can offer and the maximum throughput that can be obtained. This leads to the inherent trade off between high protection and high throughput. CS-1 offers the lowest throughput and CS-4 the highest. Unfortunately CS-1 offers the best protection from transmission errors and CS-4 offers the worst. The GPRS system dynamically chooses the coding scheme best suited for the transmission conditions at hand.

In contrast to GSM, GPRS allows a MS to use more than one timeslot per Time Division Multiple Access (TDMA) frame; this allows one MS to use up to eight timeslots simultaneously. Furthermore, uplink and downlink channels are allocated separately, which efficiently supports asymmetric data traffic.

2.2.3 GPRS Protocol Architecture

By introducing new packet support nodes and associated protocol stack [3], GPRS is able to operate as a packet mode wireless system.

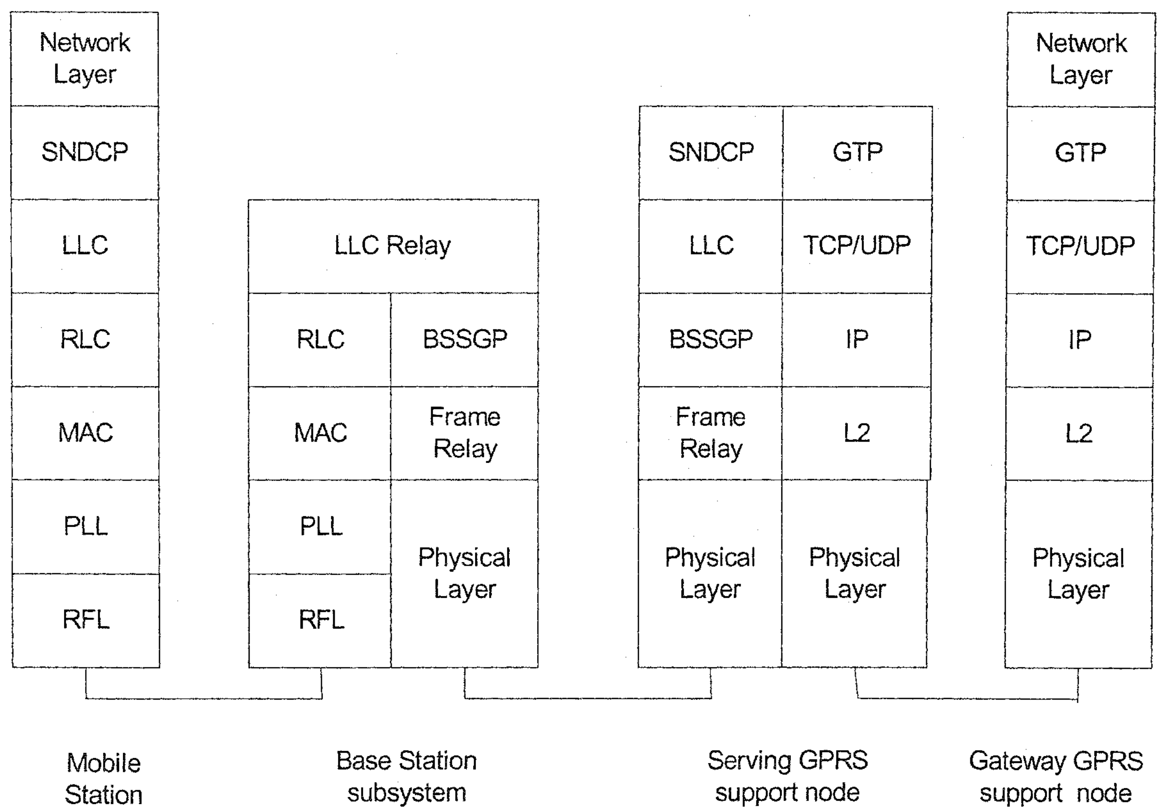


Figure 2.7 GPRS transmission plane

Figure 2.7 shows the transmission plane up to the network layer according to the International Organization for Standardization/Open System Interconnections (ISO/OSI) reference model.

- *GPRS Tunnel Protocol (GTP)* [5] tunnels the PDUs between two GSNs through the GPRS backbone network by adding routing information.
- Below the GTP, the *Transmission Control Protocol /User Datagram Protocol (TCP/UDP)* and the *Internet Protocol (IP)* are used as the GPRS backbone network layer protocols.
- Ethernet, ISDN, or Asynchronous Transfer Mode (ATM) based protocols may be used below IP depending on the operator's network architecture.
- Between the SGSN and MS, the *Subnetwork Dependent Convergence Protocol (SNDCP)* [6] maps network level protocol characteristics onto the underlying Logical Link Control (LLC) and provides functionalities like multiplexing of network layer messages onto a single virtual logical connection.
- Between the MS and BSS, the data link layer has been separated into two distinct sublayers: The Logical Link Control (LLC) [7] and Radio Link Control/Medium Access Control (RLC/MAC) [8] sublayers.
    1. The LLC layer provides a highly reliable logical link between the MS and SGSN.
    2. The RLC layer performs the segmentation and reassembly of LLC Protocol Data Unit into RLC data blocks and ARQ of un-correctable codeword.
    3. The MAC layer control the access signalling (request and grant) procedures for the radio channel, and mapping of LLC frames onto the GSM physical channels.

- Base Station System GPRS Protocol (BSSGP) used to convey routing and QoS-related information between BSS and SGSN.
- The Physical layer is split up into a physical link sublayer (PLL) and a physical RF sublayer (RFL).

In terms of error recovery at different layers, MAC layer attempts to resolve collision of request packets. RLC layer attempts to recover RLC data block errors through a selective repeat ARQ mechanism. LLC layer attempts recovery of LLC frames through a stop-and-wait ARQ mechanism. Link errors unresolved at LLC layer are passed on to higher layers (e.g., transport layer) to resolve. [17]

2.2.4 GPRS LOGICAL CHANNELS

When a network operator decides to offer GPRS-based services within a cell, one or several physical channels from the pool of available channels are dedicated to packet mode transfer. The physical channel dedicated to packet data traffic is called a Packet Data Channels (PDCHs). It represents one time slot in a TDMA frame. According to the requirement for flexible adaptation to different traffic conditions, allocation of PDCHs is based on demand. At least one PDCH (mapped on one physical time slot), acting as a master, the others, acting as slaves, are only used for user data transfer. In order to simplify the logical channel concept, the allocated PDCHs are logically grouped into master and slave channels (MPDCHs and SPDCHs).

MPDCH

The MPDCHs accommodate Common Control Channels (CCHs) that carry the signalling information required to initiate packet transfer. PBCCH and PCCCH are belong to MPDCH.

PBCCH- Packet Broadcast Control Channel

The PBCCH transmits system information to all GPRS terminals in a cell.

PCCCH- Packet Common Control Channel

(a) PRACH- The Packet Random Access Channel

The PRACH is used by MSs to initiate packet transfers or respond to paging messages. On this channel MSs transmit access bursts with long guard times. On receiving access bursts. The BSS assigns timing advance to each terminal.

(b) PPCH- The Packet Paging Channel

The PPCH is used to page an MS prior to downlink packet transfer.

(c) PAGCH- The Packet Access Grant Channel

The PAGCH is used in the packet transfer establishment phase to send resource assignment to an MS prior to the packet transfer.

(d) PNCH- The Packet Notification Channel

The PNCH is used to send a PTM-multicast (PTM-M) modification to a group of MSs prior to a PTM-M packet transfer. The notification has the form of a resource assignment for the packet transfer.

SPDCH

The SPDCHs represent the channels on which user data and dedicated signalling is transferred:

PTCH- The Packet Traffic Channels

(a) PDTCH- The Packet Data Transfer Channel

The PDTCH is a channel allocated for data transfer. One MS may use more than one PDTCH in parallel (multislot operation) for individual packet transfers.

(b) PACCH- The Packet Associated Control Channel

The PACCH is used to convey signalling information related to a given MS such as acknowledgments (ACK) and power control (PC) information. It also carries resource assignment and reassignment messages, either for allocation of a PDTCH or further occurrences of a PACCH. One PACCH is associated with one or several PDTCHs concurrently assigned to one MS.

2.2.5 Services in GPRS

The services that have to be offered by GPRS system are Point-To-Multipoint (PTM) services and Point-To-Point (PTP) services. The four QoS parameters are defined as a subscriber's QoS profile for certain GPRS application [19]. These four parameters are:

- **Precedence** (priority) indicates the relative priority of maintaining the service under abnormal conditions.
- **Reliability** indicates the transmission characteristics requested by an application. I.e., the probability of loss of, duplication of, mis-sequencing of or corruption of data units.
- **Delay** defines the maximum value for the mean delay and the 95-percentil delay to be incurred by data transfer through GPRS network.
- **Throughput** donates maximum bit rate and the mean bit rate.

2.3 Overview of EGPRS/EDGE System

EGPRS is a derivative of GPRS [10]. The standardisation of EDGE took evolutionary steps towards 3G. EDGE Phase One was the original concept aimed at enhancing GPRS and GSM circuit switched services. The main focus was on the non-real-time packet data services. This is also called EGPRS. The EDGE Phase Two started towards the completion of Phase One. The main concept was based on providing real-time services in an end-to-end packet mode.

The Enhanced General packet Radio System (EGPRS) offers higher data transmission rate than both GPRS (2.5G) and GSM (2G) standards. These higher data rates are achieved by introducing new coding and modulation schemes in the air interface. The new air interface in EGPRS is called Enhanced Data Rate for GSM Evolution (EDGE). EDGE provides smooth evolution of GSM and TDMA/136 towards 3rd generation capabilities.

To minimize the impact on current systems, EGPRS uses the same packet-switched core network as GPRS, but to achieve higher data requirement, 8-Phase Shift Keying (8-PSK) modulation is introduced in EGPRS' air interface. Unlike the GPRS core network that was primarily designed to support best-effort data services, the EGPRS core network will provide integrated services, which means will support the different services classes and different traffic classes.

2.3.1 EGPRS/EDGE System Architecture

Some GSM operators anticipate the demand for packet data services, so plan to overlay the EGPRS network on the existing cellular voice system. The packet-switched network and circuit-switched network coexist in parallel. Voice or circuit-switched data follows the path from the BSS to the MSC, and then to the PSTN. Packet-switched data traffic, on the other hand, follows the path from the BSS to the SGSN, then over the GPRS network to the GGSN, and to a packet data network. (Figure 2.8 a).

But from the network integration, utilization and operation costs point of view, it is desirable to eliminate a separated circuit-switched network and move all the traffic to the IP-based GPRS network. (Figure 2.8 b)[1]. To achieve this goal, further enhancement to the air interface and the core EGPRS network are necessary. The functionality of SGSN and GGSN needs to be enhanced to E-SGSN and E-GGSN.

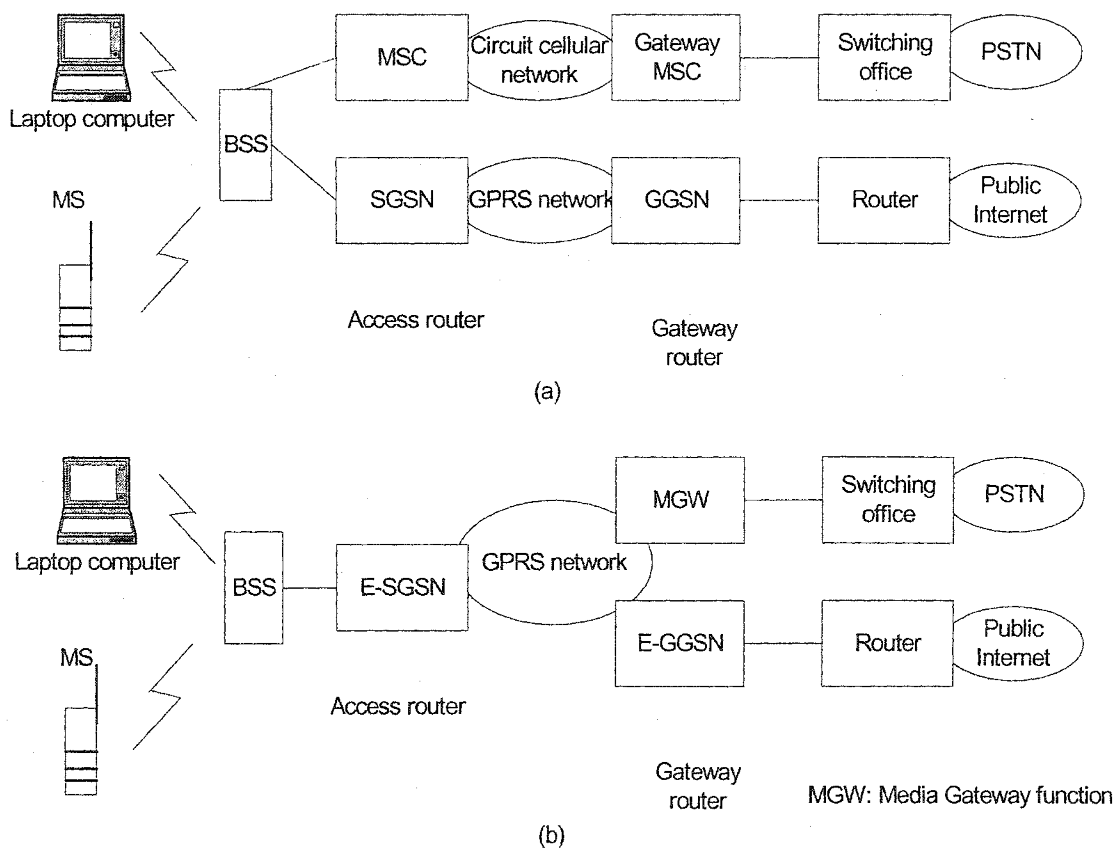


Figure 2.8 (a) Coexistence of the circuit switched cellular system and the EGPRS system (b) The EGPRS system supporting integrated services.

2.3.2 EGPRS Coding

In EGPRS, to increase data rates in existing GSM networks, 8-PSK (Phase Shift Keying) [9] [12] modulation was introduced in addition to the existing Gaussian Minimum Shift Keying (GMSK) modulation. So, there are nine new modulation and coding schemes designated MCS1- MCS9 with varying degrees of error control protection in EGPRS. Which scheme is selected for network access is dependant on the instantaneous demands of the network and the operating conditions. In fact, dynamic link adaptation between different coding and modulation schemes are of great interest for EGPRS in order of achieve superior delay/throughput performance over a wide range of operating conditions [13]. Because of the higher data rates and relaxed error control covering in many of the selectable air interface formats, the coverage range is smaller in EDGE than in HSDRC or GPRS.

In 8-PSK, three bits are mapped onto a symbol, which results in eight points in the constellation diagram, The higher number of transition states in the constellation diagram results in an increase in the symbol error rate and subsequently an increase in the Block Error Rate (BLER) under equivalent radio conditions and code rates compared with GMSK. So, in 8-PSK mode, despite the increase in bit rate by a factor of 3, the effective increase in data rate, under equivalent conditions, is lower than 3 due to the higher protection that is required for the user data. Furthermore, digital signal processing and RF devices become more complex and this leads to some compromise between implementation complexity and performance.

By using GMSK or 8PSK modulation, EGPRS supports transmission rates ranging from 8.8kbps to 59.2 kbps per channel. (Table 2.2[9]) EGPRS allows combining up to eight channels per user, so as to support data rates over 384kbps.

Table 2.2: Coding parameters for the EGPRS coding schemes

Scheme	Modulation	Maximum rate, kbps
MCS-1	8-PSK	8.8
MCS-2	8-PSK	11.2
MCS-3	8-PSK	14.8
MCS-4	8-PSK	17.6
MCS-5	8-PSK	22.4
MCS-6	GMSK	29.6
MCS-7	GMSK	44.8
MCS-8	GMSK	54.4
MCS-9	GMSK	59.2

2.3.3 Services supported by EGPRS

Four classes of services that are currently defined for Universal Mobile Telephone Service (UMTS) need to be supported by EGPRS [1]. "The rise of these new multimedia services with stringent QoS requirement is converting the existing GSM/EDGE networks into real 3G networks." [24]

- Conversational Class

  The characteristic of this class is: preserve time relation between information entities of the stream. The pattern is conversational (stringent and low delay).

  The main application is voice.

- Streaming Class

  The characteristic of this class is: preserve time relation between information entities of the stream.

  The main application is stream video.

- Interactive Class

  The characteristic of this class is: preserve payload content. The pattern is request response.

  The main application is web browsing.

- Background Class

  The characteristic of this class is: preserve payload content. Destination is not expecting the data within a certain time.

  The main application is background download of e-mail.

2.3.4 Fast Packet Control Channels in EGPRS

To provide service both to packet data and voice in EGPRS system, additional capabilities are required. Consider a packet voice service, it is desirable for a voice user to release the channel during its silent period and regain access only at the beginning of the next talk spurt. The residual capability may be used to multiplex additional delay-insensitive services along with the packet voice users. To achieve this multiplexing benefit, the following capabilities are necessary:

- Fast uplink access during an ongoing session
- Fast resource assignment for both uplink and downlink.

A new set of common control channels are proposed by EGPRS supplier to provide the additional capabilities. These channels are similar to GPRS common control channels required for call set-up with one important difference: they are designed for in-session control. In-session control has a more stringent delay requirement than session set-up control and has smaller signalling overhead, which makes it feasible to meet these delay requirements. The channels are:

F-PACH: Fast Packet Access Channel (for the uplink)

The structure of F-PACH is similar to that of the PRACH in EGPRS, but F-PACH is only used for ongoing calls. The *fast packet channel request message* carried in F-PACH contains information on the specific TBF being referenced. Based on this information, the system can uniquely identify the MS and its specific application, and therefore quickly assign the necessary uplink resource.

F-PCCH: Fast Packet Control Channel (for the downlink)

F-PCCH used to transmit access grant and polling messages to specific mobiles. The F-PCCH is split into two logical channels:

> ***F-PAGCH:*** Fast Packet Access Grant Channel
>
> The F-PAGCH is used to response to access requests received from F_PACH. This response is typically an assignment message that specifies the channels' USFs and other parameters for a set of MS.
>
> ***F-PPCH:*** Fast Packet Polling Channel
>
> The F-PPCH is used to poll different mobiles.

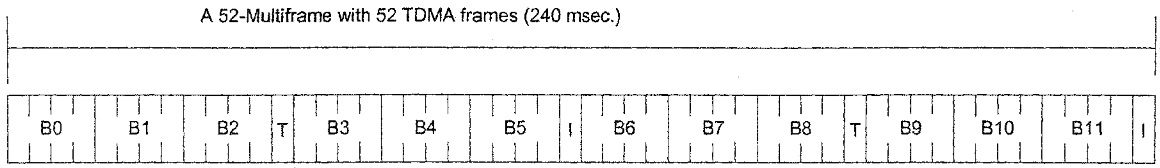
Chapter 3

Resource Allocation Scheme in GPRS/EGPRS

The allocation of radio resources in GPRS and EGPRS is an active research area and there are many papers that address this problem from different perspective. In this chapter, the general merit of the system that is contributes to the resource allocation schemes in GPRS and EGPRS is introduced first. And then, some algorithms used for resource allocation are investigated.

3.1 GPRS/EGPRS Frame and Data Structures

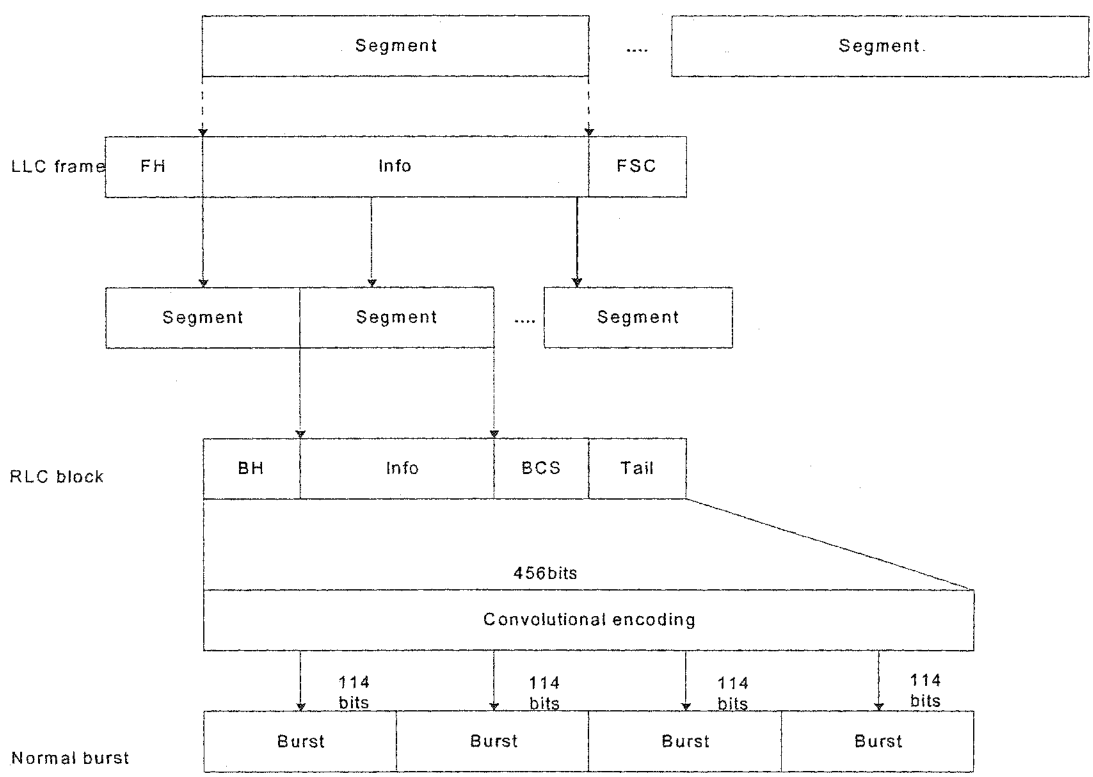
There are two multiframes defined in GSM: 26-multiframes in traffic channels and 51-multiframes in control channels. Whereas, in GPRS, a 52-multiframes has been chosen for PDCH. A multiframe structure for PDCHs consisting of 52 TDMA frames is shown in Figure 3.1 [11]. The 52-Multiframe consists of 13 blocks of 4 consecutive TDMA frames. Out of these 52 TDMA frames, 48 TDMA frames (12 blocks, B0 to B11) are used for the allocation of Packet Data Channels or Packet Control Channels. Two TDMA frames are idle and the rest two are used for Packet Timing Advance Control Channel (PTCCH) [14]. GPRS and EGPRS employ the same time slot and frame structure as GSM. Each GSM frame is 4.615 ms and consists of eight time slots.



T represents the frame carrying Packet Timing Advance Control Channel
I represents idle frame

Figure 3.1 52-Multiframe Structure for PDCH

An LLC PDU is divided into an appropriate number of RLC blocks, and each RLC block is coded and then interleaved over four GSM time slots in four consecutive GSM frames. Therefore, the basic radio packet in GPRS is RLC/MAC block, and it is transmitted in the physical layer as four GSM time slots. The four GSM time frames (20ms) are defined as one logical frame [3].



FCS: Frame check sequence BH: Block header FH: Frame header BCS: Block check sequence

Figure 3.2 Packet transformation data flow

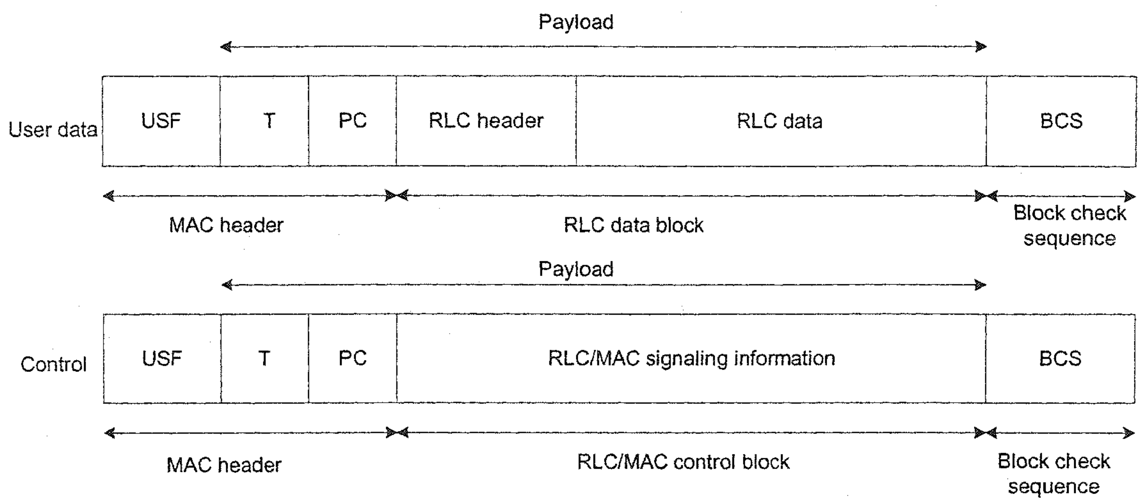


Figure 3.3 Radio Block Structure

Figure 3.3 shows the radio block structure for user data and control messages. In user data message, each radio block consists of a MAC header, an RLC data block and a Block Check Sequence (BCS). In control message, each radio block consists of a MAC header, an RLC/MAC control block, and a BCS. BCS is always carried by four normal bursts.

3.2 Air Interface Protocol

The air interface protocol of GPRS/EGPRS is concerned with communications between the MS and BSS at the physical, MAC and RLC protocol layers. The RLC/MAC sub-layers allow efficient multi-user multiplexing on the shared packet data channels and utilize a selective ARQ protocol for reliable transmission across the air interface [3]. The reason why emphasis is put on the air interface is that performance of packet data services is strongly influenced by efficient allocation of the scarce radio resource.

To meet the data service requirement, the GPRS/EGPRS air interface standard had been designed for "always-on" packet based wireless service. The radio resource will be used on the need basis and will be released immediately after the transmission of the packets. Owing to this, multiple users can share one physical channel at the same time. The GPRS radio interface consists of asymmetric and independent uplink and downlink channels. In a certain TDMA time slot, a PDCH uplink may carry data from one MS and the downlink data to another MS.

The special features of the physical protocol layer of GPRS and EGPRS that contribute to build a packet switched system have been introduced in the previous chapter. The RLC/MAC protocol layer will be highlighted in the next section.

3.2.1 RLC/MAC Protocol in GPRS/EGPRS

The RLC/MAC layer provides a bit pipe that is responsible for transferring the LLC PDU between the BSS and MS [1]. This layer proposed a selective Automatic Repeat Request (ARQ) type protocol with a Slotted ALOHA random-access-based packet reservation mechanism for uplink transmission.

The RLC layer is responsible for the transmission of data blocks across the air interface and the Backward Error Correction (BEC) procedures: selective retransmission of un-correctable blocks.

The MAC layer is derived from a Slotted ALOHA protocol and operates between the MS and BTS. It is responsible for access signalling procedures for the radio channel governing the attempts to access the channel by the MSs, and the control of that access by the network side [16]. It performs contention resolution between channel access attempts, arbitration between multiple service requests from different MSs, and medium allocation to individual users in response to service request. Implementing multislot MAC layer operation ensures high flexibility. More than one PDCH (each corresponding to a time slot in a TDMA frame) can be used by one MS for packet data transfer. It also defines the procedures that enable multiple MSs to share a common PDCH.

3.2.2 Capacity on Demand

The concept of capacity on demand has been introduced in order to meet the requirement of a cell that can provide GPRS service where there are few or no GPRS users has not the need for permanently allocated resources [3]. Load supervision is done in the MAC layer to monitor the load on the PDCH(s), and the number of allocated PDCHs in a cell can be increased or decreased according to demand. Thus the physical channels available in a cell are shared dynamically between GPRS and other GSM services. Unused channels can be allocated as PDCHs to increase the overall QoS for GPRS. If other services with higher priority request resources, de-allocation of PDCHs can take place.

3.2.3 Multiplexing Capability of RLC/MAC Layer

To serve bursty data traffic in a spectrally efficient manner, The GPRS/EGPRS RLC/MAC layer is designed to support multiple data streams (to/from different users) on the same PDCH, and to support a given data stream (to/from one user) on multiple PDCH [15]. The special features, which could enable this multiplexing, described next.

TBF (Temporary Block Flow)

Any data transfer in GPRS/EGPRS is accomplished using an entity called a TBF [1]. A TBF is a virtual connection that supports the unidirectional transfer of LLC PDUs on packet data physical channels between an MS and the BSS. It is maintained for the duration of the data transfer and comprises a number of RLC/MAC blocks. A TBF can be open-ended or close-ended. A close-ended TBF limits the MS to sending certain amount of data that has been negotiated between itself and its serving BTS during initial access. An open-ended TBF is used to transfer an arbitrary amount of data. Each TBF is identified by a TFI that is introduced next.

TFI (Temporary Flow Identifier)

A TFI is assigned to each TBF that is transmitted to or from an MS. The assigned TFI is unique among concurrent frame transfer sequences in a cell. In this way, the Selective ARQ on the RLC lever can be implemented by frame numbering. The TFI further contains job identification in order to allow multiplexing several jobs onto one Packet Traffic Channel. A TFI is 7 bits long for the uplink and 5 bits long for the downlink. The TFI is assigned by the BSS and is unique in each direction.

In downlink, multiple data streams transfer on the same PDTCH by assigning each data stream a unique TFI. Each MS listens to the assigned downlink channels and only accept the RLC blocks with it's own TFI. In this way, the BSS can communicate with the MS on several channels that are assigned to the MS, and also can multiplex several data frames destined to different MSs on the same physical channel.

USF (Uplink State Flag)

Uplink multiplexing is accomplished by assigning each data stream a set of channels and the use of 3-bit USF. The USF is three bits long means that it enables the coding of eight different USF states that implies up to eight different data streams can be multiplexed on the same channel. To an MS, the USF marks the part of the channel it can use for transmission. An MS monitors all the downlink traffic channels try to find its USF. If its USF appears, the MS uses the corresponding uplink channel in the next logical frame to transmit data. [1] This mechanism is illustrated in Figure 3.4

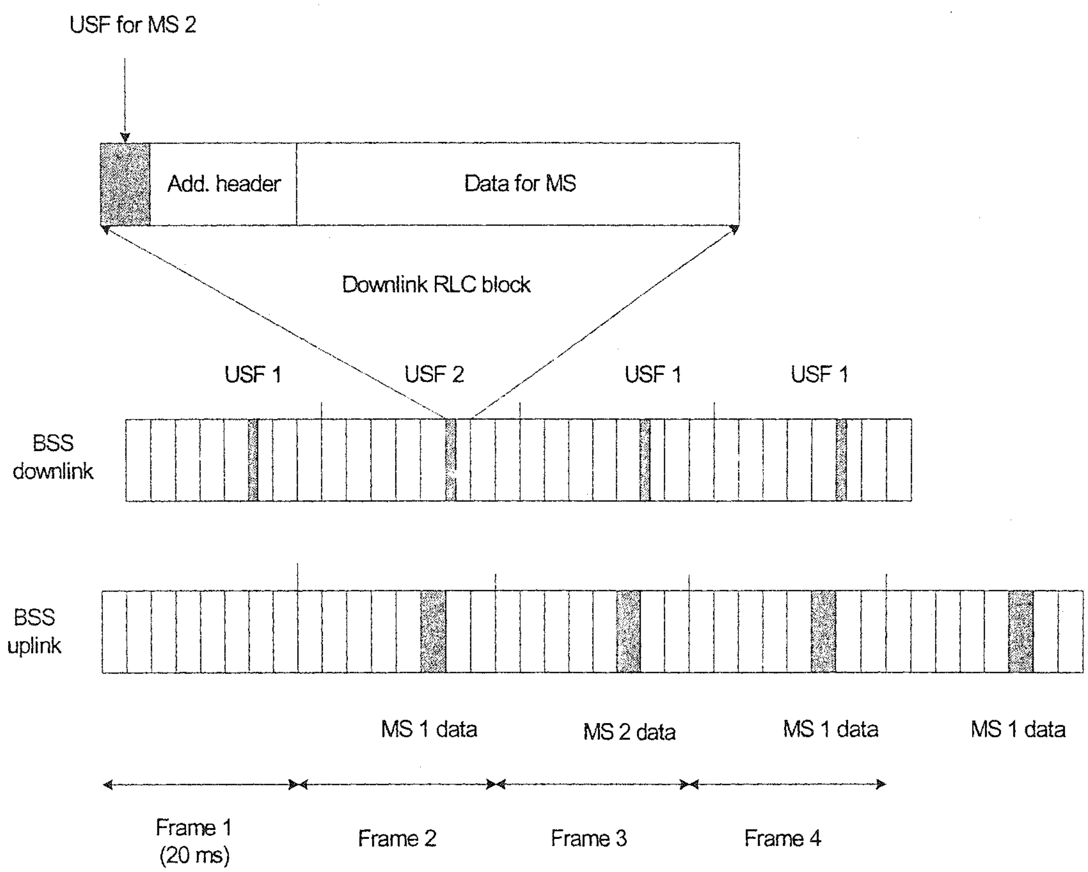


Figure 3.4 the principle of uplink multiplexing

The example of figure 3.4 shows after MS1 detected its USF in channel 6 in the downlink in frame 1, it can use the corresponding uplink channel in frame 2. MS2 detected its USF in the same channel in the downlink in frame 2; MS2 instead of MS1 can use this channel in the uplink in frame 3. This process repeats to realize the multiplexing of different users on the same physical channel.

3.3 Resource Allocation Schemes in GPRS/EGPRS

There are three channel allocation schemes used in GPRS/EGPRS. These are:

- Fixed allocation.
- Dynamic allocation.
- Extended dynamic allocation.

The fixed and dynamic allocation schemes will be discussed next. The detail of these procedures is given in [8].

3.3.1 Fixed Allocation Scheme

In Fixed Allocation Scheme, the resource allocated to a particular MS is fixed for the duration of the call without the use of USF. The network indicates the MS about the starting frame, timeslot assignment and the corresponding ALLOCATION_BITMAP in the *Packet Uplink Assignment Message* over PAGCH. The MS uses those blocks for radio block transmission. A unique TFI is assigned to each ongoing TBF and is thereafter included in each RLC data and control blocks related to that TBF. Because each radio block contains TFI, all received radio blocks are correctly associated with a particular LLC frame and a particular MS.

3.3.2 Dynamic Allocation Scheme

In Dynamic Allocation Scheme, the resource allocated to a particular MS is dynamic for the duration of the call. The Packet Uplink Assignment Message includes the list of PDCHs allocated to the MS and the corresponding USF values per PDCH. The MS monitors the USFs on the allocated PDCHs and transmits radio blocks on those that currently bear the USF value reserved for the usage of the MS. It is evident that in

Dynamic Allocation, the possibility of wastage of radio resource is very less as compared to the Fixed Allocation case.

3.3.3 Extended Dynamic Allocation Scheme

The extended dynamic allocation is a simple extension of dynamic one adapted to deliver large volume data packets. A USF value indicates the assigned block periods on several Pecks. [18]

3.4 Operation Model of Dynamic Allocation Scheme

Efficient and flexible utilization of the available spectrum for packet data traffic can be obtained by the use of MAC protocol. The MAC protocol is characterized by dynamic bandwidth allocation and multislot operation. The bandwidth may vary from allocating one to eight time slots in each TDMA frame. The operation of dynamic allocation scheme will be discussed next.

3.4.1 Mobile-Originated Transfer

Multiple Accesses

An MS initiate a packet transfer by sending a random access request on the PRACH. Upon correctly receive the access request, the BTS response to PRACH with PAGCH. Base on these requests, PDTCH slots are dynamically assigned to the MS by BTS. Allocation can be done on a one-time slot per GSM TDMA frame basis (called *single slot operation*) or multiple time slots per GSM TDMA frame basis (called *multi-slot operation*) [17]. GPRS allows two types of access procedures for data transfer: one-phase and two-phase access. Figure 3.4 is the message flow of these two types of access procedure.

In **one-phase access**, after receiving a packet channel request, the BSS replies with a packet uplink immediate assignment message over a Packet Access Grant Channel (PAGCH). This message contains the resource assignment for the MS.

In **two-phase access**, after receiving a packet channel request, the BSS first replies with a packet uplink assignment message over a Packet Access Grant Channel (PAGCH) and assigns the MS a signalling channel-Packet Associated Control Channel (PACCH). The MS then sends a detailed resource request using this signalling channel. This request contains the Temporary Logical Link Identifier of the MS and details of the service requested. The BSS then assigns the required resources to it using another packet uplink assignment message in PACCH. The MS can begin data transmission only after receiving this assignment.

The essential difference between these two procedures is that "in the one-phase procedure, the uplink data transfer begins concurrently with the service negotiation and mobile verification, whereas in the two-phase procedure, the uplink data transfer begins only after the mobile verification and service negotiation is complete."[1] The two-phase access procedure is currently used in the 2G GSM systems.

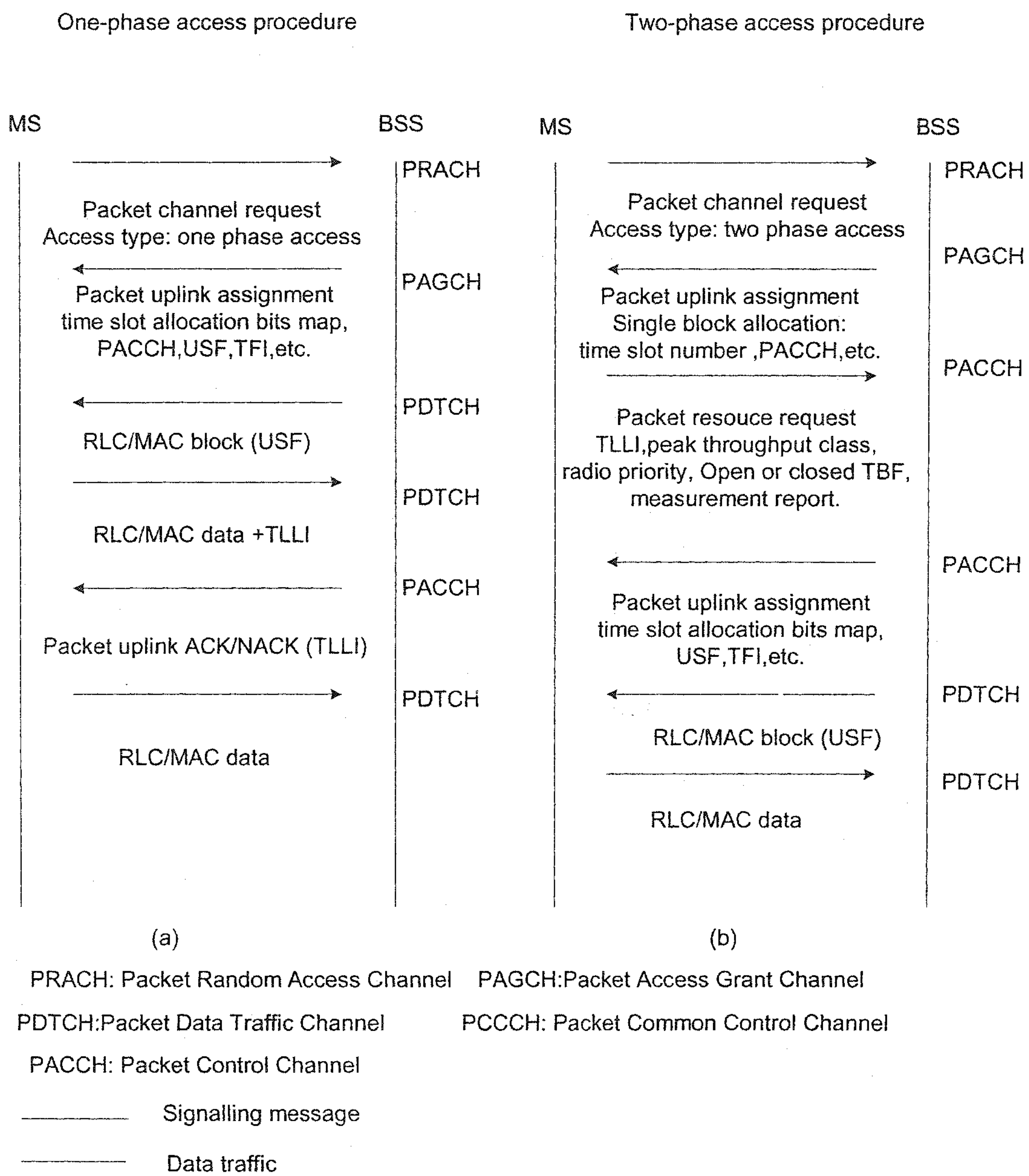


Figure 3.5 (a) message flow for one-phase access
(b) message flow for two-phase access

If the MS does not receive the response to its access request, a retransmission procedure will start after a random back off time.

Uplink Data Transfer

With multislot channel reservation scheme used on packet data traffic, the available spectrum can be utilized efficiently and flexibly. Data blocks from one MS can be sent to BSS on different PDTCH at the same time, in this way, meet the high data rate requirement from users. The bandwidth of GPRS/EGPRS systems may vary by allocating one to eight time slots in each TDMA frame.

With the use of 3-bit USF at the beginning of each radio block sent on the downlink points to the next uplink radio block, up to seven (USF =111 is reserved) data transfers can be multiplexed on each channel.

Acknowledgement

After transmission in the reserved time slots is completed, and acknowledgment (ACK) is sent by the BTS. When erroneous or missing happens, a negative acknowledgment (NACK) is sent. If the MS does not receive an ACK within a certain time, frame transfer recovery is started by a new random access.

3.4.2 Mobile-Terminated Transfer

A BSS initiates a packet transfer by sending a packet-paging request on the PPCH. If the BTS knows the location of the MS, it will directly reserve the uplink slots for uplink transmission or give an indication of downlink slots for data reception. Thus the MS may respond by either immediately starting data transmission/reception of reserved slots or sending a random access request on the PRACH. If the BTS knows the location of the MS only to a certain degree of probability, it does not reserve the uplink slots but to reserve a single slot for a paging response. When the MS gets the page, it initiates the random access and asks for a reservation.

Since TFI is included in each radio block, it is possible to multiplex radio blocks destined for different MSs on the same PDCH downlink. It is also possible that more than one PDCH is available for the downlink traffic. If the MS is capable of monitoring multiple PDTCHs, blocks belonging to the same frame can be transferred on different PDTCHs in parallel.

3.5 Current work on Resource Allocation Schemes of GPRS/EGPRS

Several studies have investigated the allocation of radio resources in GPRS and EGPRS.

In [14], two algorithms for the Fixed and the Dynamic Channel Allocation Schemes in GPRS are proposed, namely the First Fit (FF) and the Best Fit (BF) algorithms. The author of [14] made a comparison between the allocation and Bin Packing problems and found these two problems are closely related to each other.

- "N channels are equivalent to N bins."[14]
- "Resource allocated to user I is equivalent to size of item." [14]

In FF, the item is placed into the first bin (lowest indexed) that it will fit.

In BF, the item is place into a partially filled bin with the highest level.

In the Fixed Allocation scheme, the resource allocated to users is fixed throughout the packet transfer period and is equal to the minimum QoS requirement. Whereas in the Dynamic Channel Allocation scheme, in addition to the minimum QoS requirement, extra blocks on the same PDCH (which are not currently allocated to any other users) are allocated to users (who are currently engaged in this PDCH) fairly in a dynamic manner.

The simulation result is only emphasis on Request Blocking Probability. The conclusion of [14] is "the dynamic channel allocation scheme performs better than the fixed channel allocation scheme"[14].

In [20] The efficiency of different algorithms used for resource allocation in data transfer over EGPRS network is investigated. There are five algorithms are implemented " Round Robin (RR), Iterative Round Robin (IRR), Longest Queue (LQ), Total File Length (FTL) and Oldest Queue (OQ)" The simulation result is only emphasis on Delay, and the conclusion of this paper is IRR and OQ algorithms produced the lowest delay.

The emphasis of [18] is put on the service disciplines to provide best possible performance for WWW traffic in GPRS. Four scheduling algorithms for the uplink World Wide Web traffic in GPRS/EGPRS are simulated. These four algorithms are "dynamic, First Come First Served (FCFS), FCFS with priority and FCFS with windows".

The aim of [22] is to study the behaviour of the GPRS mechanisms when the application layer carries packetized voice, and determine if the quality of service provided by the network is suitable for the VoIP service. The result of the study is " GPRS can increase the capacity of a traditional circuit switched system to 125-150%, but at the price of some controlled degradation of the quality of voice caused by increased packet losses and higher delays." [22]

In [17], the author builds a mathematical model to evaluate the performance of the RLC/MAC protocols in GPRS.

The performance of different traffic models and for criteria such as Packet Efficiency, Packet Loss, Buffer Overflow, Delay and Call Blocking will be investigated in this work.

Chapter 4

Dynamic Resource Allocation Under Different Traffic Models in GPRS/EGPRS

In this chapter a simulation model is written in C++ to evaluate the effect of dynamic resource allocation in four different traffic models in GPRS/EGPRS.

These four traffic models are: Uniform, Gaussian, Two state Markov Chain, Modified Two state Markov Chain (during the ON period, the instantaneous rate change uniformly). In each model, users instantaneous data rate change every 4 multiframes. Based on the requirement of user's data rate, the system provides fixed and dynamic allocation of radio resources in two policies. From the comparison of these two schemes of allocation, the benefit of dynamic allocation will be demonstrated. A parameter called Capacity Assignment Parameter is proposed into this simulation model. The result under different value of Capacity Assignment Parameter will be discussed.

Simulation Assumptions:

Most recent mobile systems (such as IS-136 and GSM) provide voice service with tolerable call dropping probabilities around 2%, speech rate at 13kbps, data rate up to 9.6 kbps. To compare these integrated data services with GSM, we assume their characteristics are similar to GSM. The following assumptions are made:

- We have considered only a single cell without adaptive antenna, no multi-cell interference and no hand-over.

- We have not considered any retransmission.
- We have considered only uplink direction of data packets transmission.
- This simulation is based on one window that is one GSM multiframe-- Admission decision window. New calls generation based on window basis. In every 4 windows, users increase or decrease their instantaneous rate.
- All users begin their generation in the first 100 windows randomly. Once generated, user stays on the system for whole simulation duration. Which means a user who started a call in one window will not generate again in the rest of the simulation.
- Concentrate only on the packet generation and transmission processes.
- To make this simulation compatible with current GSM network system, we assume Channel Bit Rate = 13kb/s.
- GSM frame period = 4.615ms
- GSM slot period = 4.615ms/8 = 0.577ms
- 114 data bits in one slot
- The Number of Carrier = 8
- Slot number in one frame= 8
- The number of physical channels available = $8 \times 8 = 64$
- One simulation programme iteration = One simulation window = 1 GSM multiframe = 26 frames.
- One multiframe period = 120 ms.
- Simulation length = Number of Iteration = 10,000 windows.
- Total Slots = 10000(windows)$\times$26(frames/window)$\times$8(slots/frame) = 2,080,000slots.
- Total time = 2,080,000slots$\times$0.577ms = 1,200,160ms

4.1 Parameters and definition

4.1.1 Input Parameters:

There are many input parameters in this simulation. Various performances are compared and then analysed based on the different settings of these input parameters.

- User's average data rate ($\overline{R}$):

Four average data rate classes are defined in this simulation to simulate service classes (Conversational Class, Streaming Class, Interactive Class and Background Class) that need to be supported by EGPRS. These four levels of data rate are: 8kb/s ($\overline{R1}$), 16kb/s ($\overline{R2}$), 32kb/s ($\overline{R3}$) and 64kb/s ($\overline{R4}$). Each user has to clarify his $\overline{R}$ at the beginning of the transmission.

- Variance of user's data rate (σ^2):

Each user has to clarify his σ^2 at the beginning of the transmission.

- Capacity Assignment Parameter (θ):

$$\theta = \frac{\text{System initially assigned user's capacity}}{\text{user's average data rate}}$$

There are four θ values in this simulation:

$$\theta 1 = 0.25$$
$$\theta 2 = 0.5$$
$$\theta 3 = 0.75$$
$$\theta 4 = 1$$

The value of θ means the rate that system can meet the user's data rate requirement (it is one of the QoS requirements) and greatly affect the system performance. Small θ means more users will be accepted by the system but with worse service. High θ means fewer users will be accepted by the system but with better service.

- User's instantaneous data rate (R):

  Instantaneous data rate changes every 4 windows according to different traffic models, average data rate and variance.

- Normalized Offered Load or traffic intensity (ρ):

  ρ is the ratio of the summation of system initially assigned data rate of all active users(accepted or blocked) over the maximum data rate in the system.

$$\rho = \frac{\theta \times \sum_{allusers} \overline{R}}{8(carriers) \times 8(slots/carrier) \times 13Kbps}$$

  This ρ will meet the QoS requirement (symbolized by θ).

In this simulation, two kinds of packet rate combination (coming from different service classes) will be discussed.

1) $10\%(\overline{R1}) + 70\%(\overline{R2}) + 15\%(\overline{R3}) + 5\%(\overline{R4})$ Will be named "low rate type".

2) $10\%(\overline{R1}) + 40\%(\overline{R2}) + 35\%(\overline{R3}) + 15\%(\overline{R4})$ Will be named "high rate type".

Note: based on the different combination of user groups, the same ρ can have different result.

- User's buffer length (BL):

  Each user has a buffer. The larger the buffer size, the smaller the Buffer Overflow, but the longer the Queuing Delay. Two types of buffer will be discussed in this simulation:

  1) Buffer length of each user is the same:

     Buffer length of 2, 4, 8 and 16 packets will be simulated separately. The different result brought by the different buffer length will be compared.

  2) Buffer length of each user is proportional to user's average data rate:

$BL_i = 0.05\% \times \overline{R_i}$

$\overline{R1} = 8{,}000\text{bits/s},\ BL_1 = 4$ packets

$\overline{R2} = 16{,}000\text{bits/s},\ BL_2 = 8$ packets

$\overline{R3} = 32{,}000\text{bits/s},\ BL_3 = 16$ packets

$\overline{R4} = 64{,}000\text{bits/s},\ BL_4 = 32$ packets

- Iteration number: In order to have a reasonable result, the total number of iterations should be large enough. Here we set it 10,000.
- To get rid of the effect of some random extreme numbers, the simulation is repeated ten times, and the mean of all these result is taken as the final simulation result.

4.1.2 Output Parameters:

- Instantaneous rate of user (i): R_i
- Number of packets generated by user (i) in one iteration of 26 frames : δ_i.

$$\delta_i = \frac{R_i(bits/s) \times 0.12s}{114bits}$$

- Number of slots assigned to user (i) in one iteration of 26 frames: S_i.
- Instantaneous queue length, or buffer content of User (i): B_i.

B_i' : The number of packets that are left in its buffer in last iteration of user (i).

$$B_i = B_i' + \delta_i - S_i$$

4.1.3 Performance Parameters:

- **Packet Efficiency**

The average efficiency of the system is defined as the ratio of the total number of packets successfully transmitted by all the users in the system during these 10000 windows to the maximum number of packets that can be transmitted continually on the channels. Assume there are N accepted users.

$$\overline{\eta} = \frac{\sum_{i=1}^{N} \text{All Transmitted Packets from all accepted users in the whole simulation}}{\text{Number of Iteration} \times \text{Number of Carrier} \times \text{Slots in One Carrier} \times \text{Frames In One Multiframe}}$$

Variance of average efficiency: $\sigma_{\eta}^{2} = \dfrac{\sum_{i=1}^{N} (\eta_i - \overline{\eta})^2}{(N-1)}$

- **Packet Loss**

Assume there are N accepted users.

In one simulation window if the queue length of user (i) is longer than the buffer length of user (i), packet loss will happen. And $(B_i - BL_i)$ is the number of lost packets of user (i) in this window.

User (i)'s packet loss rate: $PL_i = \dfrac{\sum_{j=1}^{M} (B_{ij} - BL_{ij})}{\text{All generated packets of user i}}$

where M denotes the active iteration number of user (i)

Average packet loss rate: $\overline{PL} = \dfrac{\sum_{i=1}^{N} PL_i}{N}$

Variance of average packet loss: $\sigma^2 = \dfrac{\sum_{alluser} (PL_i - \overline{PL})^2}{(N-1)}$

If $B_i > BL_i$, after calculating Lost Packets, the simulation program reset $B_i = BL_i$

- **Buffer Overflow**

Assuming there are N accepted users.

Each user has an Overflow Counter $OverFlow_i$. This counter is incremented by 1 ($OverFlow_i = Overflow_i + 1$) in some certain iterations whenever the buffer content(B_i) exceeds the buffer limits(BL_i), which means Packet Loss happens. At the end of all iteration, we add all such counters and obtain OV_i.

User (i)'s buffer overflow rate:

$$OV_i = \frac{OverFlow_i}{\text{Total Iteration Number - Useri's Iteration Start Number}}$$

Average buffer overflow: $\overline{OV} = \frac{\sum_{i=1}^{N} OV_i}{N}$

Variance of average buffer overflow: $\sigma^2 = \frac{\sum_{i=1}^{N} (OV_i - \overline{OV})^2}{(N-1)}$

- **Queuing Delay**

Assume there are N accepted users.

The queuing delay is defined as the time duration from the instance of the packets getting into user's buffer to the instance that these packets getting out and transmitted to the channel.

Queuing delay of user (i): $B_i = \frac{\sum_{j=1}^{M} \text{buffer content of user (i) at iteration j}}{\text{All generated packets of user i}}$

where M denotes the active iteration number of user (i)

Average queuing delay: $\overline{B} = \frac{\sum_{i=1}^{N} B_i}{N}$

Variance of average queuing delay: $\sigma^2 = \frac{\sum_{i=1}^{N} (B_i - \overline{B})^2}{(N-1)}$

Call Blocking

The number of blocked users (CBL)

Once this simulation run at certain load and certain number of users, more users can not be accepted, Call Blocking will happen, in the simulation program, a certain counter is incremented by 1: CBL=(CBL+1)

$$\text{Call Block Probability} = \frac{CBL}{\text{total number of users}}$$

4.1.4 Other Definitions

- Slot (Physical Channel): Each slot has two contents:

    1. Slot Load: ranges from $\frac{0}{26}$ to $\frac{26}{26}$, and means how many times this specific slot is occupied (0-26) by users in 26 frames.
    2. User information in this slot: User Identification Number and the number of times this users occupy this slot. There can be more than one user information in this part and to simulate the USF in GPRS/EGPRS.

- Carrier Load: ranges from 0 to 8.

  Carrier load is the sum of the Slot Load of the eight slots in this carrier.

- User's Load

User's Load is defined as $\frac{\text{users data rate}}{\text{GSM bit rate(13kbps)}}$.

4.2 Traffic Models

There are four different traffic models that will be discussed in this thesis. Users' instantaneous rate will change according to these four models:

- Model 1: Uniform distribution
- Model 2: Two-state Markov Chain
- Model 3: Two-state Markov Chain and during the ON period, the instantaneous rate change uniformly
- Model 4: Gaussian distribution

To make a fair comparison of the performance of these four traffic models, same average data rate and same variance of average data rate should be assigned to each model. But, in model 2, because the variance is much bigger than other three models, to make a reasonable assumption, the variance in this model is set different from the rest. In each of these four traffic models, fixed traffic allocation and dynamic traffic allocation are compared. In fixed traffic allocation, the resource is allocated according to user's average rate and system's QoS requirements. After the slots are assigned to users, users will follow the arrangement until the end of the simulation. In dynamic traffic allocation, the user's acceptance procedure is dependent on user's average rate and system's QoS requirement. The reallocation procedure is dependent on user's instantaneous rate and carrier's capacity .This means the reallocation procedure happens every four windows as the instantaneous rate changes.

Model 1: Uniform

User's instantaneous rate uniformly changes between minimum rate and maximum rate.

Minimum rate: $R\min$

Maximum rate: $R\max$

In uniform distribution:

User's average rate: $\overline{R} = \frac{R_{\max} + R_{\min}}{2}$

User's variance of average rate: $\sigma^2 = \frac{(R_{\max} - R_{\min})^2}{12}$

User's instantaneous rate: $R_i = R_{\min} + (R_{\max} - R_{\min})x$ (x is a random number from 0 to 1)

Four classes of data rate are defined in this model:

$\overline{R1}$ = 8,000bits/s, $\sigma = 2000$, R1min=4,536 bits/s, R1max=11,464 bits/s

$\overline{R2}$ = 16,000bits/s, $\sigma = 2000$, R2min=12,536 bits/s, R2max=19,464 bits/s

$\overline{R3}$ = 32,000bits/s, $\sigma = 2000$, R3min=28,536 bits/s, R3max=35,464 bits/s

$\overline{R4}$ = 64,000bits/s, $\sigma = 2000$, R4min=60,536 bits/s, R4max=67,464 bits/s

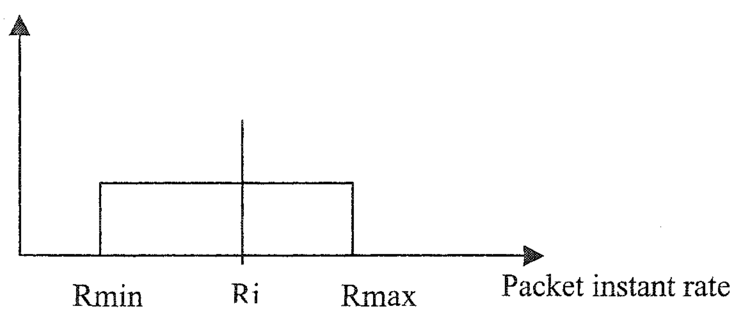


Figure 4.1 Uniform distribution

Model 2: Two-state Markov Chain

Markov chain is named after A. A. Markov who at the turn of the century studied poetry and other texts as stochastic sequences of character (symbols, letters, syllables, and words). The probabilities of a Markov chain are usually entered into a transition matrix indicating which state or symbol follows which other state or symbol [21].

User's instantaneous rate change follows Discrete Time Two-State Markov Chain that is used to model the ON-OFF source traffic as shown in Figure 4.2. In this model, source alternates between active emission periods (ON period) and idle periods (OFF period). During the ON period information is generated at a constant peak access rate ($R_{\max}$), while in the OFF period no information is emitted.

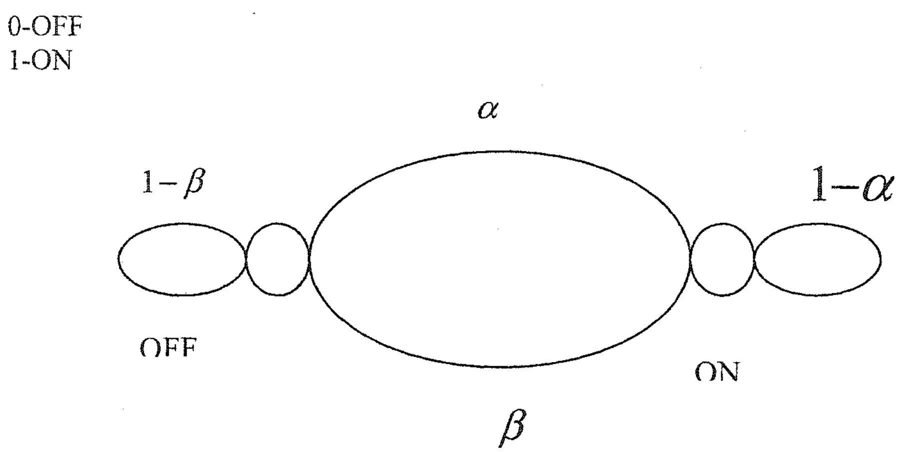


Figure 4.2 Two-state Markov model

When the call from a user is accepted by the system, the initial state of user is 1(ON). To get the status in next window, the simulation program calls a uniform distribution function: U (0,1)-Z. If $Z \leq \alpha$, user will go to state 0(OFF) in the next window. On the other hand, if $Z \geq \alpha$, user will stay in state 1(ON) in the next window as shown in Figure 4.3(a). When a user is in state 0 (OFF), the program calls another uniform distribution function: U (0,1)-g. If $g \le \beta$, user will go to state 1(ON) in the next window. If $g > \beta$, user will stay in state 0(OFF) in next window as shown in Figure 4.3(b).

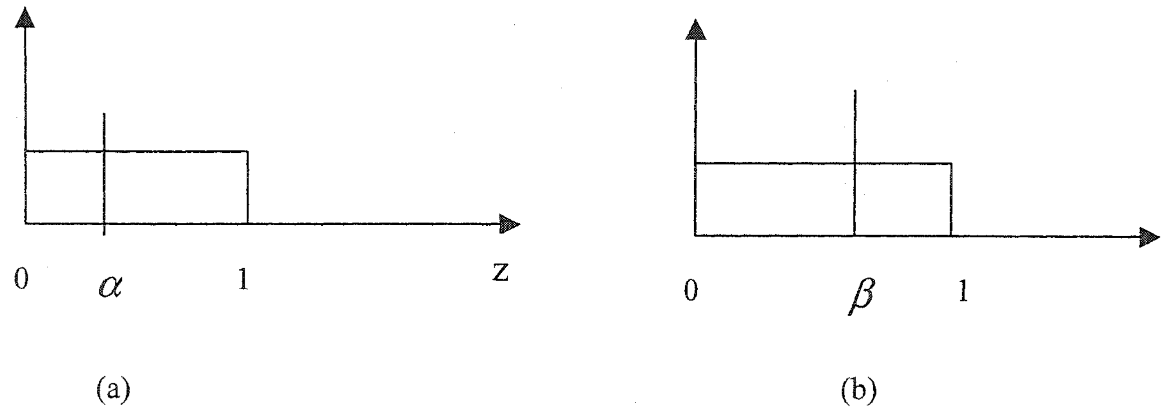


Figure 4.3 (a) α generator
(b) β generator

Data packets are periodically generated at fixed τ intervals. τ^{-1} represents the peak access rate (packets/sec). The ON and OFF periods are defined by T_{on} and T_{off}.

The probability of T_{on}: $\bar{\varepsilon} = P = \dfrac{T_{on}}{T_{on} + T_{off}} = \dfrac{\beta}{\alpha + \beta}$

The average data rate: $\bar{R} = \bar{\varepsilon} \times R_{\max} = \dfrac{\beta}{\alpha+\beta} \times R_{\max}$

The variance of data rate:

$$\begin{aligned}
\sigma^2 &= E(x^2) - E^2(x) \\
&= P \times R_{\max}{}^2 - \left(\frac{\beta}{\alpha+\beta}\right)^2 \times R_{\max}{}^2 \\
&= \frac{\beta}{(\alpha+\beta)} R_{\max}{}^2 - \left(\frac{\beta}{\alpha+\beta}\right)^2 R_{\max}{}^2 \\
&= R_{\max}{}^2 \frac{\beta}{\alpha+\beta}\left[1 - \left(\frac{\beta}{\alpha+\beta}\right)\right] \\
&= R_{\max}{}^2 \frac{\alpha\beta}{(\alpha+\beta)^2}
\end{aligned}$$

In this simulation, we take $\alpha = 0.4, \beta = 0.6$. Assume the same mean of data rate as the other three models. To make all the different data rate users have the same α and β, the variance has to be different. So, four classes of data rate are defined in this model:

$\overline{R1}$ = 8,000bits/s, $\sigma = 6500$, R1max=13,333 bits/s

$\overline{R2}$ = 16,000bits/s, $\sigma = 13000$, R2max=26,667 bits/s

$\overline{R3}$ = 32,000bits/s, $\sigma = 26000$, R3max=53,333 bits/s

$\overline{R4}$ = 64,000bits/s, $\sigma = 52000$, R4max=106,667 bits/s

Model 3: Two-state Markov Chain and during the ON period, the instantaneous rate change uniformly.

User's instantaneous rate change follows Discrete Time Two-State Markov Chain and during active state user's generation rate is uniformly distributed between fixed rate limits -- Rmin and Rmax.

The average data rate: $\overline{R} = \frac{\beta}{\alpha+\beta} \times \frac{R\max + R\min}{2}$

The variance of average data rate:

$$\sigma^2 = \frac{\beta}{\alpha+\beta} E(x^2) - (\frac{\beta}{\alpha+\beta})^2 E^2(x)$$
$$= \frac{\beta}{\alpha+\beta} \times [\frac{(R\max - R\min)^2}{12} + \frac{(R\max + R\min)^2}{4}] - \frac{\beta^2}{(\alpha+\beta)^2} \times \frac{(R\max + R\min)^2}{4}$$

In this simulation, we take $\alpha = 0.4, \beta = 0.6$. To make a fair comparison, the mean and variance of data rate of four different user groups in this model are set the same as the Uniform and Gaussian models. The parameters of four classes data rates are set as below:

$\overline{R1}$ = 8,000bits/s, $\sigma = 2000$, R1max=27,391bits/s

$\overline{R2}$ = 16,000bits/s, $\sigma = 2000$, R2max=53,702bits/s

$\overline{R3}$ = 32,000bits/s, $\sigma = 2000$, R3max=106,851 bits/s

$\overline{R4}$ = 64,000bits/s, $\sigma = 2000$, R4max=213,425bits/s

Model 4: Gaussian

User's instantaneous rate change follows Gaussian model. The distribution function of Gaussian is: $P(x) = \frac{1}{\sigma\sqrt{2\pi}} e^{-\frac{(x-\mu)^2}{2\delta^2}}$

μ : Mean

δ : The variance of average data rate

The probability that a point is in the range of -3δ to 3δ is 99.72%.

The probability that a point is in the range of $-\delta$ to δ is 68.26%.

Data rate of -3δ was taken as Rmin, and data rate of 3δ was taken as Rmax.

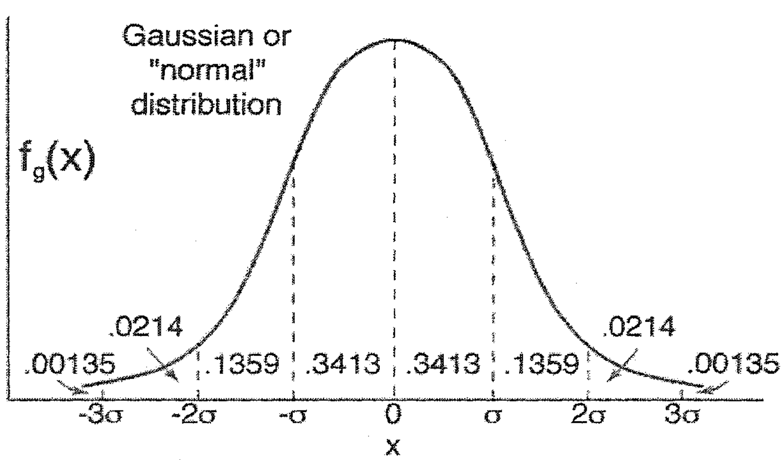


Figure 4.4 Gaussian distribution Function

Four classes of data rate are defined in this model:

$\overline{R1}$ = 8,000bits/s, $\sigma = 2000$, R1min=2,000 bits/s, R1max=14,000 bits/s

$\overline{R2}$ = 16,000bits/s, $\sigma = 2000$, R2min=10,000 bits/s, R2max=22,000 bits/s

$\overline{R3}$ = 32,000bits/s, $\sigma = 2000$, R3min=26000 bits/s, R3max=38,000 bits/s

$\overline{R4}$ = 64,000bits/s, $\sigma = 2000$, R4min=58,000 bits/s, R4max=70,000 bits/s

4.3 Simulation Descriptions

The message flows that are handled in this simulation are continuous data flows. Their instantaneous data rate depends on their traffic models, the average data rate and the variance of average data rate. We simulated data flows in the RLC/MAC layer and uplinks only.

In this simulation, call admission control procedure and resource allocation schemes are proposed to allocate the resources so that the utilization of resources is maximized and more and more users can be accommodated.

To see the performance improvement made by the dynamic resource allocation, the results of fixed resource allocation and dynamic resource allocation are compared. In fixed scheme, the fixed resource is allocated to users to meet the users' QoS requirement (embodied by the Capacity Assignment Parameter in this simulation). In dynamic scheme, the resource is initially allocated to users to meet the users' QoS requirement, then, together with the instantaneous data rate changing, all the resource in one PDCH is fairly allocated to users who are currently in this PDCH. Because the instantaneous data rate can be higher or lower than the average date rate, the reallocation can bring performance improvement to the system. When a new Packet Channel Request comes in and there is a lack of resources, the extra resource which users "steal" from the resource pool are released to provide space for the new request.

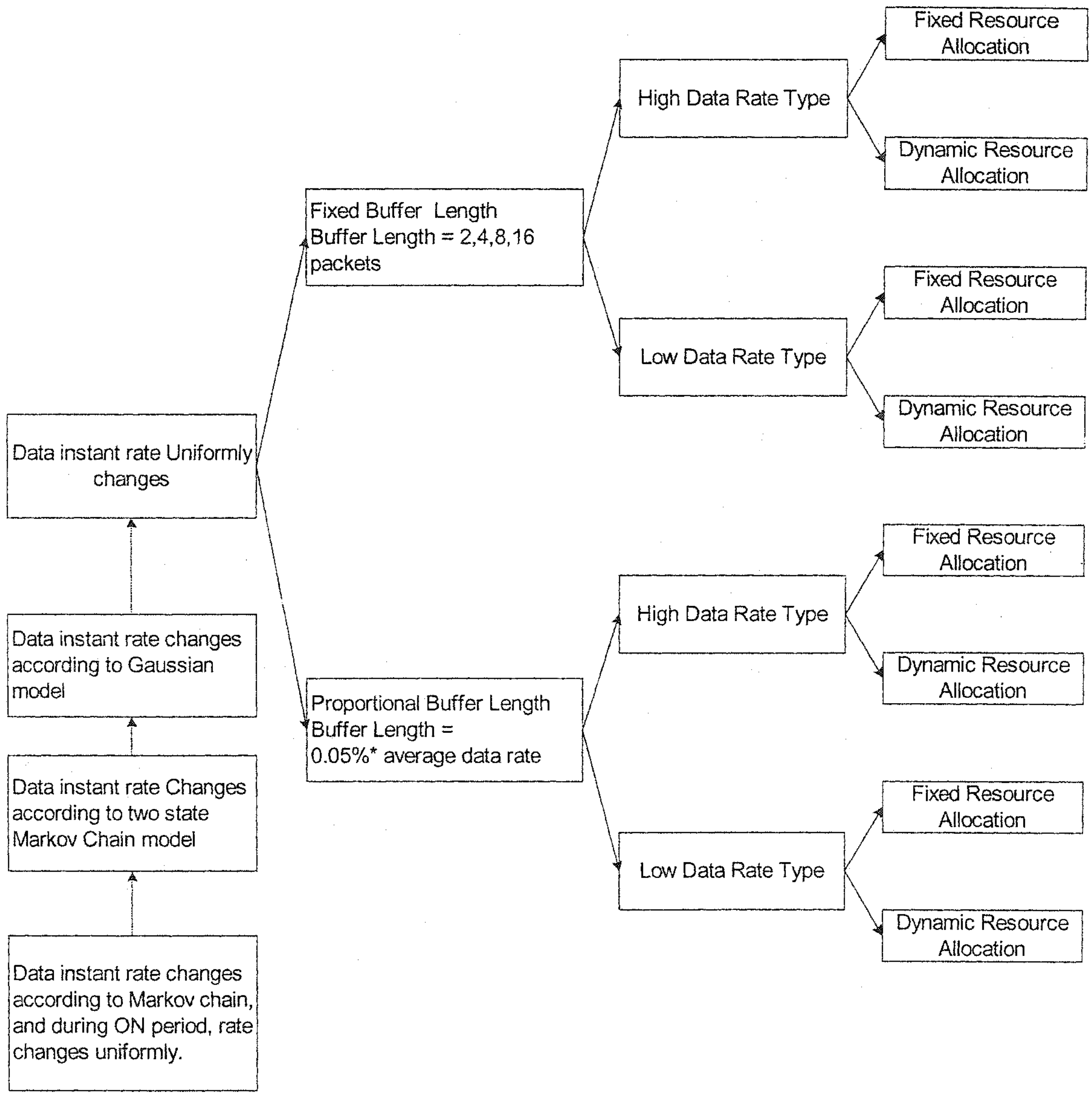


Figure 4.5 Different Conditions in simulations

To see the difference brought by different buffer length, traffic models, buffer type, and the composition of users, the following comparison are also made:

- Among users composed of high data rate type and users composed of low data rate type;

- Among four different traffic models

To get all these result, each combination of different situation is simulated. The various combinations are summarized in Figure 4.5.

In each run of the simulations, the value of Capacity Assignment Parameter (θ) is set as 0.25, 0.50, 0.75 and 1.00 to compare the effect that different value of Capacity Assignment Parameter can bring to the system. In each specific number of Capacity Assignment Parameter, the value of traffic load ρ is set from 0.0 to 1.5. The reason why the traffic load is taken bigger than 1 is to see the performance result of overloaded system.

4.3.1 Fixed Resource Allocation Scheme

In fixed resource allocation, fixed number of channels is assigned to users according to the user's Average Data Rate and Capacity Assignment Parameter (θ). During the calling period, user's Instantaneous Data Rate changes every four windows according to different traffic models. But the assignment of channels is fixed without any change until the end of the call.

The block diagram of this scheme is shown in Figure 4.6. The procedure of initiate users, call admission and resource allocation to new users are emphasized below.

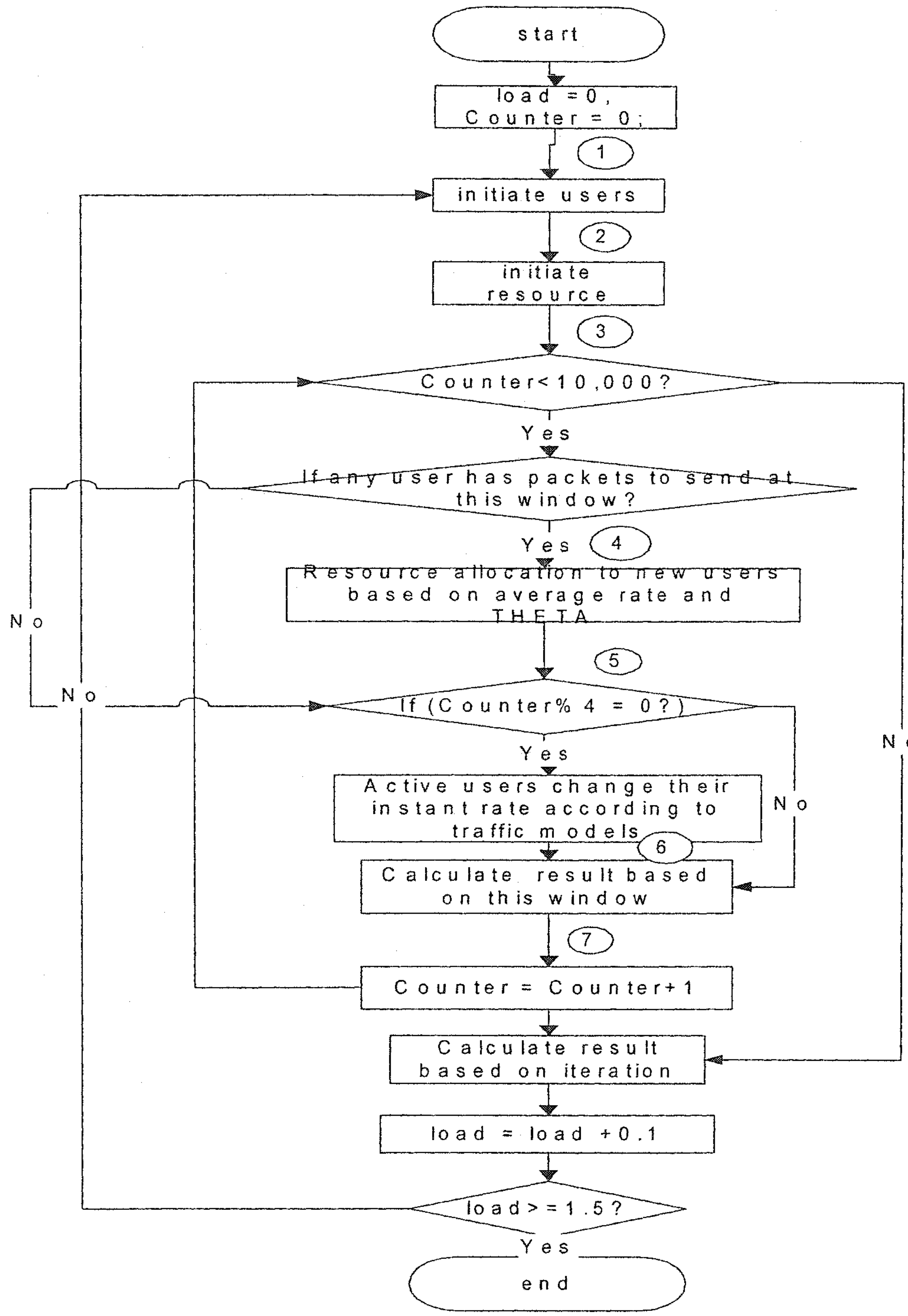


Figure 4.6 Main Block Diagram of Fixed Allocation Scheme

4.3.1.1 Users Initiate Procedure (Figure 4.7)

Based on the traffic load ρ , and the traffic type (High Rate type or Low Rate type), the total number of users is first calculated at the beginning of the simulation. The generation of each user takes place randomly in the first 100 simulation windows. The parameter of each user is initiated.

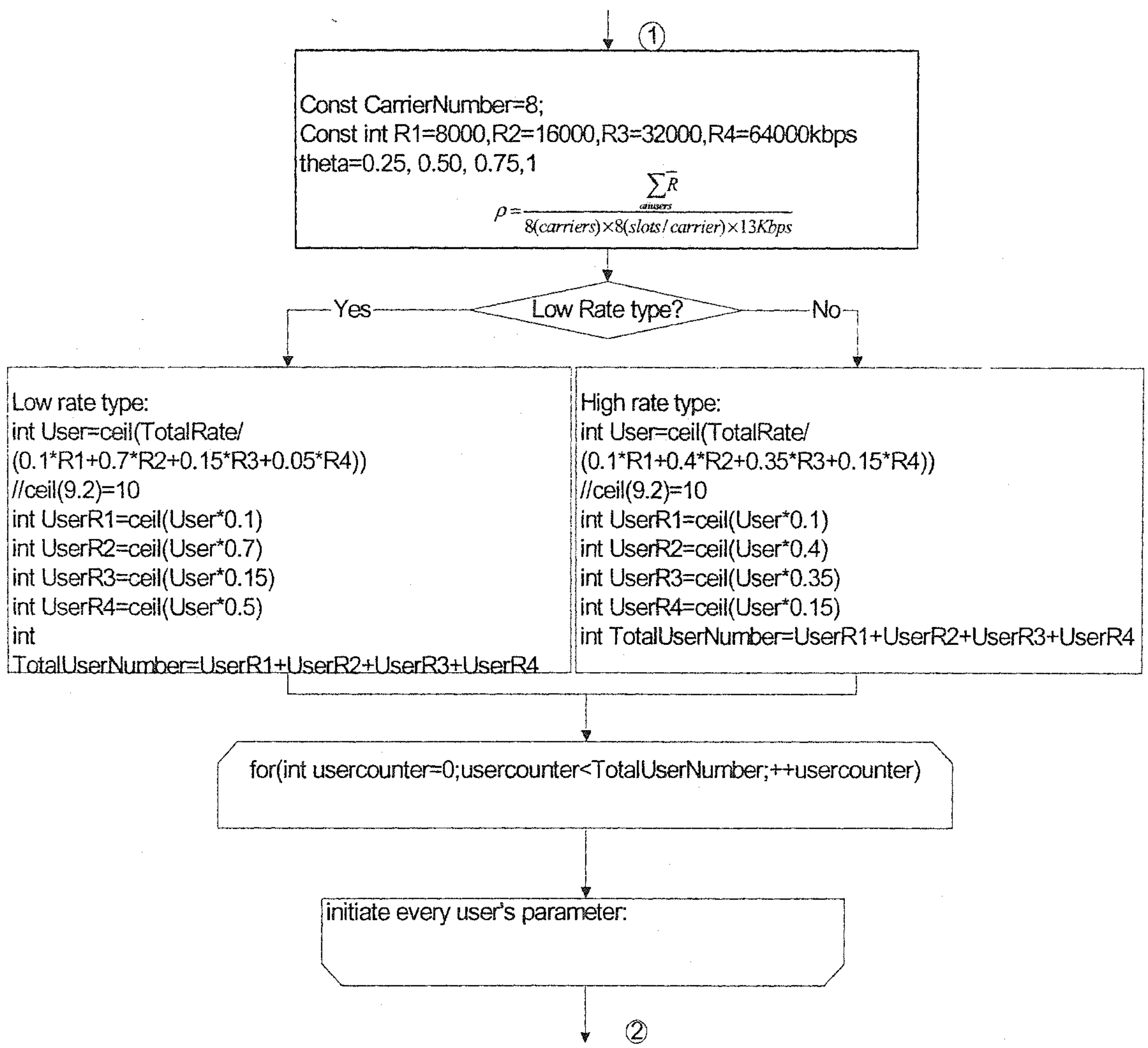


Figure 4.7 Block Diagram of Initiate users

4.3.1.2 Call Admission Control and Channel allocation Procedure (Figure 4.8)

- ***Admission control*** is the functionality in charge of estimating whether the network has enough available resources to allocate a new connection. To decide accepting or rejecting a new connection is based on two parameters in this model.

    1. The needed capacity of the new connection. It is defined as the capacity the new connection needs in order to fulfil its bit rate requirement. The needed capacity in this model is: The Capacity Assignment Parameter * the Average Data Rate
    2. The available capacities in one channel whose available capacity is the biggest in this cell.

  The details of the admission control procedure is:

  Users are first in first served to become active users. When all the resources are used, Call Blocking occurs.

In each window, when the system receives the admission request from a user, the Carrier Load of each carrier is compared and the carrier with the biggest free capacity is selected. Only when there are sufficient resources are available to fulfill the QoS requirement of the user, this call will be accepted.

- ***Channel Allocation*** is the functionality to determine which are the channels that have to be reserved for this new connection. The details of this procedure is:

  The average data rate of this user is multiplied with the Capacity Assignment Parameter to get the System Initially Assigned Data Rate. This System Initially Assigned Data Rate is divided by the GSM data rate (13kbps) to get the User's Load and the needed slots number in one window.

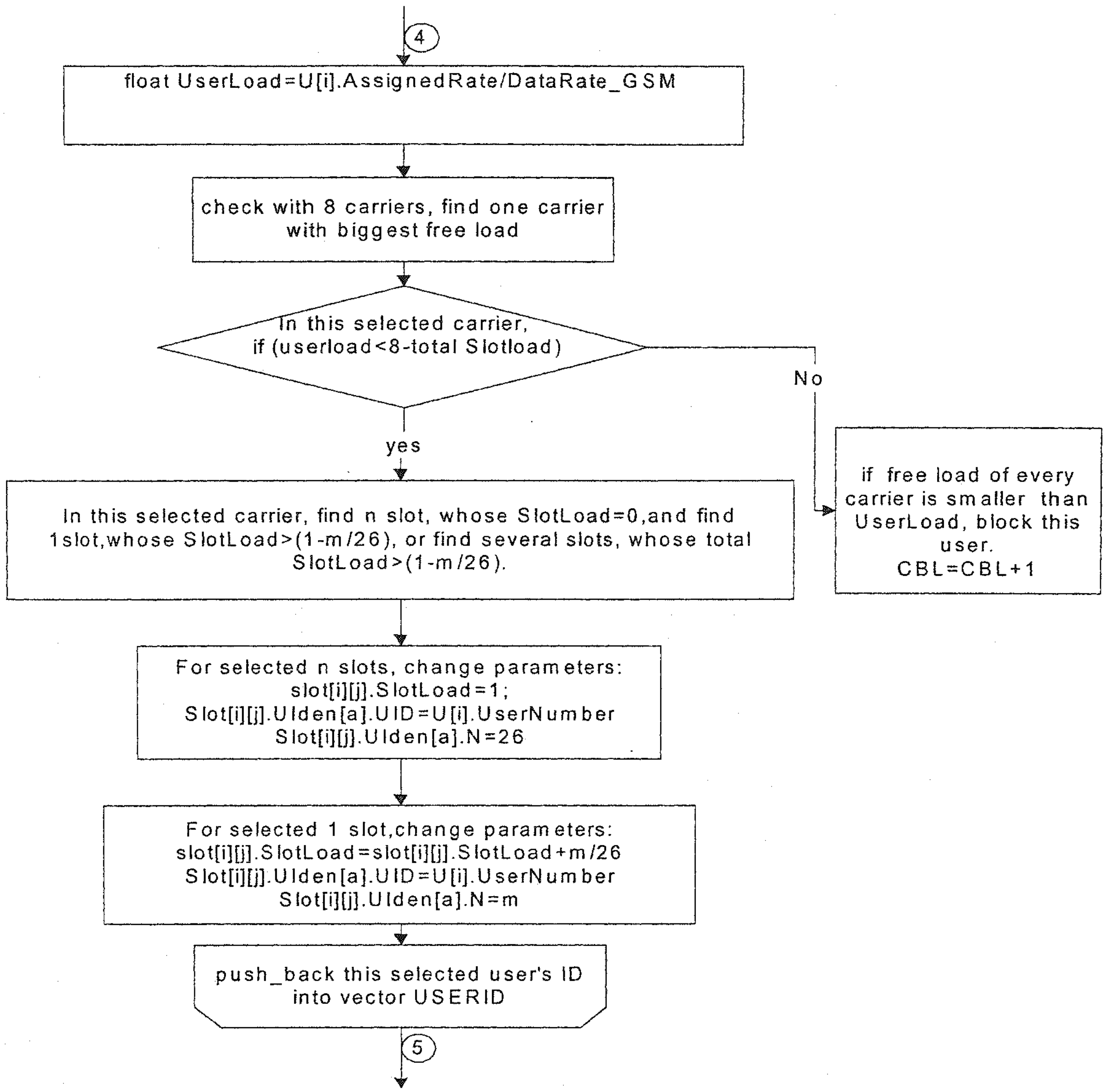


Figure 4.8 Block Diagram of Call admission and Resource Allocation to new users

The needed slots number can be expressed as: $N \times 26 + M$. Which means that in each window (26 frames) N slots needed in each frame and M extra slots needed in the total 26 frames. If the User's Load is smaller than (8- Carrier Load) which means the current carrier can tolerate the new user, then this user will transmit data on the assigned carrier and become active user. If the User's Load is bigger than (8- Carrier Load), call blocking happens on this user. If the User's Load of next user is small enough for the system to tolerant, then the system will still accept it.

4.3.1.3 Fixed Resource Allocation Scheme

In fixed traffic allocation, resource is allocated to users based on the Average Data Rate and the Capacity Assignment Parameter (θ). And this allocation keeps the same until the end of the call. After the user begins to transmit data, the instantaneous rate of user changes according to the traffic model every 4 windows.

4.3.2 Dynamic Resource Allocation Scheme

In dynamic resource allocation, Users Initiate Procedure, Call admission and Resource Allocation procedure are the same as fixed traffic allocation. The block diagram is the same as in Figure 4.9. But now a dynamic allocation procedure (Figure 4.10) is taken place.

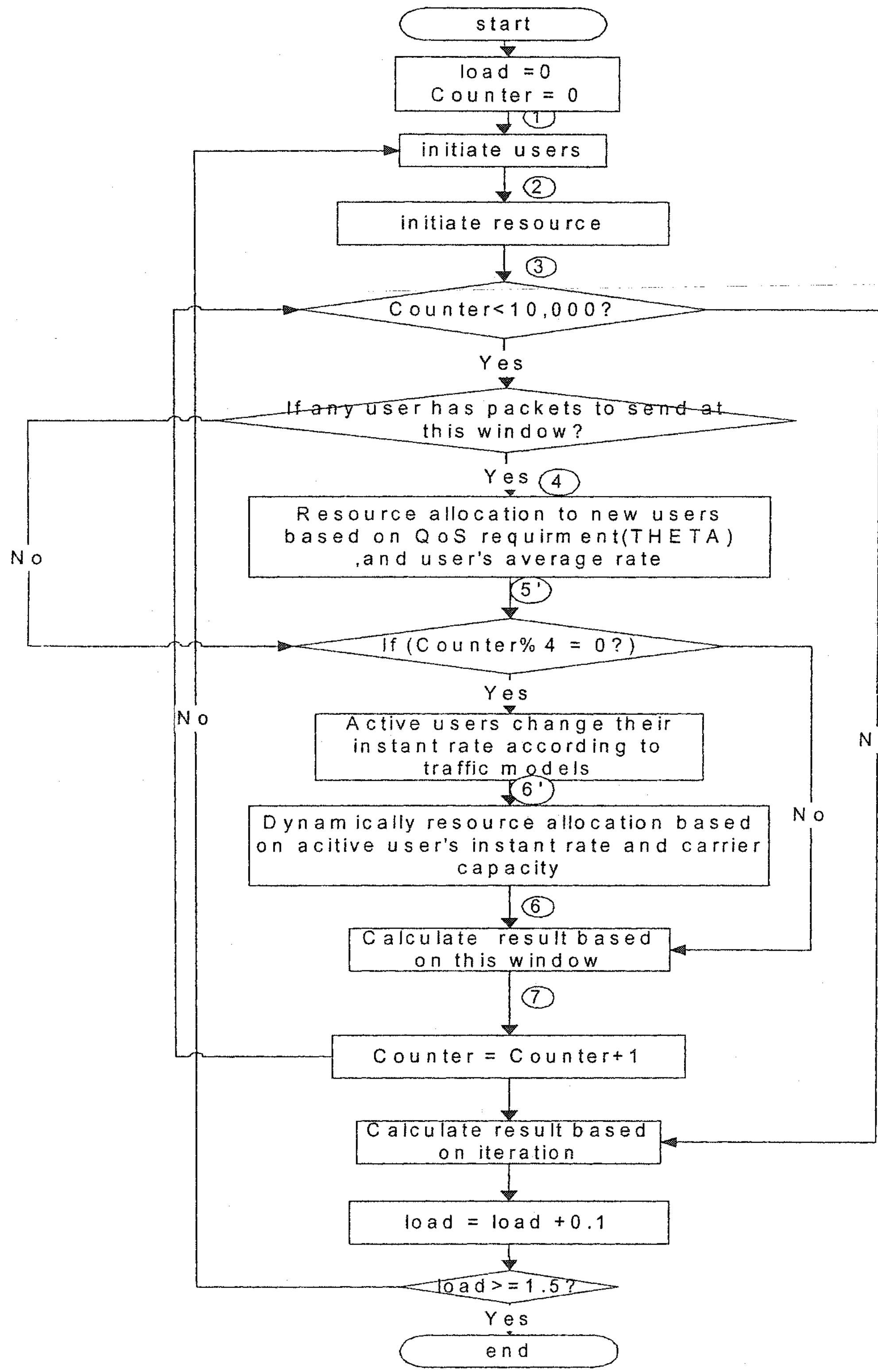


Figure 4.9 Main Block Diagram of Dynamic Allocation Scheme.

4.3.2.1 Dynamic Resource Allocation Scheme

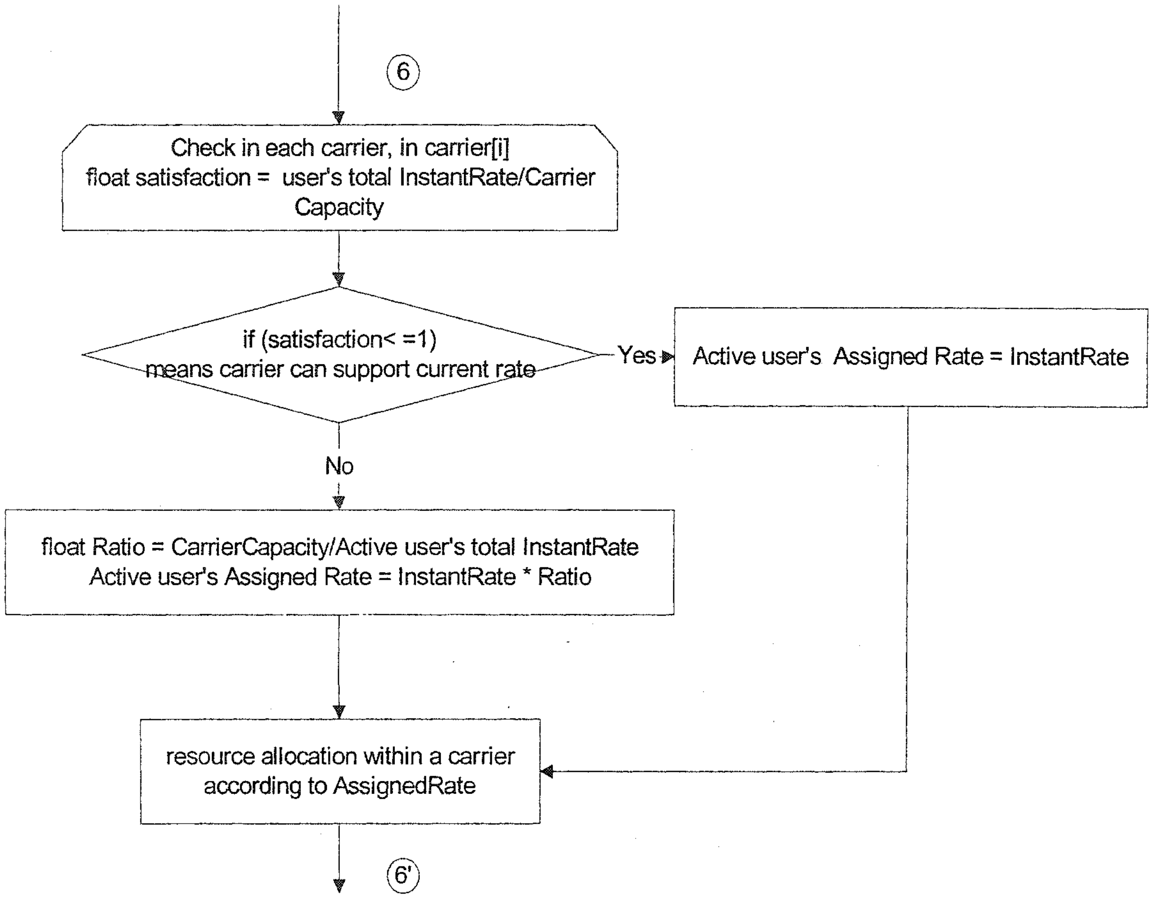


Figure 4.10 Block Diagram of Dynamic Resource Allocation

Resource is initially allocated to users based on $\overline{R} \times \theta$, the same procedure as fixed allocation, but after that, with the users instantaneous rate changes according to the traffic model every 4 windows, the reallocation happens. In reallocation, users will keep in the same carrier, which was initially assigned to them. The number of slots assigned to each user dependend on the instantaneous rate R_i and carrier capacity. If system carrier 's capacity cannot satisfy each user's instantaneous rate, then only satisfy instantaneous rate multiply by a ratio that is defined in figure 4.10. With the use of this ratio, the system capacity can be maximally utilized and at the same time, the user's assigned capacity will not exceed the system capacity limit. In this way, the active users will not drop during calling, and efficiently, fairly share the resource.

4.4 Simulation Results

The simulation results under different situations is discussed and compared in this section. The comparison is made according to the sequence of the output parameters. And the comparison of the Fixed Resource Allocation and Dynamic Resource Allocation is made at the same time.

4.4.1 Packet Efficiency

The result shown in figures is obtained from simulation of model 4. In model 2, a similar result is obtained from simulation. But, because the variance of data rate is bigger than the other three models, the packet efficiency reaches 1 slower than other models.

1. **The effect of the Capacity Assignment Parameter**

Figure 4.11 and Figure 4.12 show the performance of Packet Efficiency under different Capacity Assignment Parameters in the Gaussian model case. Both figures show that the Packet Efficiency increases with the traffic load. Figure 4.11 is the result of Fixed Resource Allocation scheme. It shows that the value of the Capacity Assignment Parameter does not affect the efficiency performance. The efficiency goes to one after the traffic load is 100%. It shows that in fixed allocation, when the traffic load is less than 100%, resources are wasted instead of being assigned to the users who demand more resources.

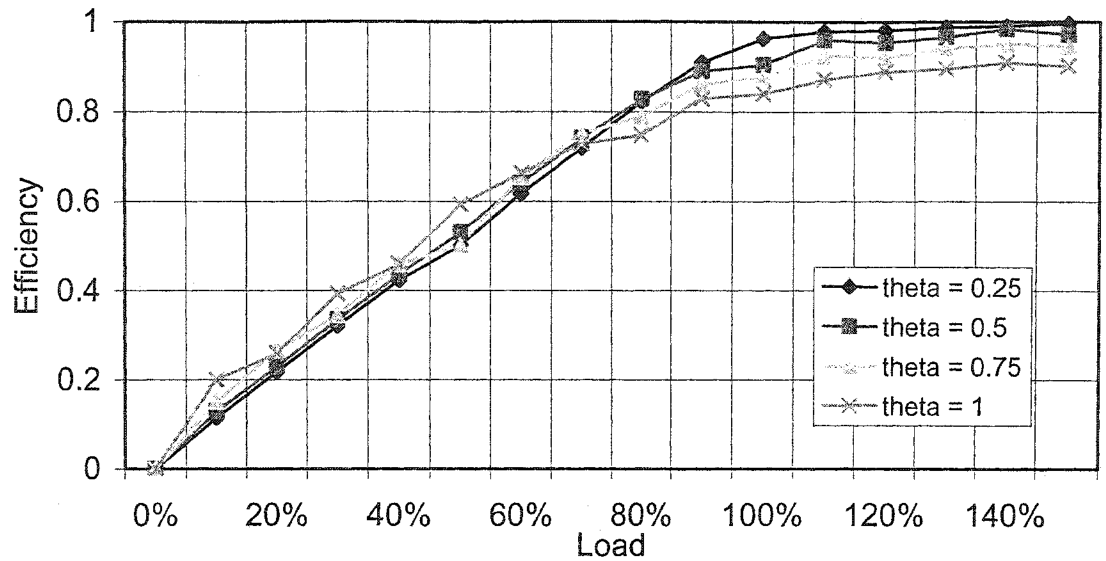


Figure 4.11 Packet Efficiency in Fixed Resource Allocation
User's instantaneous rate changes according to Gaussian model
Buffer Length = 0.05%*average rate, High rate type

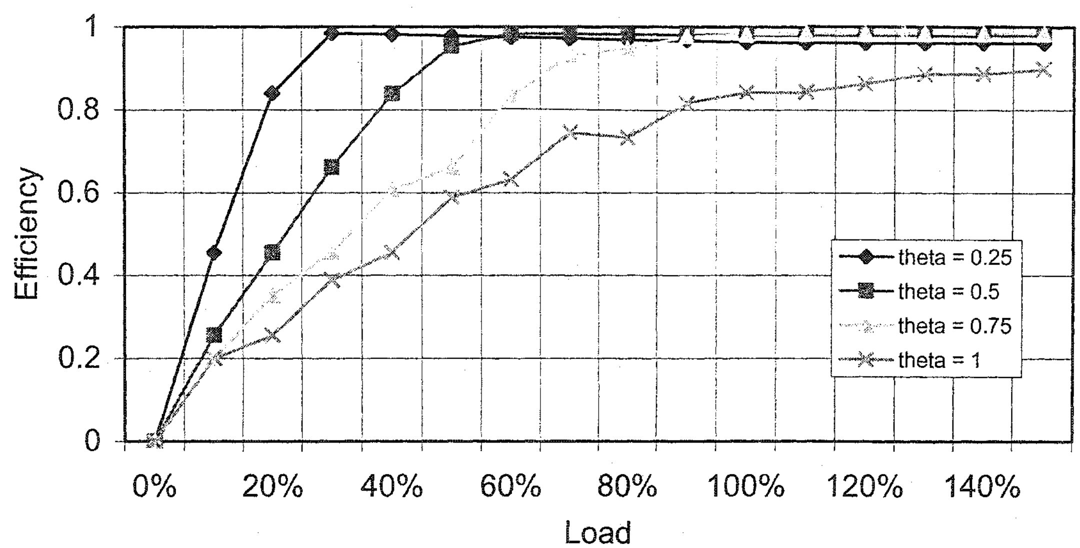


Figure 4.12 Packet Efficiency in Dynamic Resource Allocation
User's instantaneous rate changes according to Gaussian model
Buffer Length = 0.05%*average rate, high rate type

Figure 4.12 is the result of Dynamic Resource Allocation scheme. It shows that the value of the Capacity Assignment Parameter does affect the performance of efficiency. The Capacity Assignment Parameter is small means absorbing more users but with poor

QoS satisfaction. When the traffic load is smaller than 100%, the resource is dynamically assigned to users based on their data rate requirement. So, the smaller the Capacity Assignment Parameter, the better the performance of Packet Efficiency. And the efficiency in dynamic allocation scheme reaches to 1 much more early than fixed allocation scheme.

The same result also applies to Model 1, 3.

2. The effect of Buffer Length

Figure 4.13 and Figure 4.14 show the performance of Packet Efficiency under different Buffer Length but under the same Capacity Assignment Parameter (0.25 in this case) in Gaussian model. The performance of Packet Efficiency in Dynamic Resource Allocation scheme is much better than in Fixed Resource Allocation scheme. But the Buffer Length does not affect the performance of the Packet Efficiency. This is because the Capacity Assignment Parameter is more effective than Buffer Length.

The same result also applies to Model 1, 3.

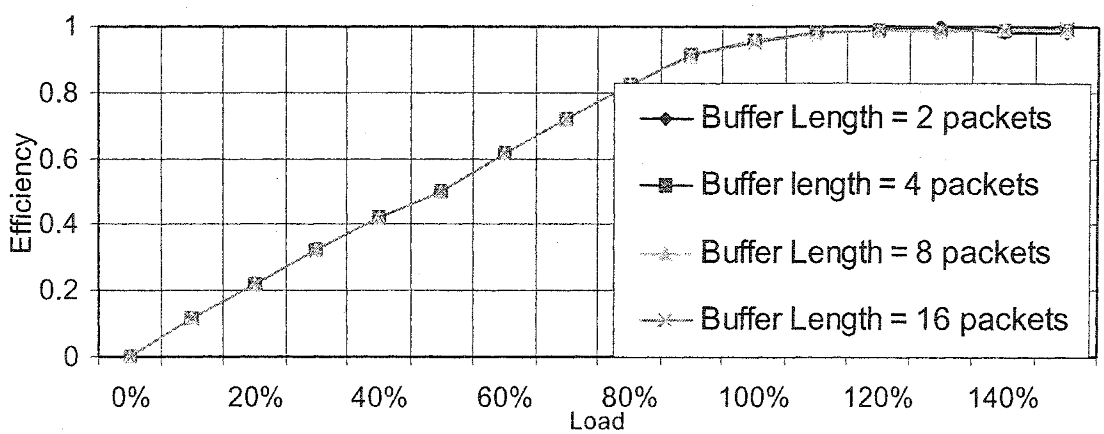


Figure 4.13 Packet Efficiency in Fixed Resource Allocation
User's instantaneous rate changes according to Gaussian model
Buffer Length =2,4,8,16 Packets, High rate type, Theta = 0.25

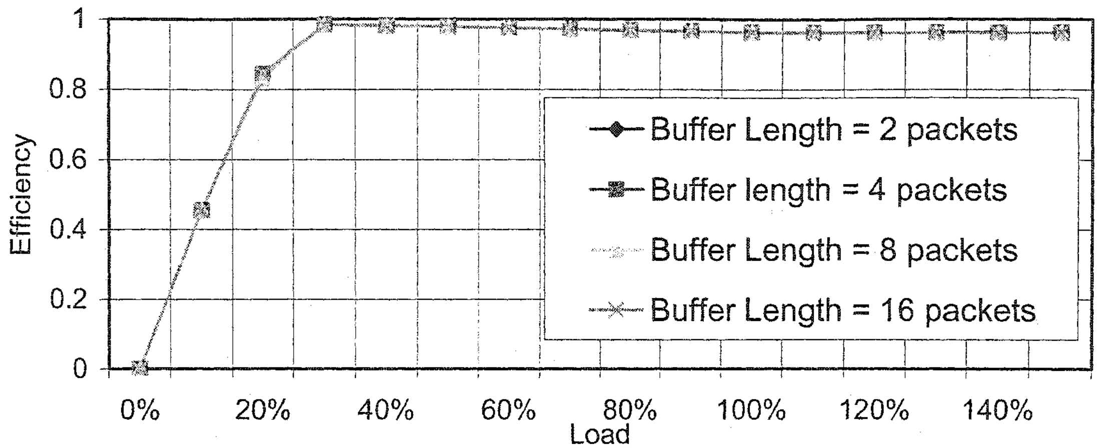


Figure 4.14 Packet Efficiency in Dynamic Resource Allocation
User's instantaneous rate change according to Gaussian model
Buffer Length =2,4,8,16 Packets, High rate type, Theta = 0.25

3. The effect of Data Rate Type (high or low)

Figure 4.15 and Figure 4.16 show the performance of Packet Efficiency under different Data Rate Type but for the same Capacity Assignment Parameter (0.25 in this case) in the Gaussian model case. The performance of Packet Efficiency in Dynamic Resource Allocation scheme is much better than in Fixed Resource Allocation scheme. But the Date Rate type does not affect the performance. Which means Packet Efficiency is not sensitive to the combination of data rate type of users. The same result also applies to model 1, 3.

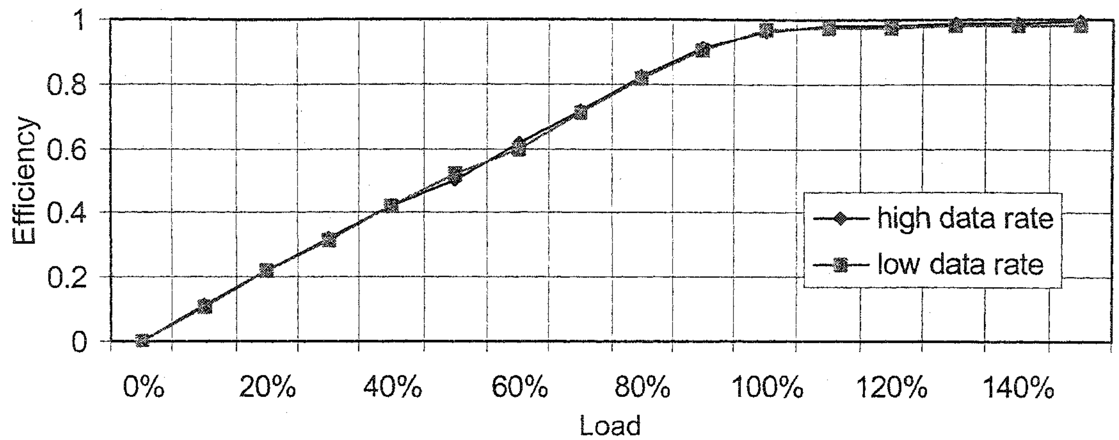


Figure 4.15 Packet Efficiency in Fixed Resource Allocation
User's instantaneous rate change according to Gaussian model
Buffer length = 0.05%*average rate, theta = 0.25

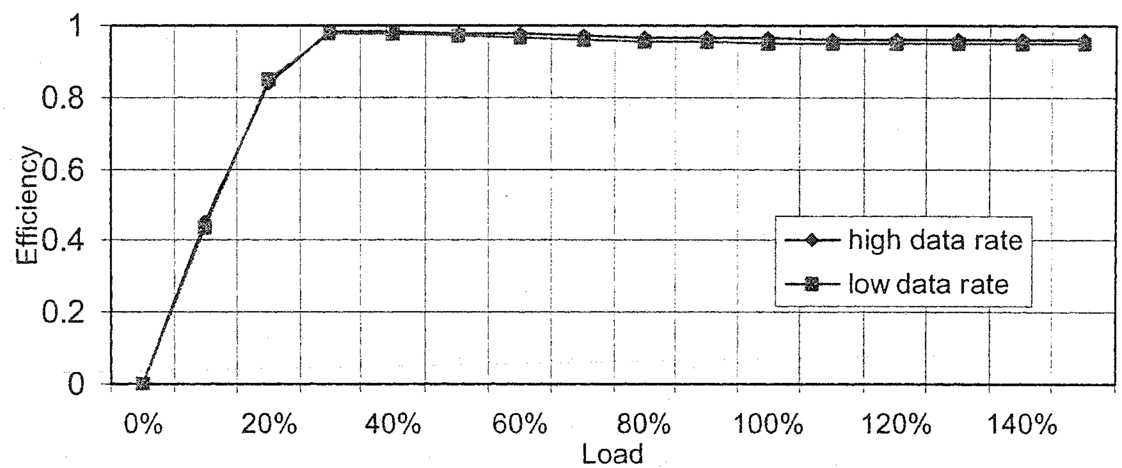


Figure 4.16 Packet Efficiency in Dynamic Resource Allocation
User's instantaneous rate change according to Gaussian model
Buffer length = 0.05%*average rate, theta = 0.25

4.4.2 Average Buffer Overflow

The result shown here applies to simulation of model 1. And it was found that the same result apply for model 3 and model 4 too. In model 2, the similar result is obtained from simulation. But, because the variance of data rate is bigger than other three models, the buffer overflow is higher than other models.

1. **The effect of the Capacity Assignment Parameter**

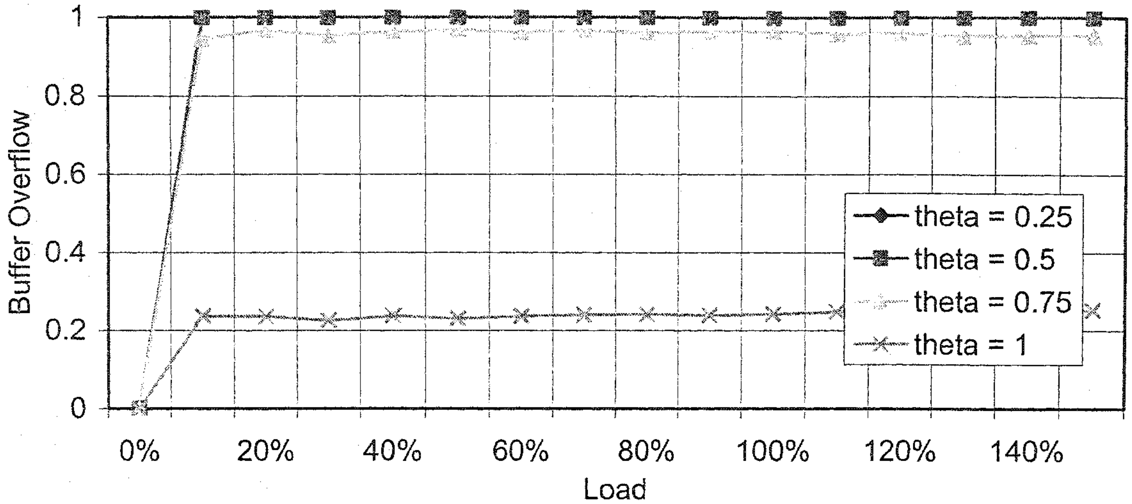


Figure 4.17 Average Buffer Overflow in Fixed Resource Allocation
User's instantaneous rate changes according to Uniform model
Buffer Length = 0.05%*average rate, high rate type

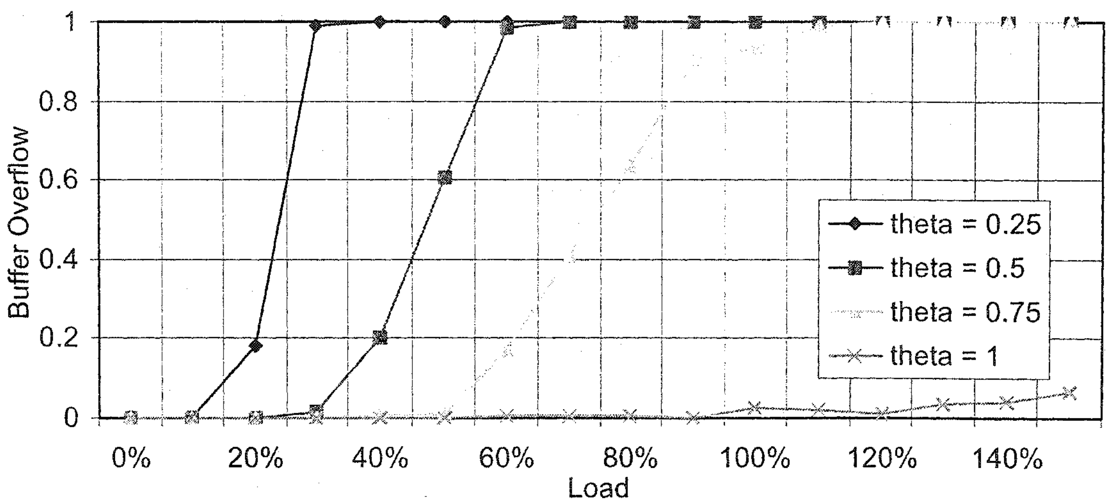


Figure 4.18 Average Buffer Overflow in Dynamic Resource Allocation
User's instantaneous rate changes according to Uniform model
Buffer Length = 0.05%*average rate, high rate type

Figure 4.17 and Figure 4.18 show the performance of Average Buffer Overflow under different Capacity Assignment parameters in the Uniform model case. The performance of Average Buffer Overflow in Dynamic Resource Allocation scheme is much better than the Fixed Resource Allocation scheme. Figure 4.17 is the result of Fixed Resource Allocation scheme. It shows that when the value of the Capacity Assignment Parameter is less than 1, the performance of the Average Buffer Overflow is almost the same, almost reaches to 1 as soon as there are traffic load. Figure 4.18 is the result of Dynamic Resource Allocation scheme. It shows that the performance of the Average Buffer Overflow is largely improved in this case. And it improves with the increasing of the Capacity Assignment Parameter. This phenomenon proves that the bigger the Capacity Assignment Parameter the better the QoS satisfaction. The same result also applies to model 3,4.

2. The effect of Buffer Length

Figure 4.19 and Figure 4.20 show the performance of Average Buffer Overflow under different Buffer Length but under the same Capacity Assignment Parameter (1.0 in this case) in the Uniform model. Figure 4.19 shows the performance of Average Buffer Overflow in Fixed Allocation Schemes improves with the increasing of Buffer Length. Figure 4.20 shows that the performance of Average Buffer Overflow is dramatically improved in Dynamic Resource Allocation scheme than in Fixed Resource Allocation scheme, but the Buffer Length does not affect the performance of Average Buffer Overflow anymore. The reason why the curves oscillate is that the overflow value is very low, random data rate's change can totally change the shape of the figure.

The same result also applies to model 3,4.

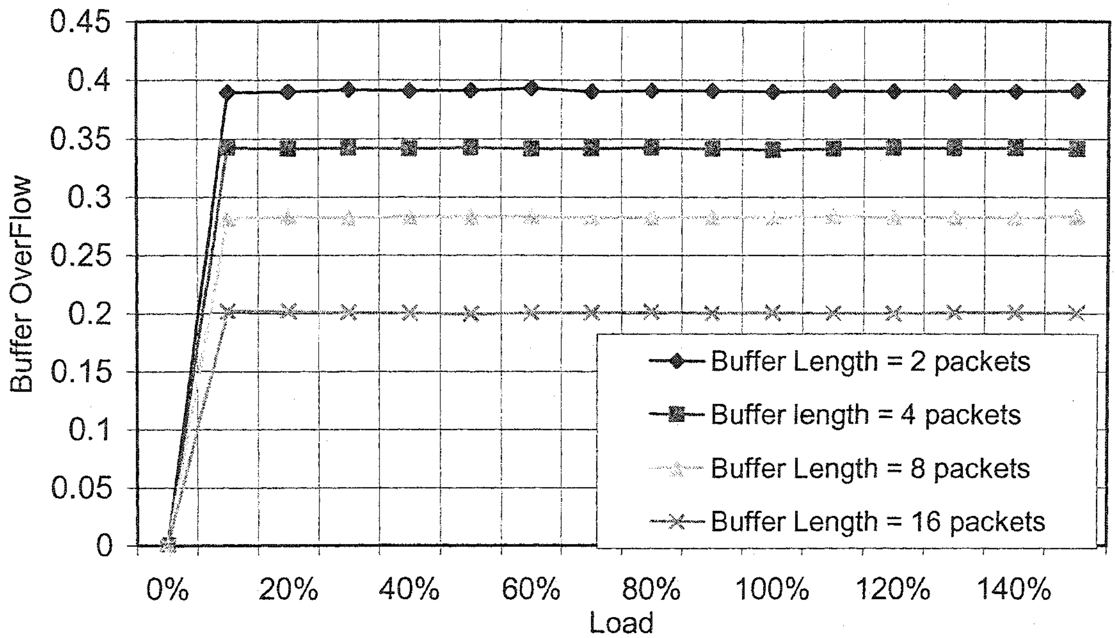


Figure 4.19 Average Buffer Overflow in Fixed Resource Allocation
User's instantaneous rate changes according to Uniform model
Buffer Length =2,4,8,16 packets, high rate type, Theta = 1

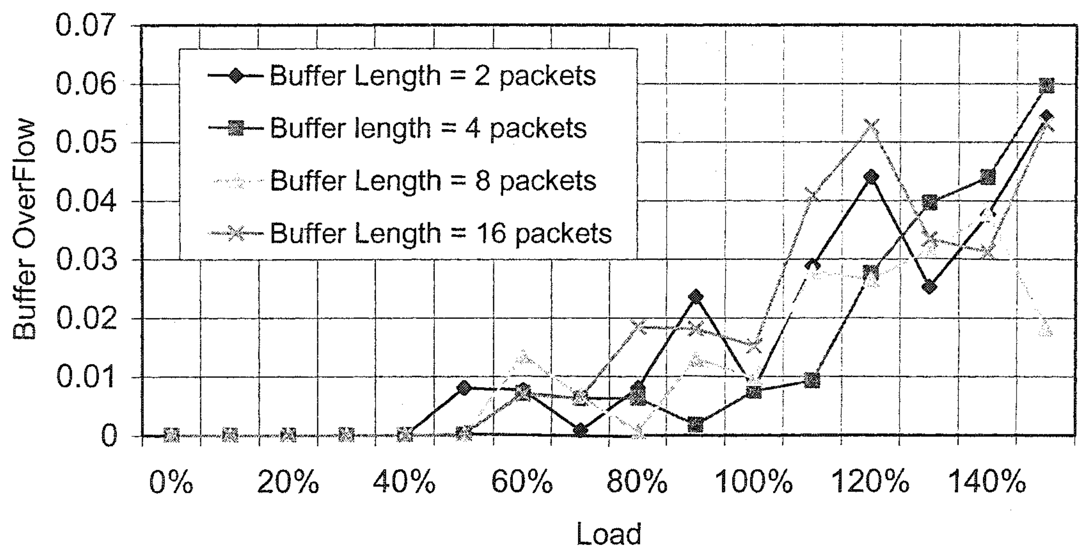


Figure 4.20 Average Buffer Overflow in Dynamic Resource Allocation
User's instantaneous rate changes according to Uniform model
Buffer Length =2,4,8,16 packets, high rate type, Theta = 1

3. The effect of Data Rate Type (high or low)

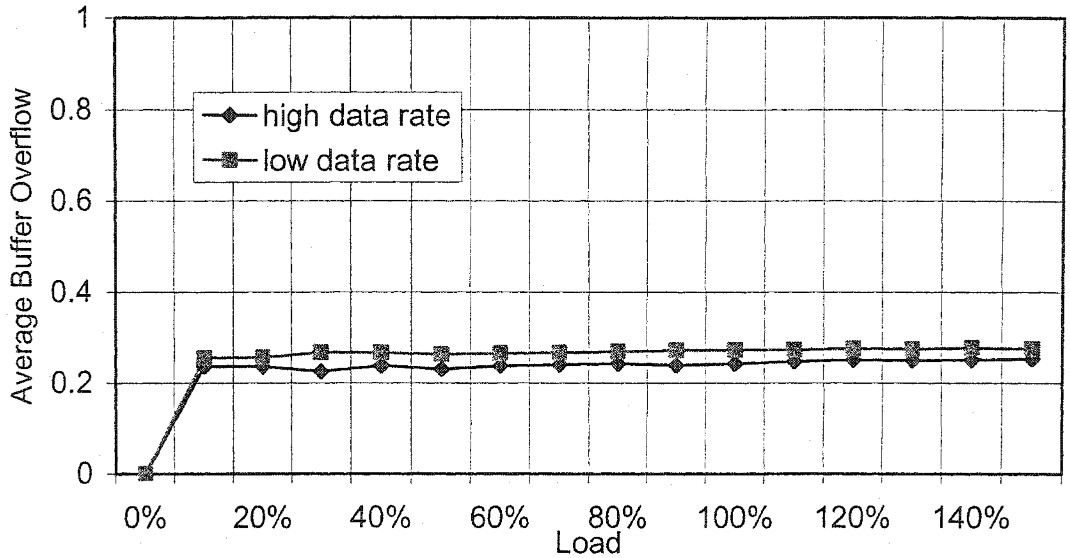


Figure 4.21 Average Buffer Overflow in Fixed Resource Allocation
User's instantaneous rate change according to Uniform model
Buffer Length = 0.05%*average rate, theta = 1.0

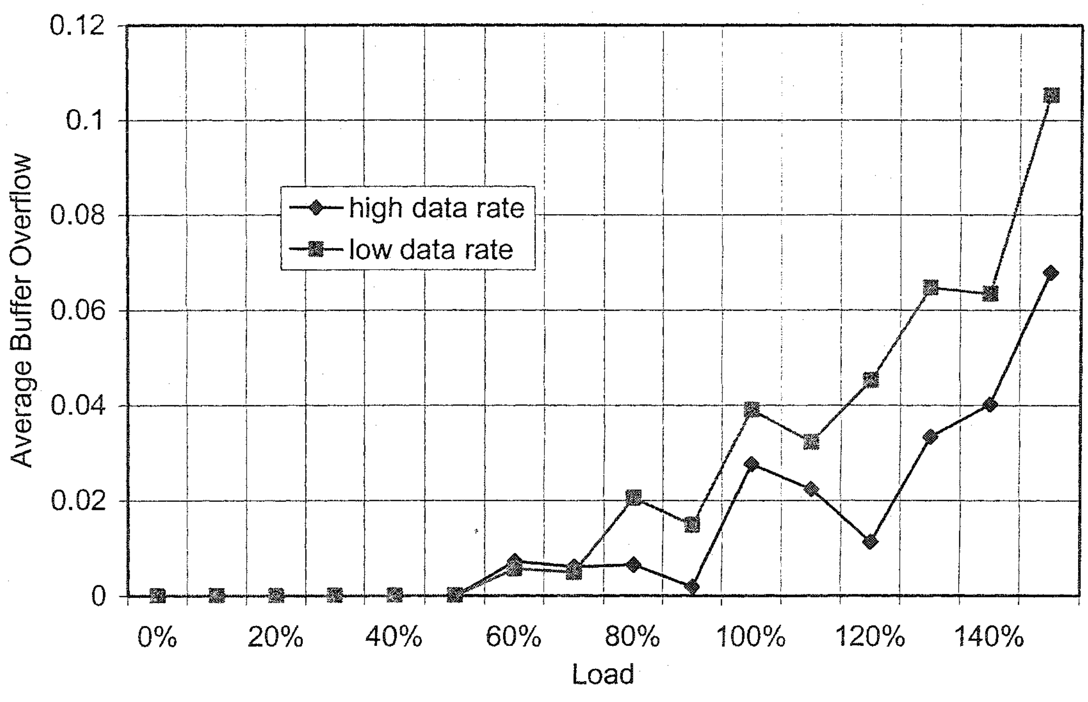


Figure 4.22 Average Buffer Overflow in Dynamic Resource Allocation
User's instantaneous rate change according to Uniform model
Buffer Length = 0.05%*average rate, theta = 1.0

Figure 4.21 and Figure 4.22 are the performance of the Average Buffer Overflow under different Data Rate Type but under the same Capacity Assignment Parameter (1.0 in this case) in the Uniform model. The performance of the Average Buffer Overflow is much improved in Dynamic Resource Allocation scheme than in Fixed Resource Allocation scheme. The performance of high data rate group users is better than the performance of low data rate group users. In the same traffic load, higher data rate means less number of users, so, less opportunity to have buffer overflow.

4.4.3 Average Queuing Delay

The result showed in this part is from simulation of model 3. And it was found that the same result also applies to model 1,4. In model 2, the similar result is got from simulation. But, because the variance of data rate is bigger than other three models, the average queuing delay is bigger than other models.

1. **The effect of the Capacity Assignment Parameter**

Figure 4.23 and Figure 4.24 show the performance of the Average Queuing Delay under different Capacity Assignment Parameters in Model 3. Figure 4.23 is the result of Fixed Resource Allocation scheme. It shows that when the value of the Capacity Assignment Parameter is less than 1, the performance of the Average Queuing Delay is almost the same. Figure 4.24 is the result of Dynamic Resource Allocation scheme. It shows that the performance of the Average Queuing Delay is largely improved in this case. And it is improves with the increasing of the Capacity Assignment Parameter. This phenomenon proves that the bigger the Capacity Assignment Parameter the better the QoS satisfaction. The same result also applies to model 1,4.

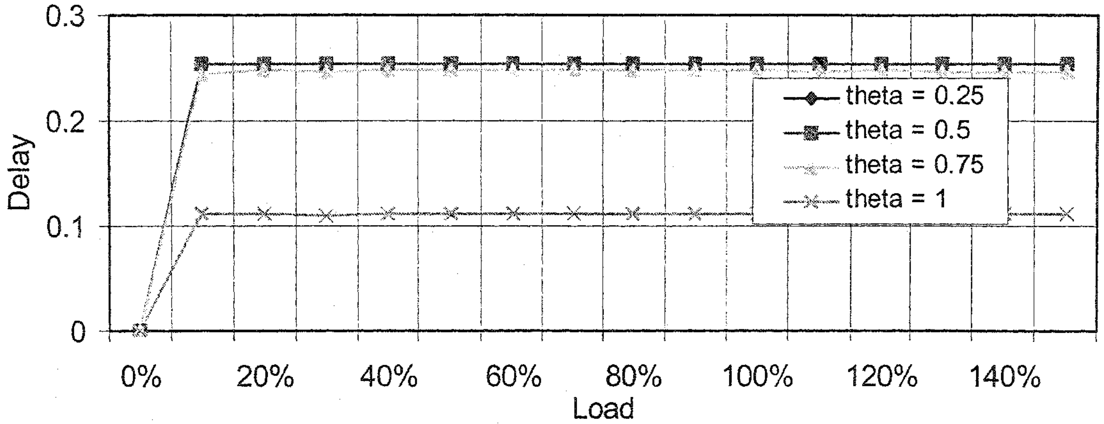


Figure 4.23 Average Queuing Delay in Fixed Resource Allocation
User's instantaneous rate changes according to model 3
Buffer Length = 0.05%*average rate, high rate type

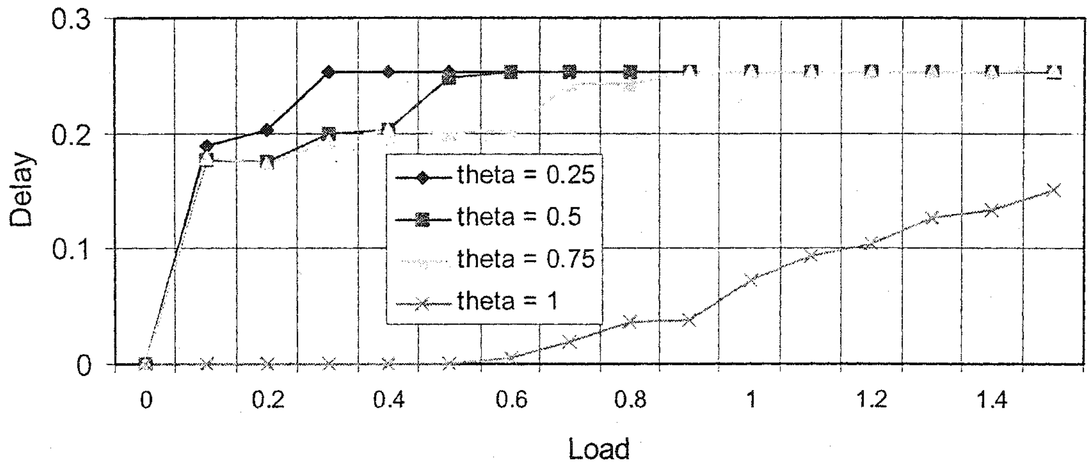


Figure 4.24 Average Queuing Delay in Dynamic Resource Allocation
User's instantaneous rate change according to model 3
Buffer Length = 0.05%*average rate, high rate type

2. The effect of Buffer Length

Figure 4.25 and Figure 4.26 show the performance of Average Queuing Delay with different Buffer Length but with the same Capacity Assignment Parameter (0.25 in this case) in model 3. The figures show that the performance related to different buffer length. Longer buffer length means user's buffer can tolerate more queuing. Figure 4.25 shows the Average Queuing Delay increases with the increasing of Buffer Length. Figure 4.26

shows the performance of Average Queuing Delay is improved in Dynamic Resource Allocation scheme than in Fixed Resource Allocation scheme.

The same result also applies to model 1,4.

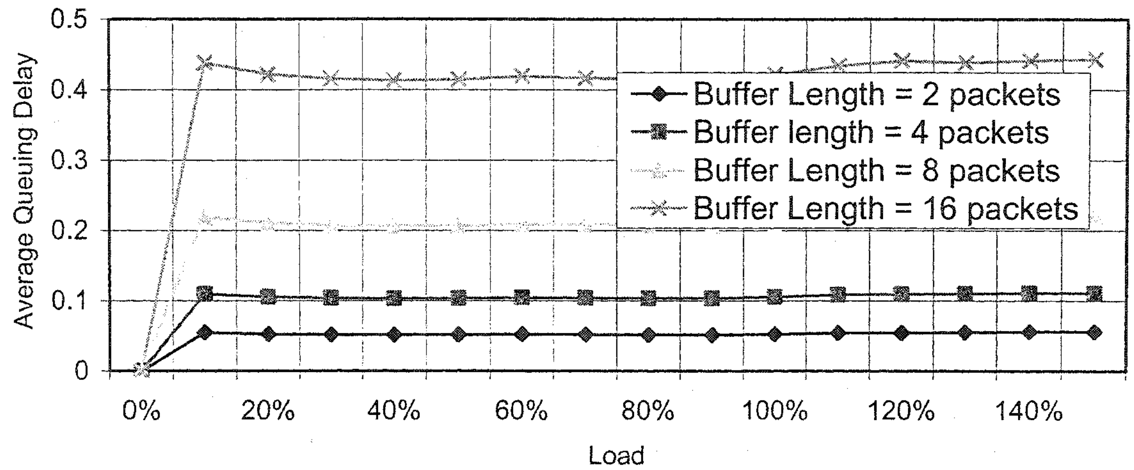


Figure 4.25 Average Queuing Delay in Fixed Resource Allocation
User's instantaneous rate change according to model 3
Buffer Length =2,4,8,16 Packets, high rate type, Theta = 0.25

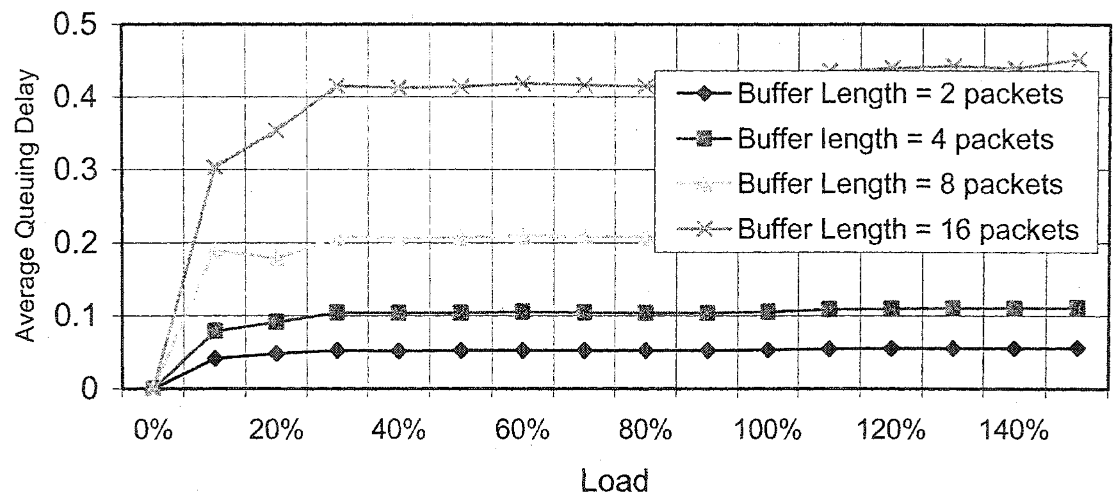


Figure 4.26 Average Queuing Delay in Dynamic Resource Allocation
User's instantaneous rate change according to model 3
Buffer Length =2,4,8,16 Packets, high rate type, Theta = 0.25,

3. **The effect of Data Rate Type (high or low)**

Figure 4.27 and Figure 4.28 are the performance of Average Queuing Delay when users are composed of different Data Rate Type but with the same Capacity Assignment Parameter (0.25) in model 3. The performance of Average Queuing Delay is slightly improved in Dynamic Resource Allocation scheme than in Fixed Resource Allocation scheme. But the combination does not affect the performance. The same result also applies to model 1,4.

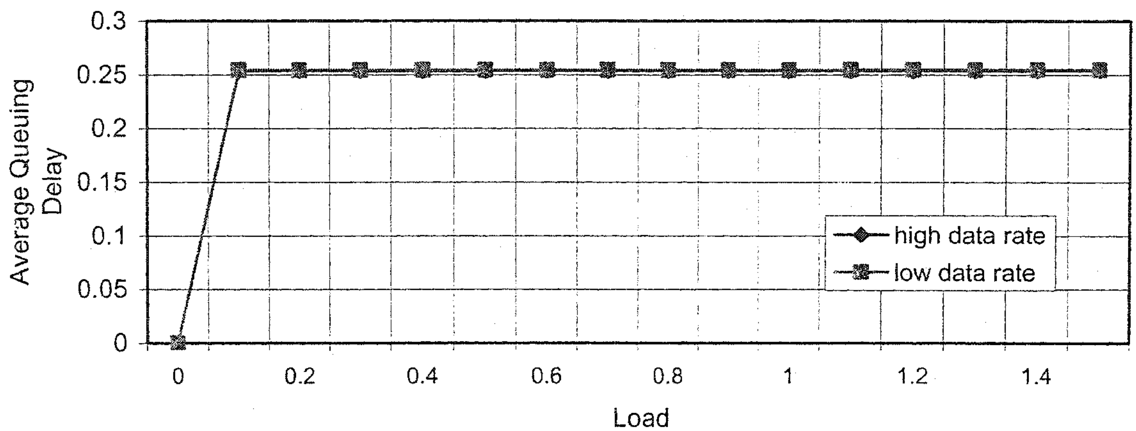


Figure 4.27 Average Queuing Delay in Fixed Resource Allocation
User's instantaneous rate change according to model 3
Buffer Length = 0.05%*average rate, theta = 0.25

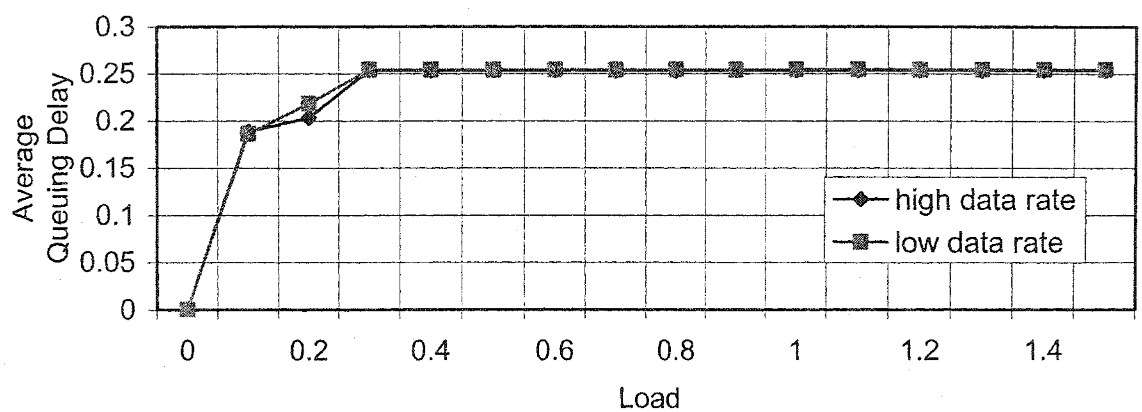


Figure 4.28 Average Queuing Delay in Dynamic Resource Allocation
User's instantaneous rate change according to model 3
Buffer Length = 0.05%*average rate, theta = 0.25

4.4.4 Average Packet Loss Rate

The result shown applies to simulation of model 4. And it was found that the same result also applies to model 1 and model 3. In model 2, the similar result is obtained from simulation. But, because the variance of data rate is bigger than other three models, the average packet loss is higher than other models.

1. **The effect of the Capacity Assignment Parameter**

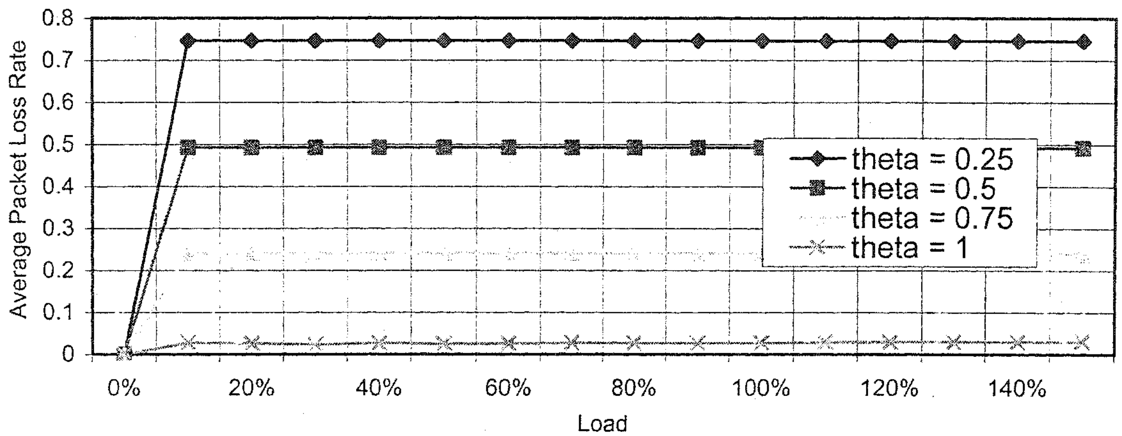


Figure 4.29 Average Packet Loss Rate in Fixed Resource Allocation
User's instantaneous rate changes according to model 4
Buffer Length = 0.05%*average rate, high rate type

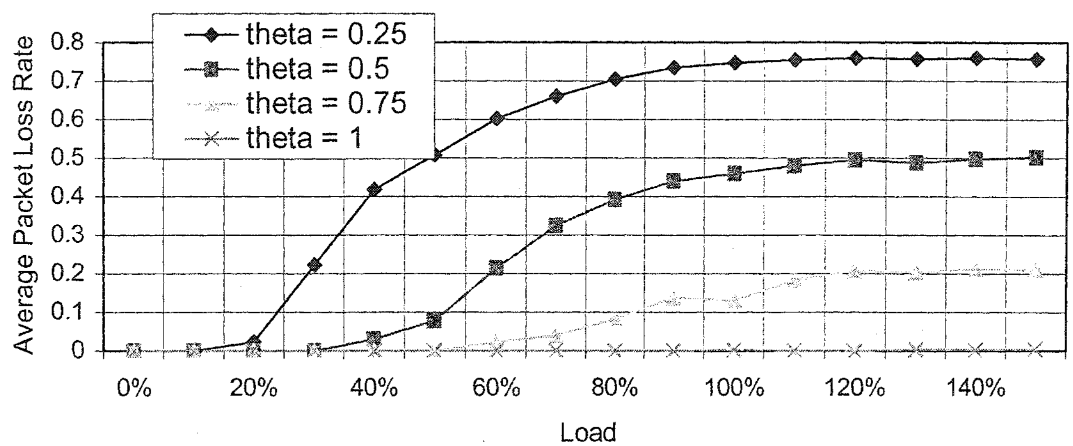


Figure 4.30 Average Packet Loss Rate in Dynamic Resource Allocation
User's instantaneous rate changes according to model 4
Buffer Length = 0.05%*average rate, high rate type

Figure 4.29 and Figure 4.30 show the performance of the Average Packet Loss Rate under different Capacity Assignment Parameters in Model 4. Both of these two figures show that the performance improves with the increasing of the Capacity Assignment Parameter. This phenomenon proves that the bigger the Capacity Assignment Parameter the better the QoS satisfaction. Figure 4.30 shows the performance of Average Packet Loss Rate is dramatically improved in the Dynamic Resource Allocation scheme than in the Fixed Resource Allocation scheme. The same result also applies to model 1,3.

2. The effect of Buffer Length

Figure 4.31 and Figure 4.32 show the performance of Average Packet Loss Rate under different Buffer Length when the Capacity Assignment Parameter is set to 1.0 in model 4. Figure 4.31 shows in fixed resource allocation schemes, the performance of Average Packet Loss Rate related to different buffer length. Longer buffer length brings less packet loss. Figure 4.32 shows the performance of Average Packet Loss Rate Average is dramatically improved in Dynamic Resource Allocation scheme than in Fixed Resource Allocation scheme. The absolute value of Average Packet Loss Rate Average is very low in this case. But the Buffer Length does not affect the performance of Average Buffer Overflow anymore. The reason why the curves oscillate is that the value of the packet loss is very low, random data rate's change can totally change the shape of the figure.

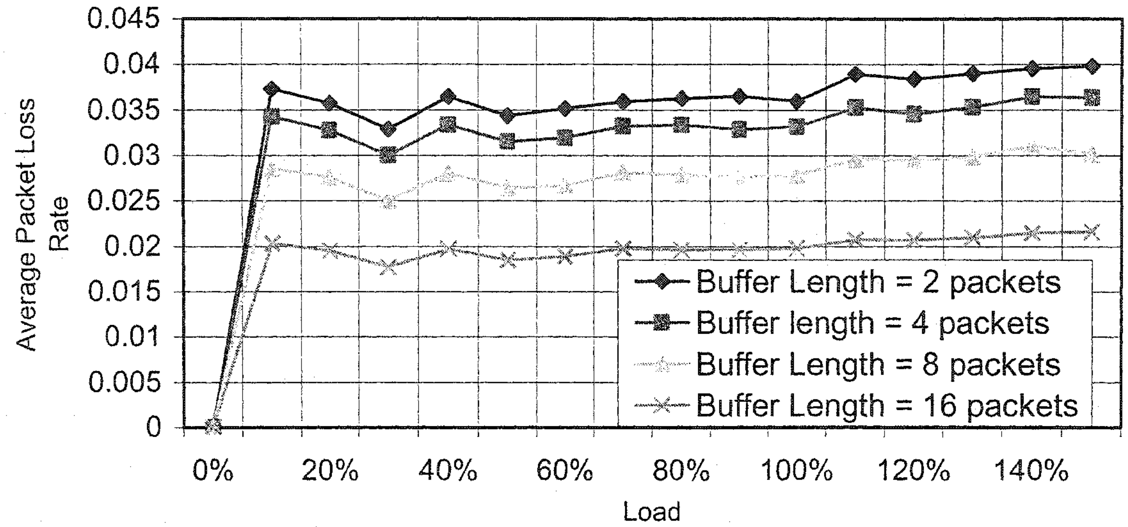


Figure 4.31 Average Packet Loss Rate in Fixed Resource Allocation
User's instantaneous rate changes according to model 4
Buffer Length =2,4,8,16 Packets, high rate type, Theta = 1.0

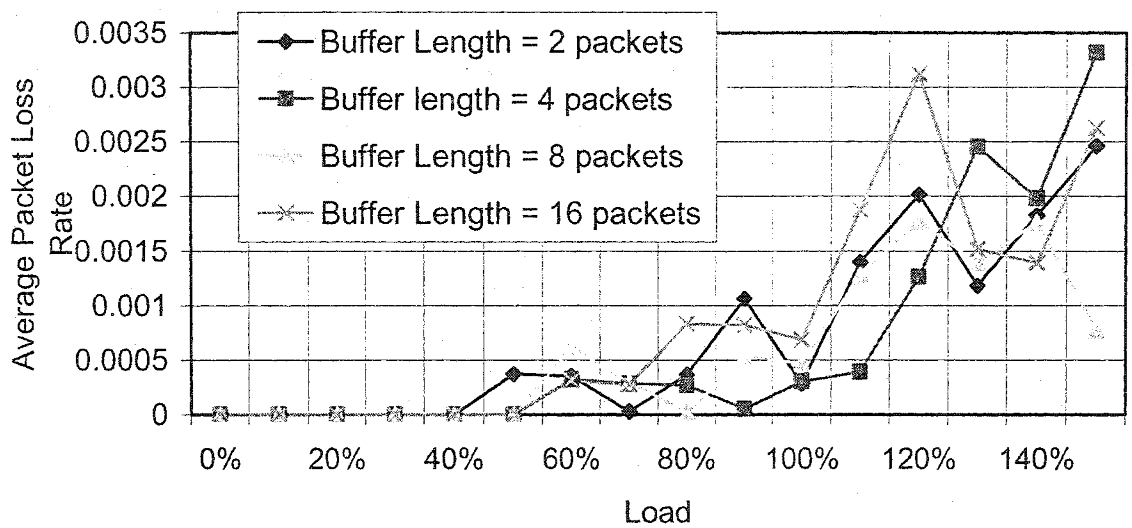


Figure 4.32 Average Packet Loss Rate in Dynamic Resource Allocation
User's instantaneous rate changes according to model 4
Buffer Length =2,4,8,16 Packets, high rate type, Theta = 1.0

Figure 4.33 and Figure 4.34 show the performance of Average Packet Loss Rate with different Buffer Length when the Capacity Assignment Parameter is set less than 1.0, here is set to 0.5. Both of these two figures show that when the Capacity Assignment Parameter is less than 1.0, the different size of buffer length does not affect the performance of Average Packet Loss Rate. But still, the performance of Average Packet Loss Rate is dramatically improved in Dynamic Resource Allocation scheme than in Fixed Resource Allocation scheme.

The same result also applies for model 1,3.

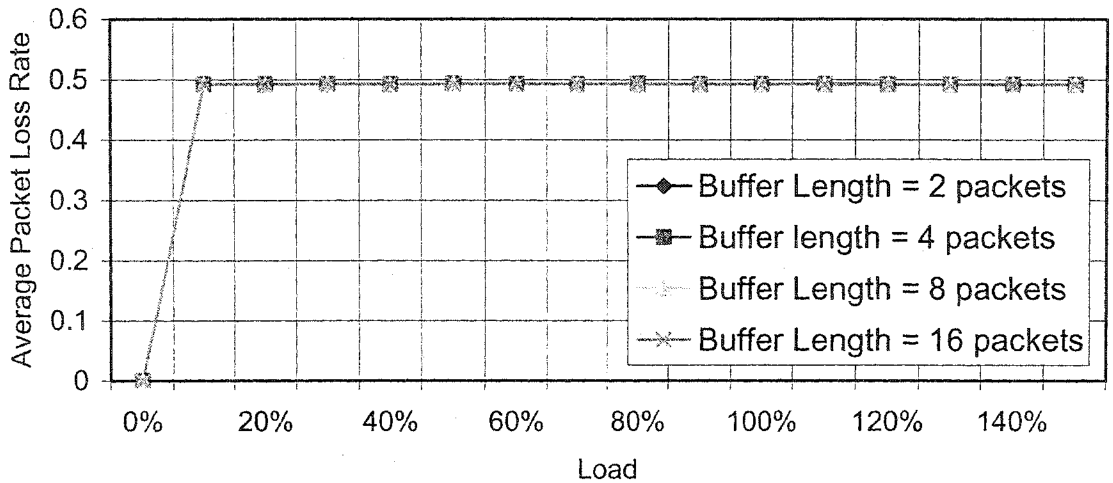


Figure 4.33 Average Packet Loss Rate in Fixed Resource Allocation
User's instantaneous rate changes according to model 4
Buffer Length =2,4,8,16 Packets, high rate type, Theta = 0.5

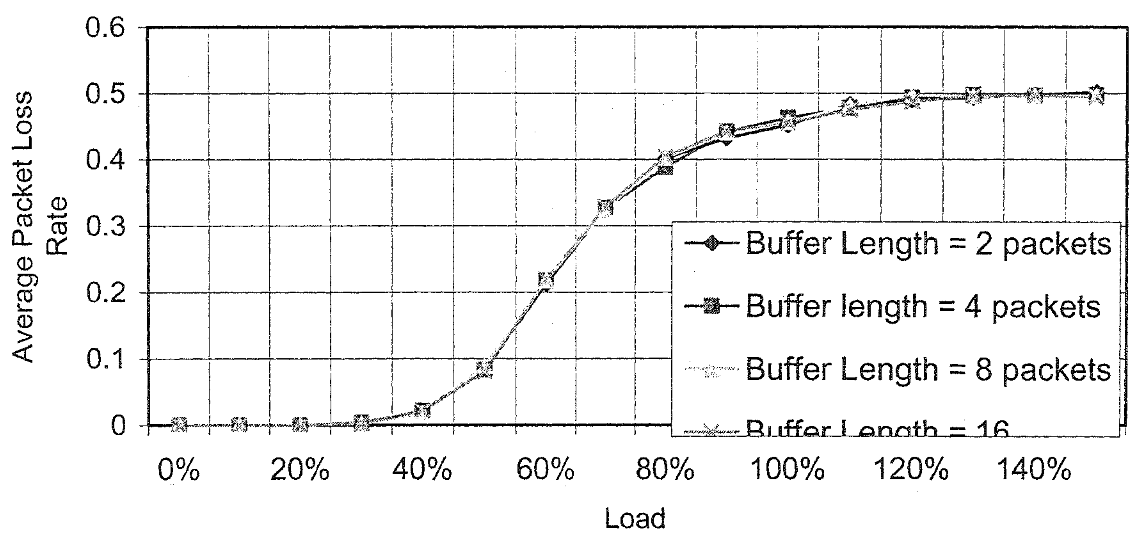


Figure 4.34 Average Packet Loss Rate in Dynamic Resource Allocation
User's instantaneous rate changes according to model 4
Buffer Length =2,4,8,16 Packets, high rate type, Theta = 0.5

3. **The effect of Data Rate Type (high or low)**

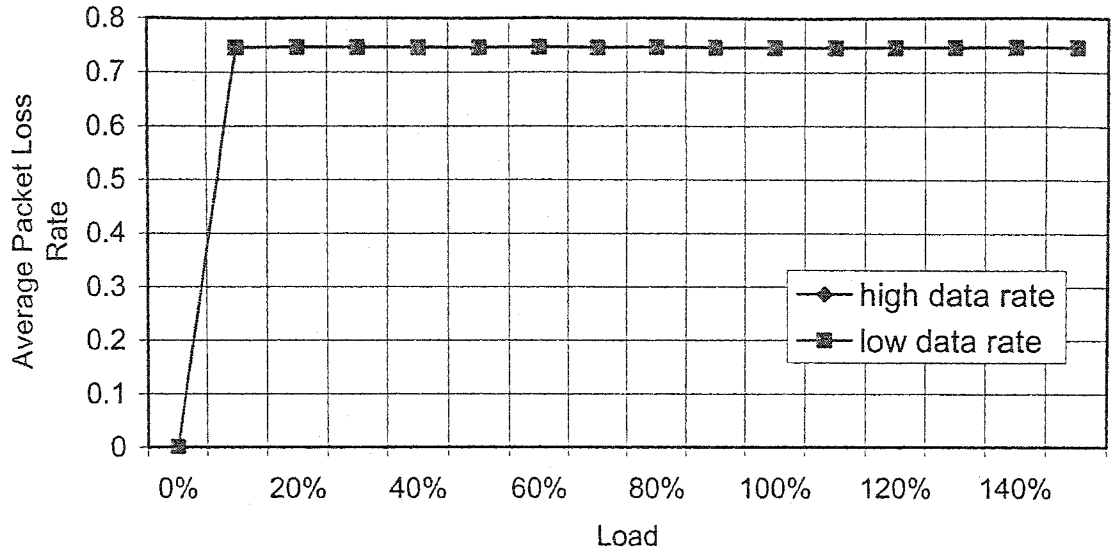


Figure 4.35 Average Packet Loss Rate in Fixed Resource Allocation
User's instantaneous rate changes according to model 4
Buffer Length = 0.05%*average rate, theta = 0.25

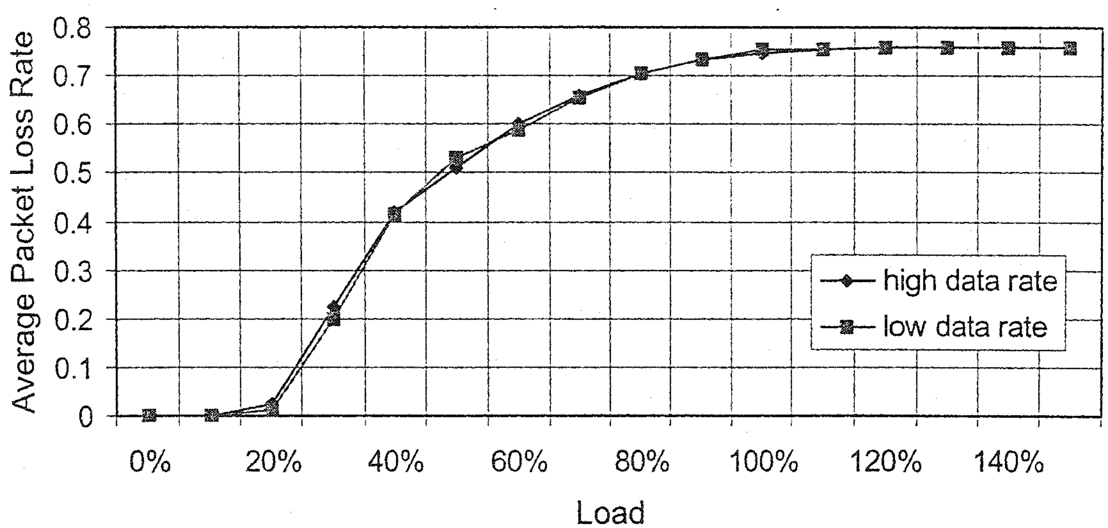


Figure 4.36 Average Packet Loss Rate in Dynamic Resource Allocation
User's instantaneous rate change according to model 4
Buffer Length = 0.05%*average rate, theta = 0.25

Figure 4.35 and Figure 4.36 show the comparison of Average Packet Loss Rate when users are composed of different Data Rate Type but with the same Capacity Assignment Parameter (0.25 in this case). The performance of Average Queuing Delay is dramatically improved in Dynamic Resource Allocation scheme than in Fixed Resource Allocation scheme. But the combination does not affect the performance. The same result also applies to model 1,3.

4.4.5 Call Blocking Probability

The results shown in this part is from simulation of model 1. And it was found that the same results are also apply to model 1, model 2 and model 3.

1. **The effect of the Capacity Assignment Parameter**

Figure 4.37 and Figure 4.38 show the performance of the Average Call Blocking Probability under different Capacity Assignment Parameters in Model 1. These two figures show that the performance is irrelevant to the resource allocation schemes. In both of these two schemes, the Average Call Blocking Probability is slightly increased together with the increasing of the Capacity Assignment Parameter.

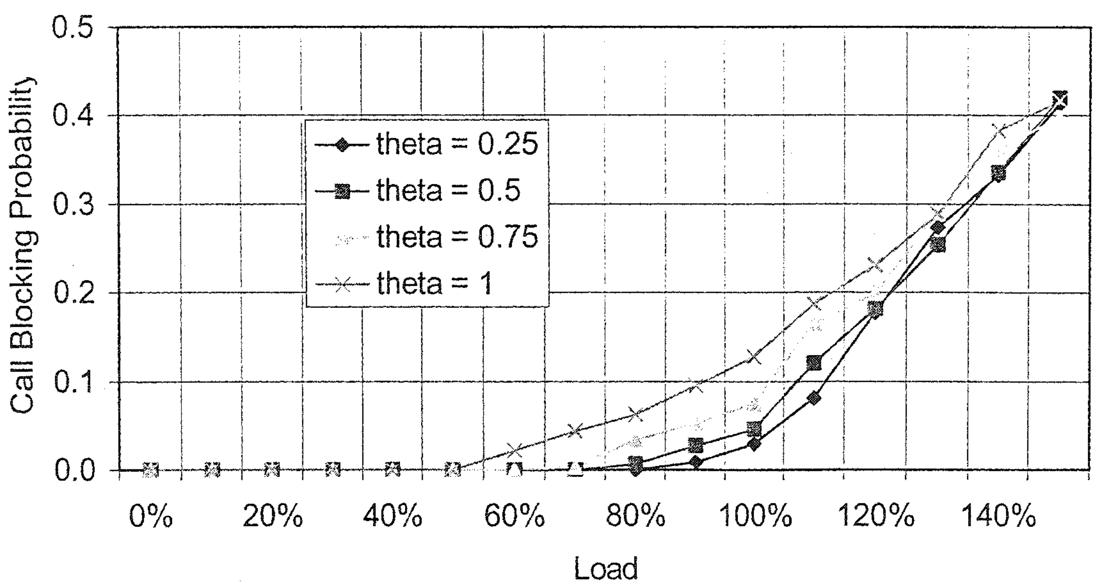


Figure 4.37 Average Call Blocking Probability in Fixed Resource Allocation
User's instantaneous rate changes according to model 1
Buffer Length = 0.05%*average rate, high rate type

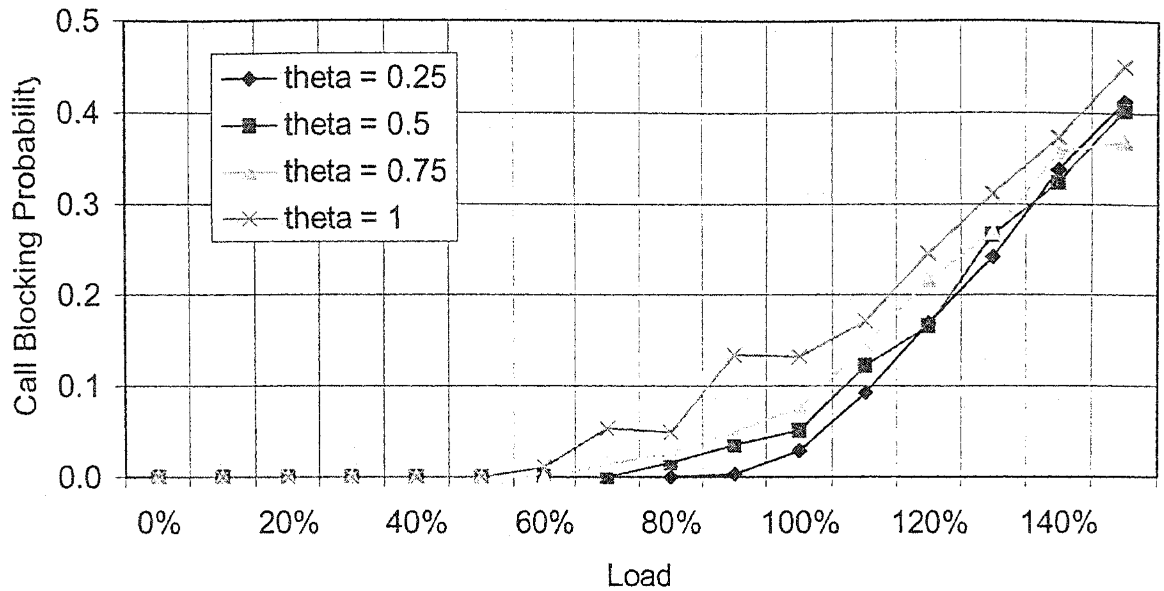


Figure 4.38 Average Call Blocking probability in Dynamic Resource Allocation
User's instantaneous rate changes according to model 1
Buffer Length = 0.05%*average rate, high rate type

2. The effect of Buffer Length

Figure 4.39 and Figure 4.40 show the performance of Average Call Blocking Probability with different Buffer Length when the Capacity Assignment Parameter is set as 1.0. Still, these two figures show that the performance is irrelevant to the resource allocation schemes and also irrelevant to the Buffer Length.

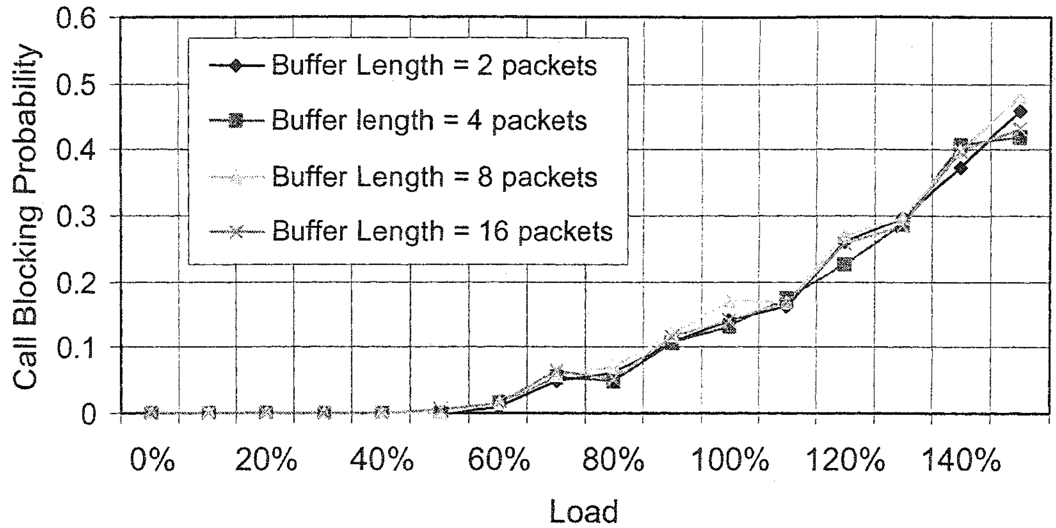


Figure 4.39 Average Call Blocking Probability in Fixed Resource Allocation
User's instantaneous rate changes according to model 1
Buffer Length =2,4,8,16 Packets, high rate type, Theta = 1

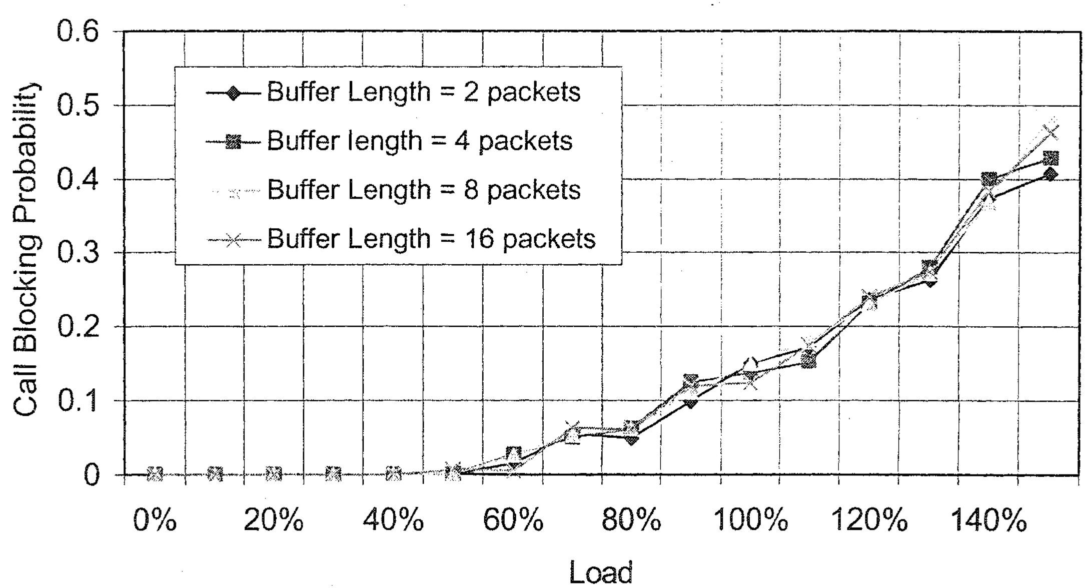


Figure 4.40 Average Call Blocking Probability in Dynamic Resource Allocation
User's instantaneous rate changes according to model 1
Buffer Length =2,4,8,16 Packets, high rate type, Theta = 1

3. The effect of Data Rate Type (high or low)

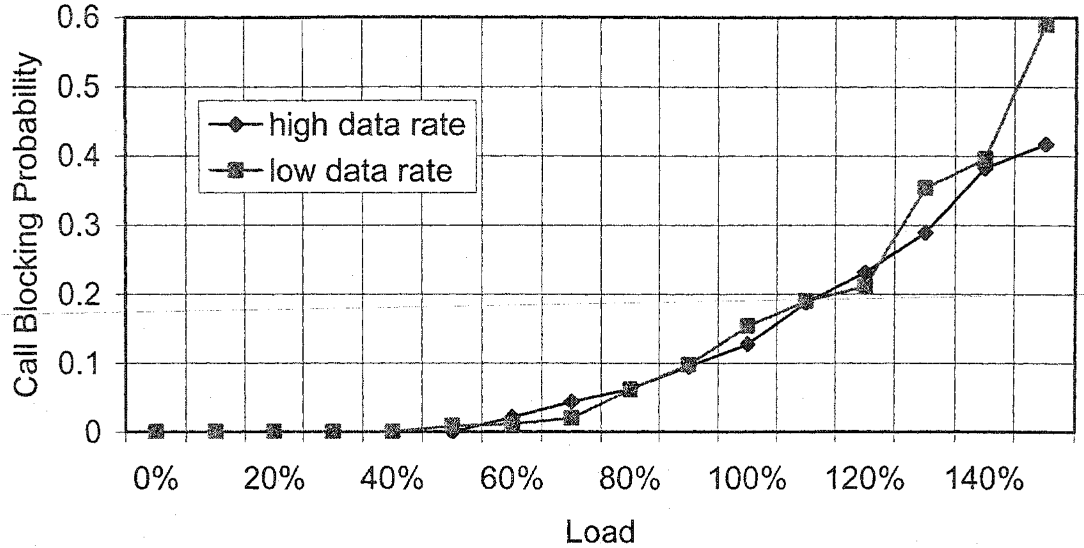


Figure 4.41 Average Call Blocking Probability in Fixed Resource Allocation
User's instantaneous rate change according to model 1
Buffer Length = 0.05%*average rate, theta = 1

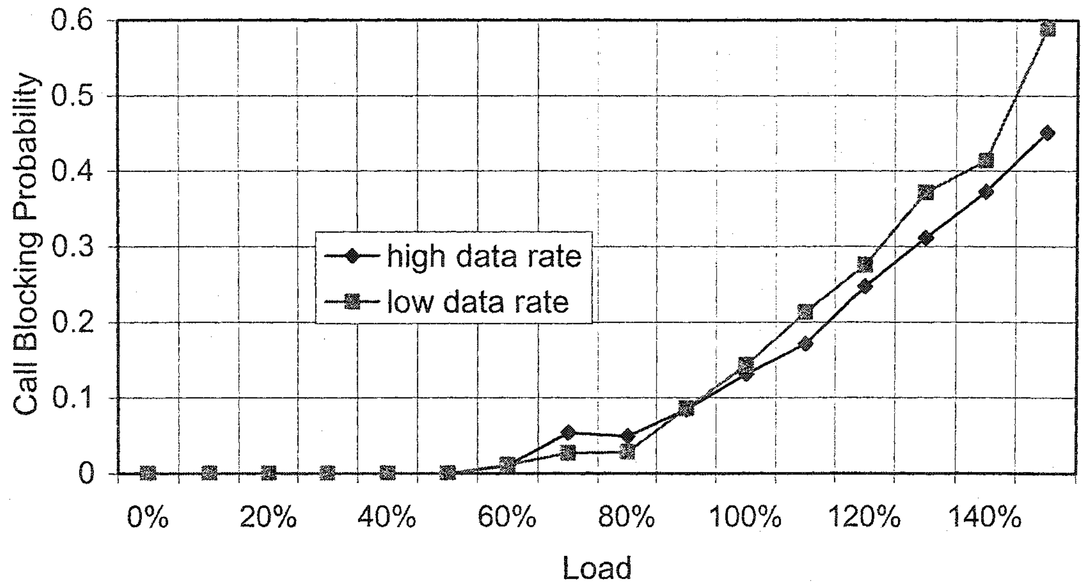


Figure 4.42 Average Call Blocking Probability in Dynamic Resource Allocation
User's instantaneous rate change according to model 1
Buffer Length = 0.05%*average rate, theta = 1

Figure 4.41 and Figure 4.42 show the comparison of Average Call Blocking Probability when users are composed of different Data Rate Type but with the same

Capacity Assignment Parameter (1.0 in this case). These two figures show that the performance of Average Call Blocking Probability is irrelevant to the resource allocation schemes. The same phenomenon in these two figures is that together with the increase of traffic load, The Call Blocking Probability is slightly high in the combination of Low Data Rate users than in the High Rate Users. The reason for that is, the Call Blocking Probability is calculated based on the number of blocked users. In the combination of Low Data Rate type, under the same traffic load, the total number of users is more than in the case of High Data Rate type.

The same result also applies to model 3,4.

4.5 Performance comparison with other papers

Some example comparisons of the performance of this thesis with other papers are given in this section to get the overall idea. Due to the difference of simulation model, simulation assumption and simulation conditions, the parameters in each model should be elaborately unified to make a fair comparison. Even so, the comparison is approximate.

4.5.1 Comparison with [25]

In [25], S.Ni and S.Haggman provide an approximation method to evaluate the GPRS performance of single-slot service. And a simulation is made. They also addressed that" The multi-slot services cause higher blocking probability and longer delay to the network than the single-slot service. However, those effects can be reduced by implementing a GPRS resource allocation scheme with flexible multi-slot services."[25]

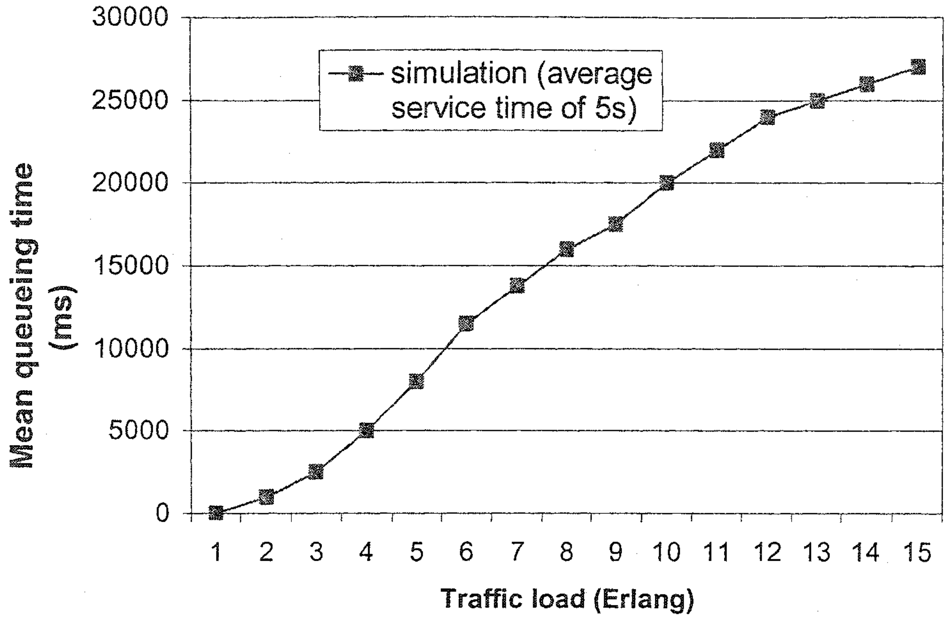


Figure 4.43 The mean queuing time of single-slot service for the average service time of 5s.

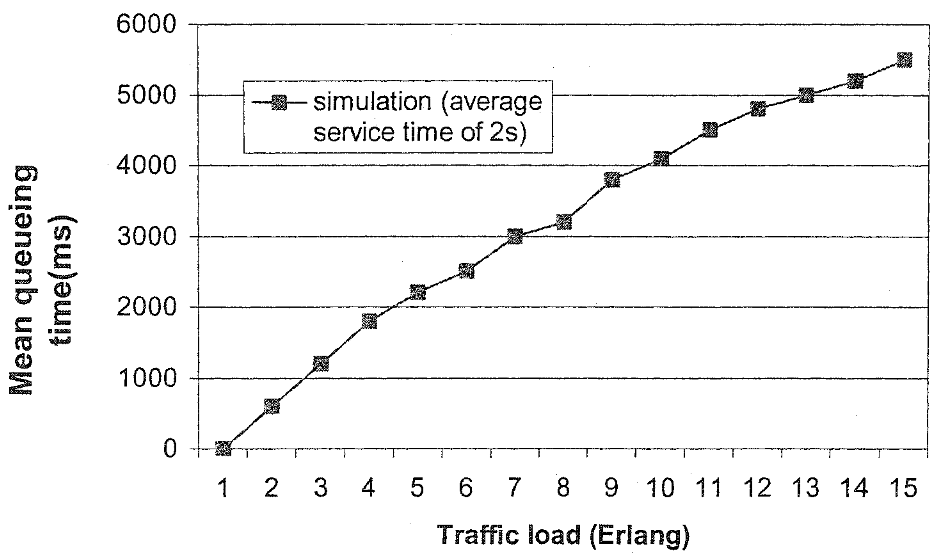


Figure 4.44 The mean queuing time of single-slot service for the average service time of 2s.

Figure 4.43 and figure 4.44 are from [25]. The assumption in the simulation of [25] is: 4 carriers, 4*8 =32 channels in a cell, The average service time of circuit switched services is exponentially distributed with a mean of 180s. Figure 4.43 is talking about the GPRS message whose message size is exponentially distributed with mean of 5*13.4kb, corresponding to the mean service time of 5s with single slot transmission. In figure 4.44, the mean service time is 2s with single slot transmission. To normalize the result, we first normalize the traffic load. From [26], we use the following definitions.

4. Random call arrival rate λ calls/second.

5. The average service time $\frac{1}{\mu}$ second

6. Traffic load $\rho = \frac{\lambda}{\mu}$ (Erlang)

7. Throughput efficiency (or Relative traffic load as in [26])

$= \frac{\rho}{\frac{c}{k}}$, ($\frac{c}{k}$ is the number of channels in one cell).

In this case, there are 32 channels in one cell, so the normalized traffic load is from 0.03 (1 Erlang) to 0.47 (15 Erlang).

To compare with the result in Figure 4.24, we first convert the normalized result of delay to the number of delayed packets. There are 26×8 slots (each slot serve a small packet) in one carrier in one multiframe, 0.1 delay from our simulation means $0.1\times 26\times 8$ packets delay in one multiframe. So, the queuing time is translated to $0.1\times 26\times 8\times 120ms = 2496ms$. In the same way, 0.2 delay means 4992ms, 0.3 delay means 7488 ms. The result looks reasonable compare with [25]. In the case when the Capacity Assignment Parameter is 1, the delay performance of Dynamic Resource Allocation is better than [25]. Figure 4.45 is the performance comparison of figure 4.24 and figure 4.44.

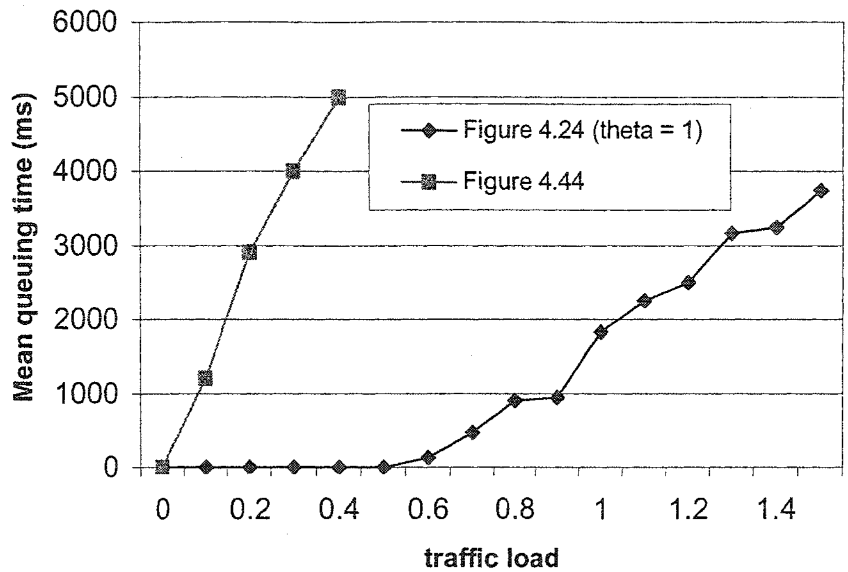


Figure 4.45 Comparison with reference [25]

4.5.2 Comparison with [27]

The authors in [27] analyzed the performance of radio resource allocation in GSM/GPRS networks and then simulated it. There are 32 channels in a cell are assumed in this simulation. Three different cases are considered depending on whether to provide buffers for GPRS data packets or not when no channels are available or when they are pre-empted by voice calls. They are the no-buffer case, buffer-only-for-pre-empted-GPRS-packets case, and buffer-for-GPRS case. We compare with the case of buffer-for-GPRS case.

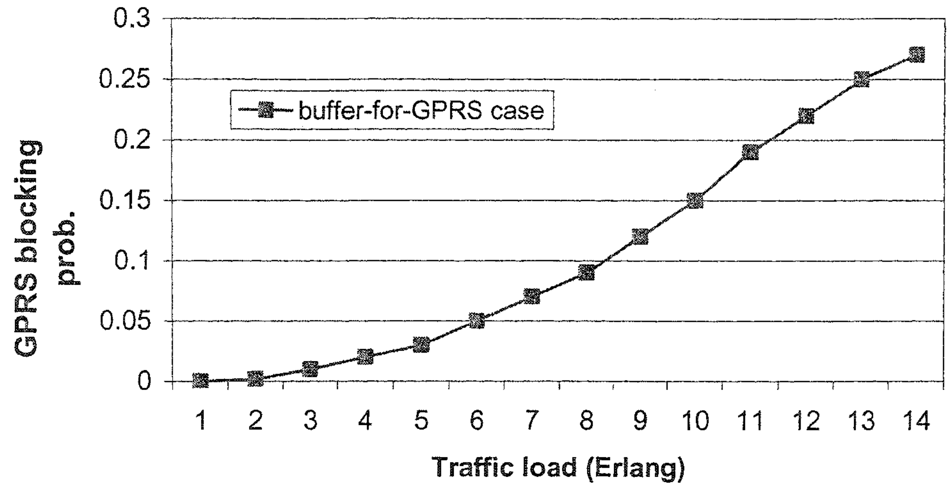


Figure 4.46 Blocking Probability of GPRS packets for the buffer-for-GPRS case

To normalize the result, we first normalize the traffic load. The same as the case before, the load is from 0.03 (1 Erlang) to 0.44 (14 Erlang).

Compare figure 4.46[27] with the case theta = 1 in figure 4.38. The two figures follow the same trend. Blocking probability increases with the increasing of traffic load. The performance in figure 4.38 is better than in figure 4.46.

Chapter 5

Conclusions and Future Work

5.1 Conclusion

In this thesis, a simulation model of resource allocation schemes in GPRS/EGPRS was built. By introducing the Capacity Assignment Parameter, the QoS requirement from users is met in this simulation. Both fixed resource allocation and dynamic resource allocation schemes are simulated and compared. In each scheme, four traffic models are simulated and compared.

The contributions of this thesis are summarized below:

1. The performance of dynamic resource allocation is fundamentally improved than of fixed resource allocation. In dynamic resource allocation schemes, the Packet Efficiency is achieved; Average Buffer Overflow, Average Queuing Delay and Average Packet Loss rate is smaller than in fixed resource allocation schemes. Call Blocking Probability is not affected by the allocation schemes.
2. The Packet Efficiency and the Call Blocking Probability are not affected by the Buffer length in both Fixed and Dynamic Resource Allocation Schemes; The Average Buffer Overflow and the Average Packet Loss rate decrease with the increasing of Buffer Length in Fixed Resource Allocation Schemes, but are not affected in the Dynamic Resource Allocation Schemes even though the performance are fundamentally improved; The Average Queuing Delay increases

with the increasing Buffer Length in both Fixed and Dynamic Resource Allocation Schemes.

3. The Packet Efficiency is not affected by the Capacity Assignment Parameter in Fixed Resource Allocation Schemes, but increases with the decreasing of the Capacity Assignment Parameter in Dynamic Resource Allocation Schemes; The Average Buffer Overflow, The Average Queuing Delay and the Average Packet Loss rate decrease with the increasing of the Capacity Assignment Parameter in both Fixed and Dynamic Resource Allocation Schemes; the Call Blocking Probability very slightly increases with the increasing of the Capacity Assignment Parameter.
4. Under different traffic models, if the Average Data Rate and the Variance of Average Data Rate is the same, then the system performance appears independent from the traffic models. If the variance is set bigger (apart from traffic models), then the system performance will worse.
5. When the traffic load is the same, the combination of the High Data Rate users or of the Low Data Rate users almost does not bring changes to the system performance, only the Call Blocking Probability is slightly high in the combination of Low Data Rate users than in the High Data Rate Users.
6. With the introduction of the Capacity Assignment Parameter in this simulation model, the results when system can only partially meet users' data rate requirements are compared.

5.2 Future Work

In this simulation, to concentrate on the problem of Resource Allocation in GPRS/EGPRS systems, we simplified this model. For future work, we suggest to add following work into the simulation:

1. Add the effect of multi-cell interference, adaptive antenna, hand-over, multipath fading, shadowing and distance attenuation into consideration.
2. In this simulation, when transmission failure happens, the packet lost. Retransmission process should be considered in the future.
3. In this simulation, only the performance of uplink direction of data packets transmission is investigated. It is necessary to simulate both uplink and downlink in the future.

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