Adaptive Resource Assignment along with Overload Control for the GSM/EGPRS Networks

Zhengliang Zhang

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ABSTRACT

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Enhanced General Packet Radio Services (EGPRS) is one of the proposals for third-generation (3G) wireless services. EGPRS is also the evolutionary path for GSM and IS-136 standards towards their next-generation wireless systems. The 3G services are categorized into the background, conversational, interactive and streaming services. Therefore, GSM towards 3G is staged into two phases. The phase one of EGPRS to provide Internet access services is known as General Packet Radio Service (GPRS). The phase two of EGPRS to provide 3G services integrates with the Enhanced Data rates for the GSM Evolution (EDGE).

To provide the various 3G services and to achieve more efficient utilization of the frequency spectrum, our work is to focus on the evolution of the system capacity and performance for the GSM/EGPRS networks. Therefore, an Adaptive Resource Assignment along with Overload Control (ARAOC) algorithm has been developed while integrating adaptive channel allocation, call admission control, frequency hopping and new congestion control schemes. Our simulation results show that this algorithm can greatly improve the system capacity and performance as well as the QoS for users. The influence of the variable parameters of user data rates, channel buffer size, and channel assignment parameter to the system capacity and performance, will be investigated.

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Abbreviations

2G: Second Generation

3G: Third Generation

8-PSK: 8-Phase Shift Keying

ABO: Average Buffer Overflow

AMPS: Advanced Mobile Phone Services

AP: Assignment Policy

AQD: Average Queuing Delay

ARAOC: Adaptive Resource Assignment along with Overload Control

AUT: Average User Throughput

BS: Base Station

BSC: Base Station Controller

BSS: Base Station Subsystem

BTS: Base Transceiver Station

CAS: Capable Average SIR

CBP: Call Blocking Probability

CBS: Channel Buffer Size

CCH: Control Channel

CDMA: Code Division Multiple Access

CDP: Call Dropping Probability

CFN: Carrier Frequency Number

CHR: Call Handoff Rate

EDGE: Enhanced Data for Global Evolution

EGPRS: Enhanced GPRS

ETSI: European Telecommunications Standards Institute

FCA: Fixed channel Allocation

FDMA: Frequency Division Multiple Access

FIFO: First in First out

GERAN: GSM/EDGE Radio Access Network

GGSN: Gateway GPRS Support Node

GMSC: Gateway Mobile Switching Centre

GMSK: Gaussian Minimum Shift Keying

GPRS: Global Packet Radio Services

GSM: Global System for Mobile Communications

HLR: Home Location Register

HSCSD: High Speed Circuit Switched Data

IMT-2000: International Mobile Telecommunication 2000 Standards for 3G

LA: Link Adaptation

MAC: Medium Access Control

MCS: Modulation Coding Scheme

NIP: Number of Iteration Program

MS: Mobile Station

MSC: Mobile Switching Centre

NSS: Network and Switching Subsystem

NUE: Network Utilization Efficiency

OSS: Operation Support Subsystem

PCU: Packet Control Unit

PDC: Personal Digital Cellular

PDP: Packet Data Protocol

PDU: Packet Data Unit

PLMN: Public Land Mobile Network

PTP: Point-To-Point

PTM: Point-To-Multipoint

QoS: Quality of Service

RLC: Radio Link Control

RMS: Required Minimum SIR

Rtype: Rate Type

SGSN: Serving GPRS Support Node

SRA: Static Resource Assignment

SIR: Signal-to-Interference Ratio

SLL: Sector Load Limit

SLT: Sector Load Threshold

SMS: Short Messaging Services

Ta: Threshold of Assault

TCH: Traffic Channels

TDMA: Time Division Multiple Access

TBF: Temporary Block Flow

TC: Traffic Class

TFI: Temporary Flow Identifier

Tr: Threshold of Reduction

TS: Time Slot

UMTS: Universal Mobile Telecommunications System

USF: Uplink State Flag

VBO: Variance Buffer Overflow

VQD: Variance Queuing Delay

WCDMA: Wideband CDMA

Chapter 1
Introduction

1.1 Overview

In recent years, the public desire for mobile communication, as evidenced by the popularity of cellular phones, pagers, and laptop computers, has grown rapidly in demand for Internet access. With over one billion mobile phone users estimated by the end of 2002, and packet-based multimedia services, including IP telephony, accounting for over 50 percent of all wireless traffic, it is only natural to increase more capacity in the mobile network, and higher bandwidth in the radio link, radio access network, and core network [1]. Presently, there are fundamentally two types of second-generation (2G) digital networks: Time-Division Multiple Access (TDMA) and Code-Division Multiple Access (CDMA). TDMA is applied in GSM, IS-136, IS-54 and Personal Digital Cellular (PDC) systems, whereas IS-95 uses CDMA. Today, GSM is the mobile radio standard with the highest worldwide penetration.

In the foreseeable future, more advanced services for 3G services will bring together, current voice and low-data-rate services to the futuristic four basic categories encompassing the background, conversational, interactive and streaming services, which can also be divided into two main services: Real Time services (e.g., voice, video conferencing, real-time image transfer, etc.), and Non Real Time services (e.g., database applications, web browsing, email, streaming video and sound, etc.) [2].

To satisfy the marketing and service needs, the ongoing drafting on the definition of third-generation (3G) mobile radio system (IMT-2000) in the America, Asia and Europe

[3] will result in the requiring technical standards. Wideband CDMA (WCDMA), also known as International Mobile Telecommunications in 2000 (IMT-2000) or Universal Mobile telecommunications system (UMTS), has emerged as one of the leading 3 G standards. These proposed systems aim to support a wide range of bearer services from voice and low-rate to high-rate data services with up to at least 384 kb/s in wide area coverage, and 2 Mb/s in local areas coverage [4].

GSM system greatly successes towards 3G services accomplished by EGPRS (Enhanced Global Packet Radio Service). The EGPRS concept is an evolution of GSM. It 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. The EGPRS specifications are to be developed in two phases. The first phase, which is already specified, being GPRS [5-7], represents a packet switched-core network and an air interface based on Gaussian Minimum Shift Keying (GMSK) modulation. GPRS is designed for best-effort packet data services. The second phase of EGPRS specifications introduces a new air interface, called Enhanced Data Rate for GSM evolution (EDGE), to support higher-level rates. The EDGE concept of using a higher-level modulation, 8-shift keying (8-PSK), supplementing to the traditional GMSK [8], with enhanced data rates for existing cellular systems in existing spectrum being standardized for both GSM and TDMA/136 (D-AMPS). Therefore, EDGE is a common evolution towards providing third generation services in two major cellular standards.

The current mobile communication networks apply the following three major multiple access techniques: Frequency Division Multiple Access (FDMA), Time Division Multiple Access (TDMA) and Code Division Multiple Access (CDMA).

FDMA systems, for example, Advance Mobile Phone Service (AMPS), allocating the entire spectrum to the service area is divided into channels of appropriate bandwidth, carrying information for users.

TDMA systems, such as IS-136, GSM, DCS1800, IS-54, and PDC systems, divide the radio spectrum into multiple frequency channels, each with several time slots, and in each slot allowing only one user to either transmit or receive.

CDMA system, like IS-95, is based on spread spectrum transmission. All users within a cell jointly share the same frequency band, and are separated by different spreading codes assigned.

In the aforementioned mobile cellular networks, the cells are grouped into clusters. The number of cells in a cluster must be determined so that the cluster can be applied continuously within the coverage area of an operator. The typical clusters contain 3, 4, 7 or 12 cells. The smaller the number of cells per cluster is, the more channels per cell will be. The capacity of each cell will therefore be increased. However, a balance must be found in order to avoid the co-channel interference that could occur between neighbouring clusters. This interference is produced by the small size of the clusters (the size of the cluster is defined by the number of cells per cluster). In reducing interference, a selective cell with coverage of 120 degrees, using a uni-direction antenna, is used.

Therefore, various channel allocation algorithms are utilized in order to reduce cell cluster size and/or co-channel interference. Among these algorithms we propose static resource assignment (SRA) and Dynamic Resource Assignment, and focus on Adaptive Resource Assignment which combines the above two algorithms.

1.2 Objective of the Thesis

Future mobile communication networks aim to provide not only high quality voice service but also integrated with various data services, such as voice conversation, file transfer, web browsing and multimedia services. To support these integrated services, the efficient use of the available resources (especially the spectrum resource) is a major issue. Thus, we will firstly propose dynamic co-channel interfering algorithms to estimate Signal-to-Interference Ratio (SIR) for each channel based on cell cluster size of 1 and 1/3, and to actualize SIR based on measurement at the mobile terminal. Secondly, a dynamic channel allocation and frequency hopping schemes, which is to be discussed in detail in chapter 4, will be developed to reduce co-channels interferences.

We will also propose an adaptive resource assignment along with overload control (ARAOC) algorithm, integrating schemes of channel allocation, call admission control, frequency hopping and a new congestion control. The consideration of algorithms is not only improving the capacity, but also lessening the quality deterioration of the on-going calls and the call dropping rate and packet loss rate, caused by admitting new calls. Finally, we will simulate the two algorithms of Static Resource Assignment (SRA) and ARAOC in GSM/EGPRS systems, and discuss their performances and QoS.

Most recent mobile systems (such as IS-136 and GSM) provide voice services with tolerable call dropping probabilities around 2%, speech rate at 13 kb/s, data rate up to 9.6 kb/s. In comparing these integrated data services with GSM, we assume the characteristics and environment as similar as those in GSM system. For instance, channel data rate of 13.0kb/s, modulation of 0.3 GMSK and call dropping probability of 2%, etc

[9]. Then we investigate the system capacity and performances both introducing various services in SRA and ARAOC algorithms. Hence, we can infer that the system, which uses different modulations (GMSK and 8-PSK) and channel data rates, can support maximum user rate and system capacity.

The objectives of this thesis work are:

- To investigate the relationship between the system capacity and the quality of service in order to increase the utilization of radio resources;
- To develop algorithms for system to manage the radio resource efficiently, and to control call admission, overload and congestion, respectively;
- To investigate the influence of various parameters with numerical insight results, such as variable user data rates, several channel buffer sizes, and different channel assignment parameters.

The simulations are programmed with Visual C++.

1.3 Outline of the Thesis

The thesis is organized as follows:

In chapter 1, we introduce current second-generation (2G) wireless system and the requirements of 3G services, as well as the method of GSM evolution towards 3G services. We then review the relationship between the cell cluster size and co-channel interference. Having introduced a new adaptive resource assignment along with overload control, we further describe the purpose, scope and organization of this thesis.

In chapter 2, a short review of GSM evaluation will be presented. We will introduce the concept, system architecture, frame structure, logical channels and signal processing of GSM.

Then, we will briefly describe the system architecture, radio resource management, logical channel, service classes and coding schemes for GPRS in Chapter 3. In addition, EDGE modulation, fast packet control channels and multiplexing capability of EGPRS will also introduced in chapter 3.

In the first part of chapter 4, two algorithms of static resource assignment (SRA) and Adaptive Resource Assignment along with Overload Control (ARAOC) are being introduced for call admission and congestion control. In the second part of chapter 4, a cellular network and model are being illustrated. Our work is focus on the performance evaluation of users and the GSM/EGPRS systems employing ARAOC algorithm. A comparable SRA algorithm is also being presented in this part. Finally, we show the simulation results of the comparison of two algorithms, and discuss the performance of systems and users.

Lastly, we summarize the conclusion of our thesis, and list the future work areas as remarked in chapter 5.

Chapter 2
Overview of GSM Network

2.1 The evolution of GSM

In 1992, the Global System for Mobile communications (GSM) is introduced as phase 1 standard to commercial service (i.e., telephone, short message). In the phase 2 of the standard, GSM is published as a Pan-European digital cellular standard by ETSI. It has completed original GSM design task, and established a framework for ongoing technology enhancement. The GSM mainly supports speech service at the rate of 13.4 kbps, and variety of data services, rating up to 9600bps. Although existing GSM systems do not support Internet access, high data rate, and video services, it is used as a worldwide mobile communication standard over 200 GSM networks (including DCS1800 and PCS 1900) in 110 countries. The number of subscribers worldwide is expected to exceed one billion by the end of 2003.

In order to accomplish the requirements for 3 G services, which include background, conversational, interactive and video streaming services, GSM evolution can satisfy 3G service demands. The solution is the phase 2+ of GSM, called the General Packet Radio Service (GPRS) and Enhanced GPRS (EGPRS). It is implemented in two steps.

The first step towards 3G is to choose Global Packet Radio Service (GPRS) by introducing two IP components in their cellular networks, namely, the Serving GPRS Supporting Node (SGSN) and the Gateway GPRS Supporting node (GGSN). The GPRS can rate up to 171 kbps.

The next step is to increase data rate up to 384 kb/s by changing the modulation scheme from Gaussian minimum Shift Keying (GMSK) to 8-PSK, while keeping the GPRS core structures. This step is called the Enhanced GPRS (EGPRS) with enhanced data rate for GSM evolution (EDGE) [22, 23]. With the introduction of the radio spectrum, EDGE will support up to 2 Mb/s in low-mobility environments [10]. The GSM/EDGE radio access network (GERAN) [20,21] will be able to offer the same services as WCDMA by connecting to similar core network.

2.2 GSM System Architecture

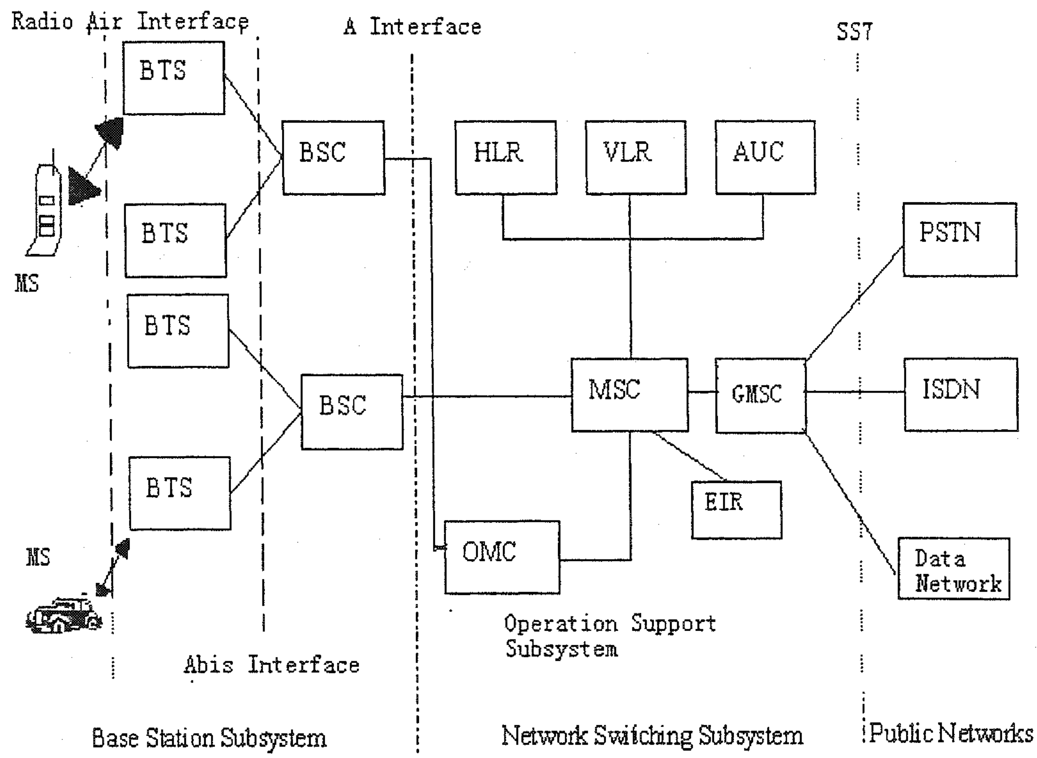


Figure 2.1 GSM system architecture and interfaces

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 Support Subsystem (OSS).

Figure 2.1 shows the block diagram of the GSM system architecture and interfaces. The Mobile Stations (MS) communicate with the Base Station Subsystem (BSS) over the radio air interface.

A. The Base Station Subsystem (BSS)

The BSS, also known as the radio subsystem, provides radio transmission paths between the Mobile Stations (MS) and the Mobile Switching Centre (MSC). The BSS also manages the radio interface between the mobile stations and all other subsystems of GSM. It consists of BTSs (Base Transceiver Stations) and BSCs (Base Station Controllers).

BTS (Base Transceiver Station)

The BTS corresponds to the transceivers and antennas used in each cell. Its transmitting power defines the size of a cell. The BTS comprises the radio transmission and reception devices, and also manages the signal processing related to the air interface. Each BTS has between one and sixteen transceivers.

BSC (Base Station Controller)

The BSC controls a group of BTS and manages their radio resources. A BSC is principally in charge of handovers, frequency hopping, exchange functions and control of the radio frequency power levels of the BTSs.

B. The Network and Switching Subsystem (NSS)

The NSS, which includes several databases in order to store information about the subscribers, is responsible for managing the communications between mobile users and other users, and their mobility.

MSC (Mobile Switching Centre)

The MSC is basically an ISDN-switch, coordinating and setting up calls to and from MSs. It performs authentication to verify the user's identity and to ensure the confidentiality of the calls.

HLR (Home Location Register)

The HLR database is used to store permanent data of subscribers belonging to the covering area of a MSC. It also stores service profiles, location area, and activity status.

The Authentication Centre (AuC)

The AuC database contains the subscriber authentication keys and the algorithm required to calculate the authentication parameters to be transferred to the HLR.

The Equipment Identity Register (EIR)

The EIR database contains information about the capabilities and identity of the mobile equipment. It prevents calls from unauthorized MSs.

VLR (Visitor Location Register)

The VLR database contains information from a subscriber's HLR necessary in order to provide the subscribed services to visiting users, to assure the subscribed services without needing to ask the HLR when a communication is established. The temporary information in VLR is cleared when the mobile station roams out of this service area.

GMSC (Gateway Mobile Switching Centre)

A GMSC is a gateway node interconnecting two networks, and the interface between the mobile cellular network and PSTN networks. It is often implemented in same machines as the MSC.

C. Interfaces

- **Um:** The radio air interface of GSM is also called Um. It is used for exchanges between a MS and a BSS.

- **Abis:** The interface, which connects a BTS to a BSC, is called the Abis interface. The Abis interface carries traffic and maintenance data, and is specified by GSM to be standardized for all manufacturers. The Abis interface allows control of the radio equipment and radio frequency allocation in the BTS.

- **A:** The A interface uses an SS7 protocol between the MSC and the BSS, as well as network message between the individual subscribers and the MSC. The A interface manages the allocation of suitable radio resources to the MSs and mobility management.

- SS7: SS7 protocol, called the Signalling Correction Control Part (SCCP), supports communication between the MSC and other public network.

2.3 The GSM radio interface

The radio interface, one of the most important interfaces of the GSM system, is the interface between the mobile stations and the BSS [11].

GSM uses a combination of TDMA scheme and FDMA scheme to provide base stations with simultaneous access to multiple users. Two frequency bands of 25 MHz apart have been reserved for GSM operation: 890-915 MHz for uplink, 935-960 MHz for downlink. The total number of available channels within 25 MHz is 125 (assuming no guard band) single carrier channels of 200 KHz width. In practical implementations, a guard band of 100 kHz is provided at the upper and lower end of the GSM spectrum, and only 124 channels are implemented. There are a total of 992 traffic channels within GSM from each radio channel consists of 8 timeslots. GSM mainly provides telephone services at the rate of 13.4 kbps, and supports up to 9.6kbps data services, and supplementary ISDN services such as Short Messaging Services (SMS). A GSM mobile station uses the same time slots in the uplink as in the downlink. A channel of GSM is permanently allocated for a particular user during the entire call period. A detailed GSM air interface specification is as shown in the Table2.1 [12]:

Table 2.1 GSM Air Interface Specifications

Parameter	Specification
Reverse Channel Frequency	890-915 MHz
Forward Channel Frequency	935-960 MHz
Tx/Rx Frequency Spacing Tx/Rx Time Slot Spacing	45 MHz 3 Time slots
Modulation Data Rate	270.833333kbps
Frame Period	4.615 ms
Users per frame (Full Rate)	8
Time slot Period	576.9μs
Bit Period	3.692μs
Modulation	0.3 GMSK
Channel Spacing	200 kHz
Interleaving (max. delay)	40 ms
Voice Coder Bit Rate	13.4 kbps

2.4 GSM Frame Structure

Each of 25 MHz widths is divided into 124 single carrier channels of 200 kHz. By dividing each of the 200 kHz frequency channels, we have 8 TDMA channels corresponding with 8 time slots (bursts). The 8 time slots in a carrier form a GSM frame. Each time slot of GSM frame lasts for a duration of 156.25 bit times, which equals to 15/26 ms = 576.9 μs; thus a frame takes 4.615 ms. A GSM mobile station use the same time slot in uplink as in the downlink. On top of the GSM physical channels, several logical channels are defined to perform different functions.

As shown in the figure 2.2 TDMA/GSM frame structure, traffic channels are defined using a group of 26 TDMA frames called a 26-Multiframe. In this 26-multiframe structure, the traffic channels for the downlink and uplink are separated by 3 bursts. However, a 51-Multiframe of control signaling includes 51 TDMA frames (235.365ms). The frame rate is 270.833 kbps/1250 bits/frame. The 13th or 26th frame is not used for traffic, but for control purposes. The normal speech frames are grouped into multi-frames, and multi-frames are grouped into super-frames.

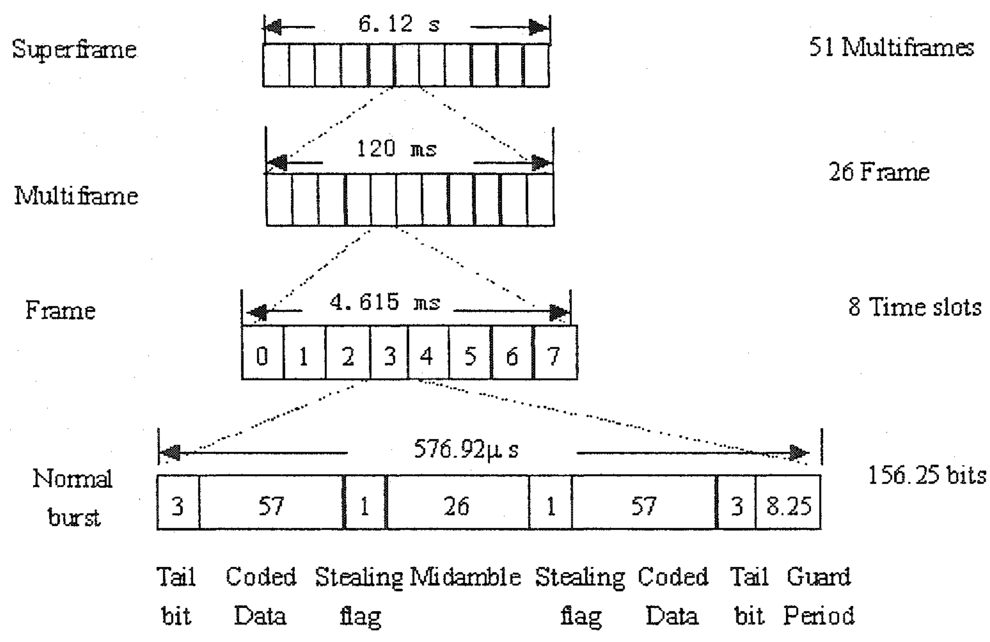


Figure 2.2 A TDMA/GSM frame structure

2.5 Logical Channels

There are two types of GSM logical channels, namely: traffic channel (TCH) and control channel. TCH carries digitally encoded user speech or data and has identical functions and formats on both the foreword & reverse link. Control channels transmit signaling and synchronizing commands between Mobile Station (MS) and Base Station.

A. Traffic Channels (TCH)

GSM traffic channels can transmit digital speech or user data at full rate or half-rate. There are 7 traffic channels, which have 4 channels carried at 22.8 kbps for full-rate TCH, and 3 channels carried at 11.4 kbps for half-rate TCH.

Full-rate TCH

- Full-rate Speech Channel (TCH/FS): It carries user speech digitized at 13 kbps.
- Full-rate Data Channel for 9600 bps (TCH/F9.6): it carries user data at 9600 bps.
- Full-rate Data Channel for 4800 bps (TCH/F4.8): it carries user data at 4800 bps.
- Full-rate Data Channel for 2400 bps (TCH/F2.4): it carries user data at 2400 bps.

Half-rate TCH

- Half-rate Speech Channel (TCH/HS): It carries user speech digitized at 6.5 kbps.
- Half-rate Data Channel for 4800 bps (TCH/H4.8): it carries user data at 4800 bps.
- Half-rate Data Channel for 2400 bps (TCH/H2.4): it carries user data at 2400 bps.

B. GSM Control Channel (CCH)

The CCH is divided into three major control channels: broadcast channel (BCH), common control channel (CCCH), and dedicated control channel (DCCH). Only random access channel is a reverse link channel within TS0, all other BCCH and CCCH almost are forward control channels occupying TS0, as shown in Figure 2.3.

Broadcast Channel

- Broadcast Control Channel (BCCH): To broadcast information and operating characteristics of the cell within TS0.

- Frequency Correction Channel (FCCH): To synchronize MS internal frequency standard to the exact frequency of the base station within TS0.
- Synchronization Channel (SCH): To synchronize MS frame with BS within TS0.

Control Multiframe = 51 TDMA Frames (235)

0	1	2	3	4	5	6	7	9	10	11	13		20	21	22		39	40		49	50
F	S	B	B	B	B	C	C	C	F	S	C		F	S	C		C	F		C	I

F: FCCH burst (BCH) S: SCH burst (BCH) B: BCCH burst (BCH)

C: PCH/AGCH burst (CCCH) I: Idle

(a)

0	1	2	3	4	....	45	46	47	48	49	50
R	R	R	R	R		R	R	R	R	R	R

Control Multiframe = 51 TDMA Frames (235 ms)

R: Reverse RACH burst (CCCH)

(b)

Figure 2.3 The control channel multiframe at TS0: (a) for Forward link, (b) For reverse link

Common Control Channels (CCCH)

- Random Access Control Channel (RACCH): To originate a call, and to acknowledge a page from the PCH by MS.

- Access Grant Control Channel (AGCCH): To instruct MS to operate in particular physical channel (time slot and ARFCN) with a DCCH.
- Paging Control Channel (PCCH): To provide paging signals from BS to all MS in the cell, and notifies a specific mobile of an incoming call which originates from the PSTN.

Dedicated Control Channel (DCCH)

- Stand-alone Dedicated Control Channels (SDCCH): To carry signalling data following the connection of MS with the BS before a TCH is assigned.
- Slow Associated Control Channel (SACCH): It is always associated with a traffic channel or a SDCCH, and maps onto the same physical channel.
- Fast Associated Control Channels (FACCH): To carry urgent messages, and to contain essentially the same type of information as the SDCCH.

2.6 Signal Processing in GSM

Figure 2.4 illustrates transmitter of the GSM operations from speech input to radio output. The receiver runs the oppose direction in the signal processing.

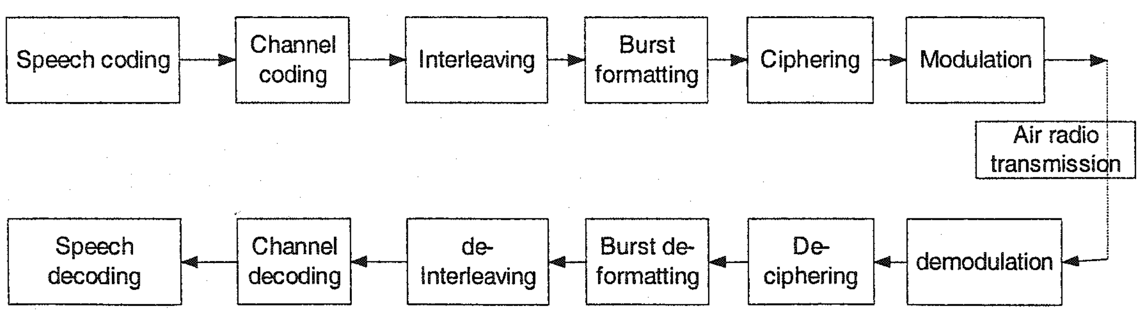


Figure 2.4 GSM operations from input speech to output speech

Speech Coding

The GSM speech coder is based on the Residually Excited Linear Predictive Coder (RELP). The coder provides 260 bits for each 20 ms blocks of speech, which yields a bit rate of 13 kbps. By incorporating a voice activity detector (VAD) in the speech coder, GSM systems operate in a discontinuous transmission mode (DTX).

Channel Coding: A half-rate convolution encoder is used to encode the data.

Interleaving: A total of 456 encoded bits are broken into 8*57 bits sub-blocks, which are being spread over four consecutive TCH time slots.

Ciphering: It modifies the contents of 8 interleaved blocks through the use of encryption techniques known only to the particular MS and BTS.

Burst formatting: It adds binary data to the ciphered blocks to help synchronization.

Modulation: GSM is modulated at 0.3 GMSK, where 0.3 describes the 3 dB bandwidth of the Gaussian pulse-shaping filter in relation to the bit rate.

Frequency Hopping: In order to avoid important differences in the quality of the channels, the slow frequency hopping may be implemented to combat the multipath fading or the effects of co-channel interference. It changes the frequency with every TDMA frame. The frequency-hopping algorithm selected is sent through the broadcast control channels.

Chapter 3
Overview of the GPRS &EGPRS Networks

3.1 Introduction to GPRS

The phase 2+ of GSM is called the General Packet Radio Service (GPRS) and the Enhanced GPRS (EGPRS). GPRS is also called phase one of EGPRS. It is a packet switching based system added on the existing GSM. GPRS can be implemented in existing GSM systems using the same cell structure. It is embedded in the physical channel of GSM frame structure. As a consequence, only minor changes will be required to introduce GPRS in an existing GSM network.

3.2 GPRS functional groups

GPRS at function is categorized into 6 groups: network access, packet routing and transfer, mobility management, logical link management, radio resource management, and network management.

A. GPRS **network access**: It is responsible for the standard point-to-point data transfer and anonymous access. It includes 6 functions: registration; authentication and authorization; admission control to determine if the radio and network resources be used for communication of an MS; message screening, which filters out unsought messages; packet terminal adaptation, which automatically adjusts data transmission across the GPRS network; and billing information collection for packet transmission in GPRS and external networks.

B. **Packet routing and transfer**: It is responsible for routing the data between an MS and the destination through the serving and gateway GPRS support Nodes (GSNs). There are 5 main functions: Relay function that is used to forward packets between an MS and a SGSN by BSS, and also used to forward packets between a BSS and a serving or gateway GSN by SGSN; Routing function that is used to determines the destinations of packets; Address translation and mapping function that is used to convert a GPRS network address to an external data network address and vice versa; Encapsulation and tunnelling functions which encapsulate packets at the source of a tunnel, deliver the packets through the tunnel and decapsulate them at the destination; and domain name service functions that is used to resolve logical GSN names to their IP addresses.

C. **Logical link management:** It is responsible for maintaining the communication channels between an MS and the GSM network across the radio interface. It consists of logical link establishment, logical link maintenance and logical link release.

D. **Radio resource management**: It is used to assign radio resources, and to maintain radio communication paths. Its functions include:

1) **Um management:** To determine the amount of radio resources to be allocated for GPRS usage.

2) **Cell selection:** To select the optimal cell for radio communications of an MS

3) **Um-tranx:** To provide packet data transfer capability, such as medium access control, packet multiplexing, packet discrimination, error detection and correction, and flow control across the radio interface between the MS and the BSS.

4) **Path management**: To maintain the transfer paths between the BSS and the SGSNs.

E. **Mobility management:** It is responsible to track the current location of an MS.

F. **Network management**: It is used to provide mechanisms for supporting OA&M functions related to GPRS.

3.3 GPRS Network Architecture

In order to integrate GPRS into the existing GSM architecture, modifications of the GSM network are required. The MS, BSS, BSC/VLR and HLR in GSM network are modified for GPRS networks. Some of the nodes already implemented in current GSM systems can be shared between GPRS and GSM. A new class of network nodes, called GPRS support nodes (GSN), has been introduced [8]. GSNs are responsible for the delivering and routing of data packets between the mobile stations and the external packet data networks (PDN). GSNs are divided into two new node types, called Serving GPRS Support Node (SGSN) and Gateway GPRS Support Node (GGSN) [13].

Figure 3.1 illustrates architecture of GPRS. A GPRS MS consists of a Mobile Terminal (MT) and a Terminal Equipment (TE). A TE can be a computer attached to the MT. The MT is equipped with software for GPRS functionality. The user profile, the current SGSN address, and the PDP address are stored in HLR for each GPRS user in the Public Land Mobile Network (PLMN). In addition, the MSC/VLR may be modified with functions that allow efficient coordination between packet switched (GPRS) and circuit switched (GSM) services.

A SGSN is at the same hierarchical level as the mobile switching centre (MSC). It is responsible to keep track of the location of the GPRS mobiles, and the delivery of data packets from and to the mobile stations within its service area. It also performs security checking, packet routing and transfer, mobility management, logical link management, and authentication and charging functions. The location information and user profiles of all GPRS registered users are stored in location register of the SGSN.

A GGSN is used as inter-working node between the PLMN and external packet data networks (PDN), and also as an interface by routing data between the GPRS backbone network and the external packet data networks. In GGSN, the GPRS packets coming from the SGSN are converted into appropriate packet data protocol (PDP) format, and sent out on the corresponding packet data network. In the opposite direction, GGSN converts PDP addresses of incoming data packets into the GSM address of the destination user. The readdressed packets are sent to the responsible SGSN. For this purpose, the current SGSN address of the user and his or her profile are stored in its location register by GGSN. The GGSN also performs authentication and charging functions.

Figure3.1 also shows the interfaces between the new network nodes and the GSM network. These interfaces are defined by ETSI in [24]. The communications between the MS and the BSS are via the air Um air interface. The BSS and the SGSN are connected by the Gb interface using frame relay. User data and signalling data are transmitted between the GSNs via the Gn and Gp interfaces. The Gn interface will be used when SGSN and GGSN are located in the same PLMN, whereas the Gp interface will be used if they are located in different PLMNs. The Gi interface connects the PLMN with external public or private PDNs, such as the Internet or corporate intranets. The existing GSM D interface is used to connect between the HLR and VLR. Interfaces A, Gs, Gr, Gc, and D are used for signalling, without involving user data transmission in GPRS. Note that the A interface is used for both signalling and voice transmission in GSM. Interfaces Um, Gb, Gn, Gp, and Gi are used for both signalling and transmission in GPRS.

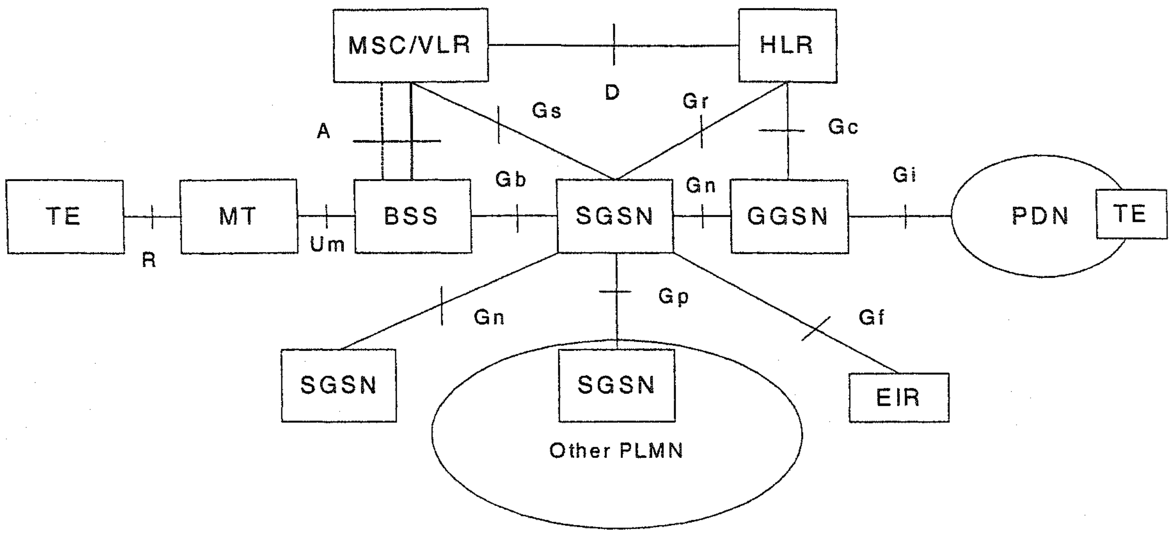


Figure 3.1 illustrate Architecture of GPRS with nodes and interfaces

3.4 Protocol Architecture

Figure3.2 illustrates the protocol architecture of the GPRS transmission plane [25], which consists of a layered protocol structure for user information transfer and its associated signalling, e.g., flow control, error detection, and error correction. GPRS-specific protocols include SNDCP, LLC, RLC, MAC, BSSGP, BSSAP+, and GTP. GMM/SM and MAP are modified to accommodate GPRS.TCAP, SCCP and MTP are SS7 layers. The other protocols are standard data protocols [38-40].

GPRS backbone: The GPRS Tunnelling Protocol (GTP) tunnels the user data packets and related signalling information between the GPRS support nodes (GSNs) [37]. In the transmission plane, GTP employs a tunnel mechanism to transfer user data packets. In the signalling plane, GTP specifies a tunnel control and management protocol. The signalling is used to create, modify, and delete tunnels.

In the GPRS backbone, user data packets are encapsulated through IP/X.25 - over GTP - over UDP/TCP - over IP transport architecture.

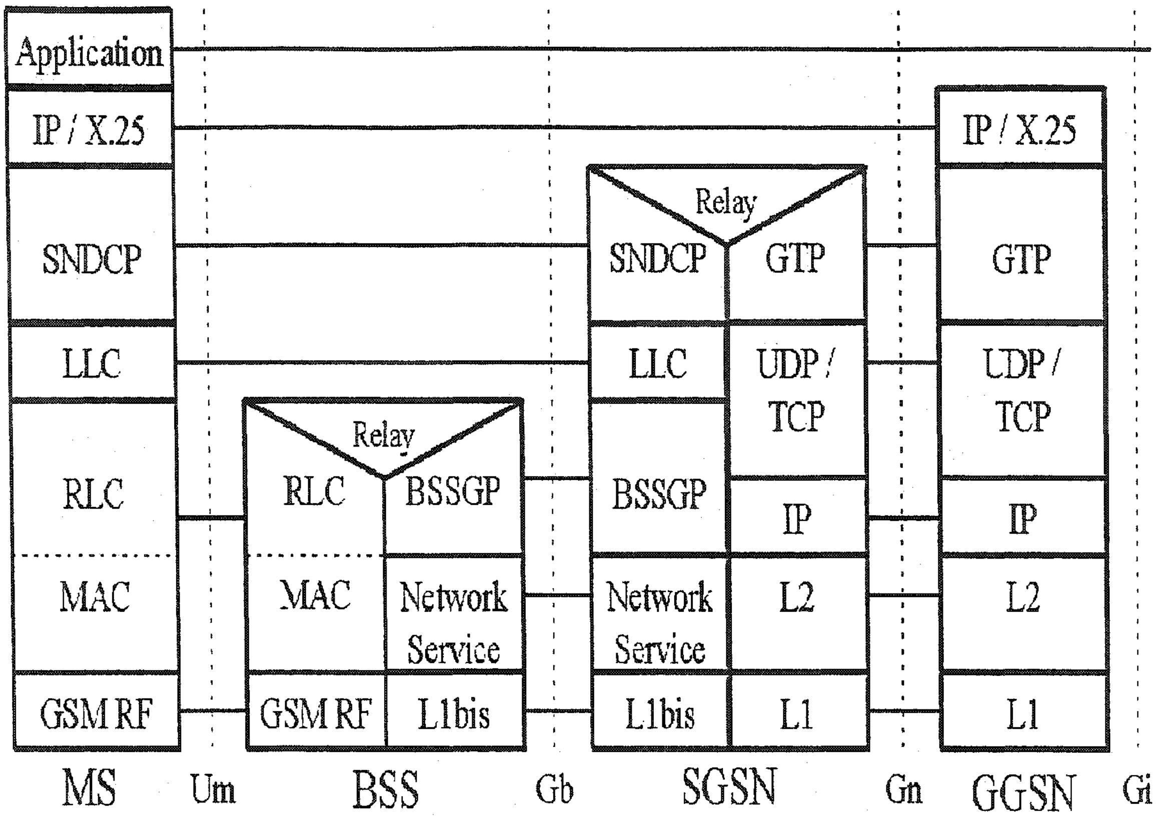


Figure 3.2 GPRS Transmission Plane.

The GPRS protocols for transmission planes include:

- **GTP (GPRS Tunneling Protocol)** [26]: To be used to tunnel user data and signaling between GSNs in the GPRS backbone network.
- **TCP**: To be used to carry GTP packet data units (PDUs) for protocols that need a reliable data link (e.g., X.25)
- **UDP (User Datagram Protocol)**: To be used to carry GTP PDUs for protocols that do not need a reliable data link (e.g., IP)
- **IP**: It is a GPRS backbone network protocol used for routing user data and control signaling.
- **SNDCP (SubNetwork Dependent Convergence Protocol)** [27]: To be used to map

network-level characteristics onto the characteristics of the underlying network. It is used to transfer data packets between SGSN and MS. Its functionality includes:

- Multiplex of several connections of the network layer onto one virtual logical connection of the underlying LLC layer.
- Compression and decompression of user data and redundant header information.

- **LLC (Logical Link Control)** [28]: To provide a highly reliable ciphered logical link.
  - Independent of the underlying radio interface protocols.
- **RLC/MAC** layer [29] at the air interface includes two functions:
  - **RLC (Radio Link Control)**: To provide a radio-solution-dependent reliable link between the MS and the BSS. It segments and reassembles LLC frames into RLC data blocks and ARQ of uncorrectable codeword.
  - **MAC ((Medium Access Control)**: To control the access signaling (request and grant) procedures for the radio channel, and mapping of LLC frames onto the GSM physical channel.
- **Relay**:
  - In BSS, relays LLC PDUs between Um and Gb interfaces.
  - In SGSN, relays PDP PDUs between Gb and Gn interfaces.
- **BSSGP (Base Station System GPRS Protocol)**[27]: To be used to convey routing- and QoS-related information between BSS and SGSN.
- **NS (Network Service)**: To transport BSSGP PDUs and is based on the Frame Relay connection between BSS and SGSN.

The protocol architecture of the signalling plane is defined in [25]. It includes:

- **GMM/SM (GPRS Mobility Management and Session Management protocol)**: To be used to perform functions such as GPRS attach/detach, security functions, PDP context activation, and routing area updating between MS and SGSN above LLC layer.

- **MAP (Mobile Application Part):** To be used between SGSN and HLR as well as between SGSN and EIR. It transports the signalling information related to location updating, routing information, user profiles, and handovers.
- **SS7(Signalling System #7)** layers: include **TCAP** (Transaction Capabilities Application Part), **SCCP** (Signalling Connection Control Part), and **MTP** layers.

3.5 Services

3.5.1 Bearer services and supplement services

The bearer services of GPRS support end-to-end packet switched data transfer. GPRS services are divided into two categories: Point-to-Point (PTP) and Point-to-Multipoint (PTM) services.

The PTP service [30] provides transfer of data packets between two users. Possible PTP services include: data base access and information retrieval; the Internet; messaging and conversational services from user to user; credit card validation, etc. It is offered in both connectionless mode (PTP-CLNS: PTP connectionless network service) and connection-oriented mode (PTP-CONS: PTP connection-oriented network service).

The PTM service provides transfer of data packets from one user to multiple users. PTM services include: unidirectional distribution of information such as news and weather reports, conferencing services between multiple users, etc. There exist two kinds of PTM services [31]:

- **Multicast service PTM-M**: Data packets are broadcast in a certain geographical area. A group identifier indicates whether the packets are intended for all users or for a group of users.
- **Group call service PTM-G**: Data packets are addressed to a group of users (PTM group) and are sent out in geographical areas where the group members are currently located.

The PTM service can send SMS messages over GPRS. In addition, it can also implement supplementary services, such as call forwarding unconditional (CFU), call forwarding on mobile subscriber not reachable (CFNRc), and closed user group (CUG).

3.5.2 Simultaneous usage of packet switched and circuit switched services

Both circuit switched services (speech and data) and GPRS services can be used simultaneously in a GSM/EGPRS network. Three MS operation modes were introduced in [30]:

- **Class A**: A mobile station is allowed to use simultaneous operation of GPRS and conventional GSM services.
- **Class B**: A mobile station is able to register with the network for both GPRS and conventional GSM services simultaneously. In contrast to an MS of class A, MS can only choose dynamically one of the two services at a time.
- **Class C**: A mobile station only support packet-switched data service. An exception is SMS messages, which can be received and sent at any time.

3.5.3 Quality of service

In order to satisfy the service requirements for various subscribers, UMTS has proposed four different traffic classes [14]: the background, conversational, interactive, and streaming classes respectively. According to the requirements of QoS, they are also being divided into two main classes: Real Time services (e.g., voice, video conferencing, etc.), and Non Real Time services (e.g., database applications, web browsing, email, streaming video and sound, etc.). The Quality of Service (QoS) requirements of typical mobile packet data applications are very diverse. Support of different QoS classes, which can be specified for each individual session, is therefore an important feature.

GPRS allows defining QoS profiles using the parameters precedence, reliability, delay, and throughput classes [30]:

- **Precedence class:** To specify three-transmission priority levels (i.e., high, normal, and low). During congestion, the packets with lower priorities are discarded.
- **Reliability class**: To indicate the transmission characteristics required by an application. Three reliability classes are introduced, which guarantee certain maximum values for the probability of loss, duplication, mis-sequencing, and corruption (an undetected error) of packets.
- **Delay class**: To specify maximum values for the mean delay and the 95-percentile delay. The latter is the maximum delay guaranteed in 95 percent of all transfers. The delay is defined as the end-to-end transmission time between two communicating MSs or between a MS and the Gi interface to an external packet data network. This includes all delays within the GPRS network.
- **Throughput class**: To specify the expected maximum data transmission rate and the mean bit rate. Using these QoS classes, QoS profiles can be negotiated

between the mobile user and the network for each session, depending on the QoS demand and the current available resources. The billing of the service is then based on the transmitted data volume, the type of service, and the chosen QoS profile.

Table 3.1 lists these four traffic classes with their application, fundamental characteristics, and QoS requirements.

Table 3.1 The Four (E) GPRS Traffic Classes

Traffic class	Background class	Conversational class	Interactive class	Streaming class
Application	-Background download of e-mail -File transfer	Voice over IP	-Web browsing -Database retrieval	-Real-time video - Non real-time video -Video conferencing
Fundamental characteristics	-Preserve payload content	-Preserve time relation between entities making up the stream -Conversation based on perception -Real-time	-Bounded response time -Preserve the payload content	-Preserve time relation between entities making up the stream -Real-time -Non real-time
Relevant QoS requirements	-Low BER	-Low jitter -Low delay	-Low jitter	-Round delay time -Low BER

3.6 Mobility Management

3.6.1 Attachment and Detachment Procedure

The GPRS attach procedure establishes a logical link between the MS and the SGSN. Before a mobile station can use GPRS services, it must register with an SGSN of the GPRS network. To attach to the network, the MS provides its identity and indicates which type of attach procedure is to be performed. Then, the system copies the user profile from the HLR to the SGSN, and assigns a packet temporary mobile subscriber identity (P-TMSI) to the user. This procedure is called as GPRS attach. The GPRS detach procedure is disconnection from the GPRS network, and is initiated by the MS or by the network (SGSN or HLR).

3.6.2 Session Management

In order to exchange data packets with external PDNs after a successful GPRS attach, a MS must apply for one or more addresses used in the PDN, e.g., for an IP address in case the PDN is an IP network. This address is called PDP address (Packet Data Protocol address). For each session, a PDP context is created, which describes the characteristics of the session. It contains the PDP type, the PDP address assigned to the MS, the requested QoS, and the address of a GGSN that serves as the access point to the PDN. This context is stored in the MS, the SGSN and the GGSN. With an active PDP context, the MS is "visible" for the external PDN and is able to send and receive data packets. The mapping between the two addresses, PDP and IMSI, enables the GGSN to transfer data packets between PDN and MS. A user may have several simultaneous PDP contexts active at a given time.

The assignment of the PDP address can be static or dynamic. In the static case, the network operator of the user's home-PLMN permanently assigns a PDP address to the user. In dynamic PDP address allocation, a PDP address is assigned to the user upon activation of a PDP context; GGSN is responsible for the assignment and the activation/deactivation of the PDP addresses. Figure3.3 illustrates the PDP context activation procedure.

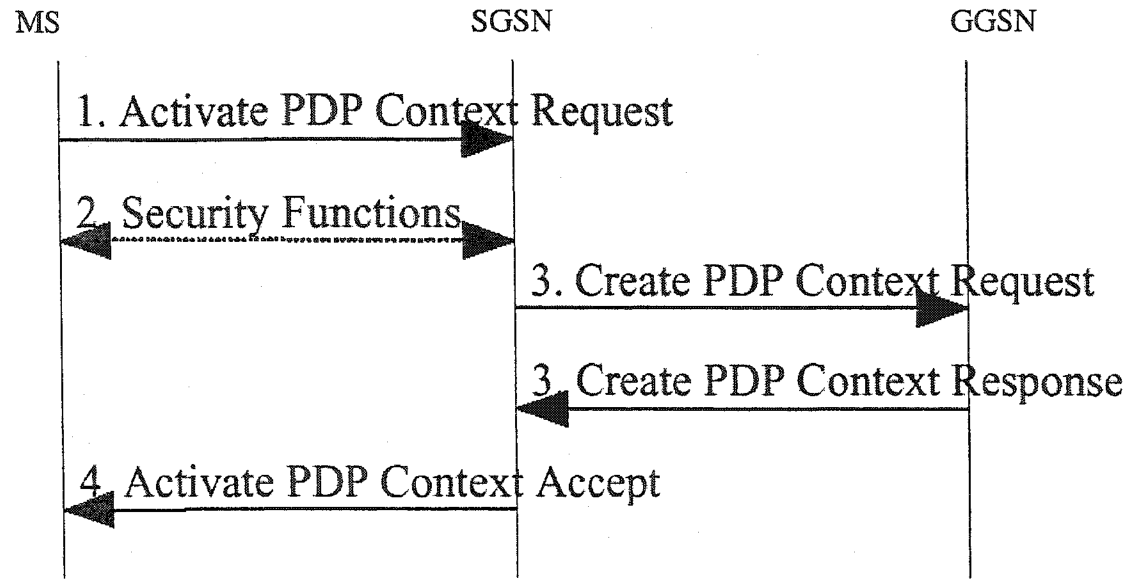
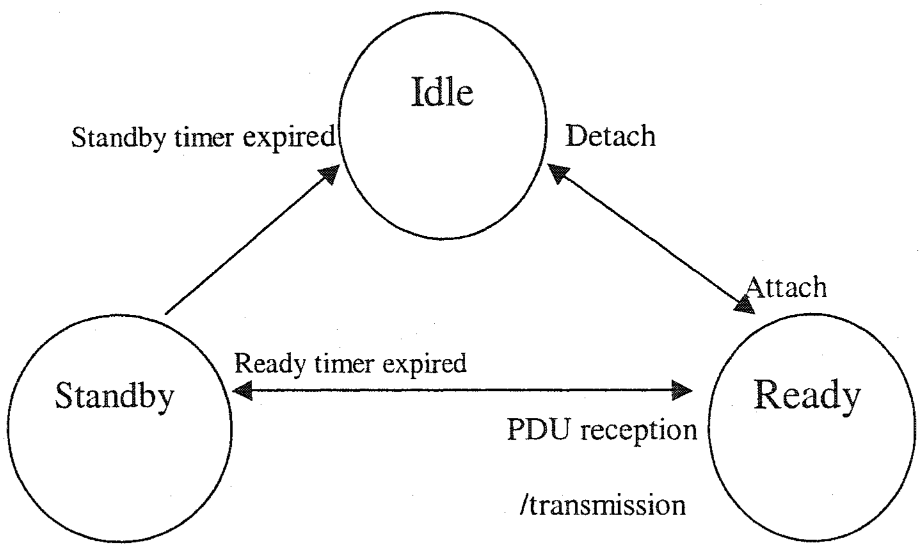


Figure 3.3 PDP context activation

3.6.3 Location Management

Location management is mainly responsible to keep track of the user's current location for routing incoming packets to his or her MS. For this purpose, the MS has to send location updating messages to its current SGSN. In order to know the MS in accuracy of the routing area, and to decrease the consumption of uplink radio capacity and battery power for mobility management, a good location management strategy must be a compromise. For this reason, a state model shown in Figure 3.4 has been defined for location management in GPRS [25]. There are 3 mobility management (MM) states related to a GPRS subscriber, which each state describes the level of functionality and information allocated; the location update frequency is dependent on the state of the MS.

In IDLE state the MS is not reachable, and not yet attached to the GPRS mobility management (GMM), so that no location updating be performed. After successful GPRS attach, the MS gets into READY state, and informs its SGSN of every movement to a new cell. In ready state, the MS can receive and send data for all relevant service types. If the ready timer expires, the MS will move to the standby state, the MS is still attached to the GMM and is known in the accuracy of the routing area. In standby state, if the MS wants to send or receive data, a PDP context must be activated in advance. If the standby timer expires, the MM contexts in both the MS and SGSN independently return to the idle state. With a GPRS detach it may disconnect from the network and fall back to IDLE



state. All PDP contexts will be deleted.

Figure 3.4 State model of a GPRS mobile station

3.7 Air Interface – Physical Layer

3.7.1 Multiple Access and Radio Resource Management

EGPRS uses the same frequency bands and frame structure as GSM, but the channel allocation and multi-frames in EGPRS is different from the original GSM.

The recurrence of one particular time slot defines a physical channel. A GSM mobile station uses the same time slots in the uplink as in the downlink [32].

In GPRS, the channel assignment is different from the original GSM. A single mobile station can transmit on multiple time slots of the same TDMA frame. This results in a very flexible channel allocation: one to eight time slots per TDMA frame can be allocated for one mobile station. Moreover, uplink and downlink are allocated separately, which efficiently supports asymmetric data traffic (e.g., Web browsing) [41, 42].

In contrast to GSM, the channels in GPRS are only allocated when data packets are sent or received, and they are released after the transmission. For bursty traffic this results in a much more efficient usage of the scarce radio resources. With this principle, multiple users can share one physical channel.

A cell supporting GPRS may allocate physical channels for GPRS traffic. Such a physical channel is denoted as packet data channel (PDCH), which is taken from the common pool of all channels available in the cell. Thus, all GPRS and non-GPRS mobile stations located in this cell share the radio resources. The mapping of physical channels to either packet switched (GPRS) or circuit switched (conventional GSM) services can be performed dynamically (capacity on demand principle [33]), depending on the current traffic load, the priority of the service, and the multislot class. A load supervision procedure monitors the load of the PDCHs in the cell. According to the current demand, the number of channels allocated for GPRS (i.e., the number of PDCHs) can be changed. Physical channels not currently in use by conventional GSM can be allocated as PDCHs to increase the quality of service for GPRS. When there is a resource demand for services with higher priority, PDCHs can be de-allocated.

3.7.2 GPRS multiframe structure

The EGPRS multiframe is composed by 52 TDMA frames, instead of 51-frames like

GSM ones. A multiframe structure is required for mapping the logical channel to the physical channels. The mapping of logical channels onto physical channels has two components: in frequency and in time. A multiframe structure for PDCHs consisting of 52 TDMA frames is shown in Figure3.5 [34]. Four consecutive TDMA frames form one block (12 blocks, B0 – B11), two TDMA frames are reserved for transmission of the PTCCH, and the remaining two frames are idle frames. The mapping of the logical channels onto the blocks B0 – B11 of the multiframe can vary from block to block and is controlled by parameters that are broadcast on the PBCCH.

In [34], it is defined that a logical channel may use some time slots. Besides the 52-multiframe, which can be used by all logical GPRS channels, a 51-multiframe structure is used for PDCHs carrying only the logical channels PCCCH and PBCCH and no other logical channels.

Multiframe with 52 TDMA frames (240 ms)

B0	B1	B2	T	B3	B4	B5	X	B6	B7	B8	T	B9	B10	B11	X

B0-B11: Four frames for each   T: a frame for PTCCH   X: an idle frame

Figure 3.5. GPRS multiframe structure with 52 frames

3.7.3 Logical Channels in GPRS

Table 3.2 lists the packet data logical channels defined in GPRS [33]. GPRS uses similar logical channel structure as GSM. These logical channels are also being divided into 2 categories: traffic channels and signaling control channels. The packet data traffic channel (PDTCH) is employed for the transfer of user data. It is assigned to one mobile station (MS), and one MS can use several PDTCHs simultaneously. A series of logical channels are defined on top of the physical channels to perform a multiplicity of functions, e.g., signaling, broadcasting of general system information, synchronization, channel assignment, paging, or payload transport. The signalling control channels include packet broadcast control channel, packet common control channel, and packet-dedicated control channels.

The packet broadcast control channel (**PBCCH**) is a unidirectional point-to-multipoint signaling channel from the base station subsystem (BSS) to all GPRS mobile stations for a cell. It is used by the BSS to broadcast specific information about the organization of the GPRS radio network, and important system information about circuit switched services, so that a GSM/GPRS mobile station does not need to listen to the broadcast control channel (BCCH).

The packet common control channel (PCCCH) is a bi-directional point-to-multipoint signaling channel. It is used by MSs to initiate packet transmission or respond to paging messages. On this channel MSs transmit access bursts with long guard times. On receiving access bursts, the BSS assigns a timing advance to each terminal. Packet random access channel is the only uplink PCCCH. Other PCCCHs are downlink, sent from the BTS to the MS. It consists of four sub-channels:

- **Packet Random Access Channel (PRACH).** Used by the mobile to request one or more PDTCH, and to initiate uplink transfer for data or signaling.
- **Packet Access Grant Channel (PAGCH).** Used in the packet transfer

establishment phase for resource assignment, which can allocate one or more PDTCH to a mobile station.

- **Packet Paging Channel (PPCH)** Used by the BSS to find out the location of a mobile station (paging) prior to downlink packet transmission.
- **Packet Notification Channel (PNCH).** Used to send a point-to-multipoint multicast (PTM-M) notification to a group of MSs prior to a PTM-M packet transfer.

The following packet-dedicated control channels (PDCCH) are defined in GPRS:

- **Packet Associated Control Channel (PACCH).** Conveys signalling information, such as power control, resource assignment, and reassignment information. The PACCH shares resources via PDTCHs.
- **Packet timing advance control channel (PTCCH).** Used for adaptive frame synchronization.

Table 3.2 Logical Channels in GPRS

Group	Channel	Function	Direction
Packet data traffic channel	PDTCH	Data traffic	Downlink/uplink
Packet broadcast control channel	PBCCH	Broadcast control	Downlink
Packet common control channel (PCCCH)	PRACH PAGCH PPCH PNCH	Random access Access grant Paging Notification	Uplink Downlink Downlink Downlink
Packet dedicated control channels (PDCCH)	PACCH PTCCH	Associated control Timing advance Control	Downlink/uplink Downlink/uplink

3.7.4 GPRS channel management

There are two concepts for GPRS channel management: master-slave and capacity-on-demand.

In the master-slave concept, a master PDCH accommodates PCCCHs to carry all necessary control signalling for initiating packet transfer. Other PDCHs serve as slaves for user data transfer (PDTCH) and for dedicated signalling.

In the capacity-on-demand concept, PDCHs are dynamically allocated based on actual amount of packet transfers. Also, the number of allocated PDCHs in a cell can be increased or decreased according to traffic changes.

GPRS performs a fast release of the PDCH to share the pool of radio resources for both packet- and circuit-switched services.

3.8 Channel Coding Schemes in GPRS

The channel coding schemes in GPRS are quite similar to the one employed in GSM. An outer block coding, an inner convolutional coding, and an interleaving scheme is used. Four different coding schemes are defined in [35] to be able to adaptively react to current channel quality. Their parameters are listed in Table 3.3. The first coding scheme is used in GSM: 1/2-rate convolutional coding and a 40-bit fire code are applied. The second and third schemes are punctured versions of the first one with rates of 2/3 and 3/4, respectively. The fourth coding scheme does not use a convolutional coder. The latter three schemes use a 16-bit frame check sequence for error detection. In order to speed up decoding of USF, schemes CS-2 to CS-4 generate a 12-bit block USF code word. For scheme CS-1, the entire block is coded, and USF must be decoded as part of the data. The coding scheme is indicated by the GSM stealing bits of the four consecutive bursts that belong to one block using an 8-bit block code with a Hamming distance of 5.

Table 3.3 Channel Coding Schemes for logical traffic channels in GPRS

MCS	User rate [Kbps]	Code Rate	Header Code Rate
CS-4 (GMSK)	21.4	1.0	0.51
CS-3 --	15.6	~3/4	0.51
CS-2 --	13.4	~2/3	0.51
CS-1 --	9.05	1/2	0.51

For the coding of the traffic channel (PDTCH), one of the four coding schemes (CS-1 to CS-4) is chosen, depending on the quality of the channel. Under very bad channel conditions, the coding scheme of CS-1 with a data rate of 9.05 kbit/s per GSM time slot is chosen for a very reliable coding. Under good channel conditions, the CS-4 without convolutional coding is chosen to achieve a data rate of 21.4 kbit/s per time slot. With eight time slots, a maximum data rate of 171.2 kbit/s will be obtained. In practice, multiple users can share the same time slot, resulting in a much lower bit rate available to the individual user. After encoding, the codeword is put into a block interleaver of depth 4. On the receiver side, the codeword is de-interleaved. The decoding is performed using the well known Viterbi Algorithm (see, e.g., [36]).

3.9 Enhanced Data rate for GSM Evolution (EDGE)

3.9.1 Introduction to EDGE

With GSM radio technology, GPRS provides only limited data capacity. To increase the GSM data rate, Enhanced Data Rate for GSM Evolution (EDGE), called as phase 2 of EGPRS, was introduced with the same frame structure, carrier spacing (200 KHz), symbol rate (271 Ksymb/s), burst format and spectrum as GSM. Phase 2 of Enhanced GPRS (EGPRS) based on EDGE continues to use the GPRS core network. EGPRS provides user data rates (up to 470 Kbps for indoor and 144 Kbps for outdoor) 2-3 times higher than GPRS, and spectrum efficiency 2-6 times higher than GPRS. These goals are accomplished mainly by using a higher-level modulation-- 8-phase shift keying (8-PSK), in addition to the traditional GMSK, and by adapting the user rate based on channel quality. Thus, EGPRS can reuse GSM sits and frequency plan.

The Phase 2 of EGPRS needs to further enhance the air interface and the core GPRS network. The functionality of the SGSN and GGSN needs to be enhanced to be E-SGSN and E-GGSN respectively. There are many new features introduced to EGPRS that result in higher data rates at link level as well as in spectral efficiency at system level. The EGPRS data rates at link level are in the range of 8.8-59.2Kbps per time slot. This is achieved through 9 coding schemes.

EDGE combines link adaptation and incremental redundancy. Link adaptation (LA) provides dynamic switching between coding and modulation schemes based on link-quality measurements. However, in addition to LA, a more intelligent link quality control scheme, termed as incremental redundancy (IR), which enhances the performance at both link level and system level, increases robustness for retransmission in EDGE.

An LLC packet data unit in EGPRS is broken into 20ms RLC data blocks in accordance with link Adaptation (LA). A RLC data block contains the user data, which defines the effective data rate per time slot. The RLC/MAC headers are then added to the user data to form a RLC radio block.

Table 3.4 provides the details of the code rate together with the user data rate per time slot for each coding scheme. In an EGPRS network, conventional circuit switched services and EGPRS service can be used in parallel. Three classes of mobile station in the family have mentioned in section 4.5.2.

Table 3.4 EDGE Data and Code Rates [17]

MCS	Channel rate [Kbps]	Code Rate	Header Code Rate	Family
9 (8-PSK)	59.2	1	0.36	A
8 --	54.4	0.92	0.36	A
7 --	44.8	0.76	0.36	B
6 --	29.6	0.49	1/3	A
5 --	22.4	0.37	1/3	B
4 (GMSK)	17.6	1.0	0.51	C
3 --	14.8	0.85	0.51	A
2 --	11.2	0.66	0.51	B
1 --	8.8	0.53	0.51	C

3.9.2 Fast Packet Control Channels

The phase two of EGPRS needs to further enhance the core EGPRS network. By enhancing the GPRS RLC/MAC design to efficiently support in-session access, a set of common control channels are needed to provide the additional capabilities for ongoing calls. These channels are similar to the common control channels required for call set-up with one vital difference: they are designed for in-session control. While in-session control has a more stringent delay requirement than session set-up control, it also has smaller signalling overhead, which makes it feasible to meet these delay requirements.

To accomplish in-session control purpose, two fast control channels are introduced.

Fast packet access channel (F-PACH) for the uplink

The structure of the F-PACH is similar to that of the PRACH in GPRS, in the sense that messages are transmitted in individual bursts. The difference between the two is that since the F-PACH is used exclusively for ongoing calls, the fast packet channel requesting message carried in F-PACH contains information on the specific TBF being referenced (i.e., the uplink TFI assigned to the MS and other relevant information). Based on this information, the base station can uniquely identify the MS and its specific application, and therefore quickly assign the necessary uplink resource.

The F-PACH can be used either as a fast packet random access channel (F-PRACH) for random access, or as a fast packet dedicated access channel (F-PDACH) for dedicated access. The F-PRACH and F-PDACH can be time multiplexed on the same physical channel (e.g., time slot 0) of some selected carriers. The characteristics of these channels are as follow:

F-PRACH is designed to transmit single burst fast contention access messages for fast random access, which carries the TFI and other identifying information of the requesting application.

F-PDACH is designed for fast dedicated access. It permits the mobile to use contention-free access and may be useful for applications that do not permit any delay variability.

a. Fast packet control channel (F-PCCH) for the downlink

The F-PCCH serves two major functions: to transmit access grant and to poll messages to specific mobiles. Thus, the F-PCCH is split into two logical channels: an F-PAGCH and an F-PPCH. These two channels can be time multiplexed on the same physical channel located on specific time slots (e.g.. time slot 0) of some selected carriers. Each pair of F-PACH and F-PAGCH/F-PPCH may carry the fast uplink access request, access grant, polling, and polling response messages for a set of carrier frequencies.

Fast packet access grant channel (F-PAGCH)

The F-PAGCH is used to respond to access requests received on the F-PACH. This response is a typical assignment message that specifies the channels, USFs, and other parameters for a set of MSs.

Fast packet polling channel (F-PPCH)

It is used to poll different mobiles.If an MS has an ongoing downlink data transfer, it is possible to use the PACCH to transmit control messages to the MS. Otherwise; the BBS can use the downlink F-PCCH to communicate with the MS.

3.9.3 Multiplexing Capability

The EGPRS RLC/MAC layer is designed to efficiently support multiple data streams on the same packet data traffic channel (PDTCH), and a given data stream on multiple channel [19]. Data transfer in EGPRS is accomplished by using an entitled temporary block flow (TBF). 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. Each TBF is identified by a temporary flow identifier (TFI). 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. RLC blocks destined to different MSs are distinguished by their attached TFIs. After completion of the data transfer, the TBF is terminated and the TFI is released.

Downlink multiplexing of multiple data streams on the same PDTCH is accomplished by assigning each data transfer a unique TFI. Each MS listens to its set of assigned downlink channels and only accepts RLC blocks with its TFI.

Uplink multiplexing is accomplished by assigning each data transfer a set of channels and a unique uplink state flag (USF) for each of these channels. Up to 8 mobiles may be assigned to the same uplink traffic channel but with different USFs, which is 3 bits long. An MS listens to all the downlink traffic channels that are paired with the uplink channels assigned. If its USF appears in the downlink channel, the MS uses the corresponding uplink channel in the next logical frame.

3.9.4 Multiple Access Procedure

The multiple access procedure is shown in Figure 3.6 [18]:

- When a new call starts, the MS sends an access request to the BSS over the normal PRACH. Using the initial access procedure (one- or two-phase), it establishes a TBF and obtains a TFI, USF(s), and PDTCH(s). In addition, the BSS can request the MS to send information like measurement reports by using the packet polling channel (PPCH). The ongoing service doesn't have link – level retransmissions, others have link – level retransmissions. A close-ended TBF is established for the background service, but open-ended TBF are used for other service.
- At the end of each active period (e.g., no more data to send): For the background service, the MS release its TFI, USF(s), and PDTCH(s). For others service, the MS only release its USF(s) & PDTCH(s), but keep its TFI. In addition, while it does not have an ongoing downlink data transfer, the MS camps on the fast downlink control channel.
- At the beginning of each new period of activity: For the background service, the MS goes through the entire PRACH access procedure. For other services, the MS accesses the system using F-PRACH along with the service access probability and

access procedure. The MS receives USF and PDTCH assignment through an assignment message sent on the F-PAGCH or on a PACCH if it has an ongoing downlink data transfer. In addition, the BSS can request the MS to send information like measurement reports by using the fast packet polling channel (F-PPCH). The polling schedule is determined by the BSS and can take into account the current channel availability.

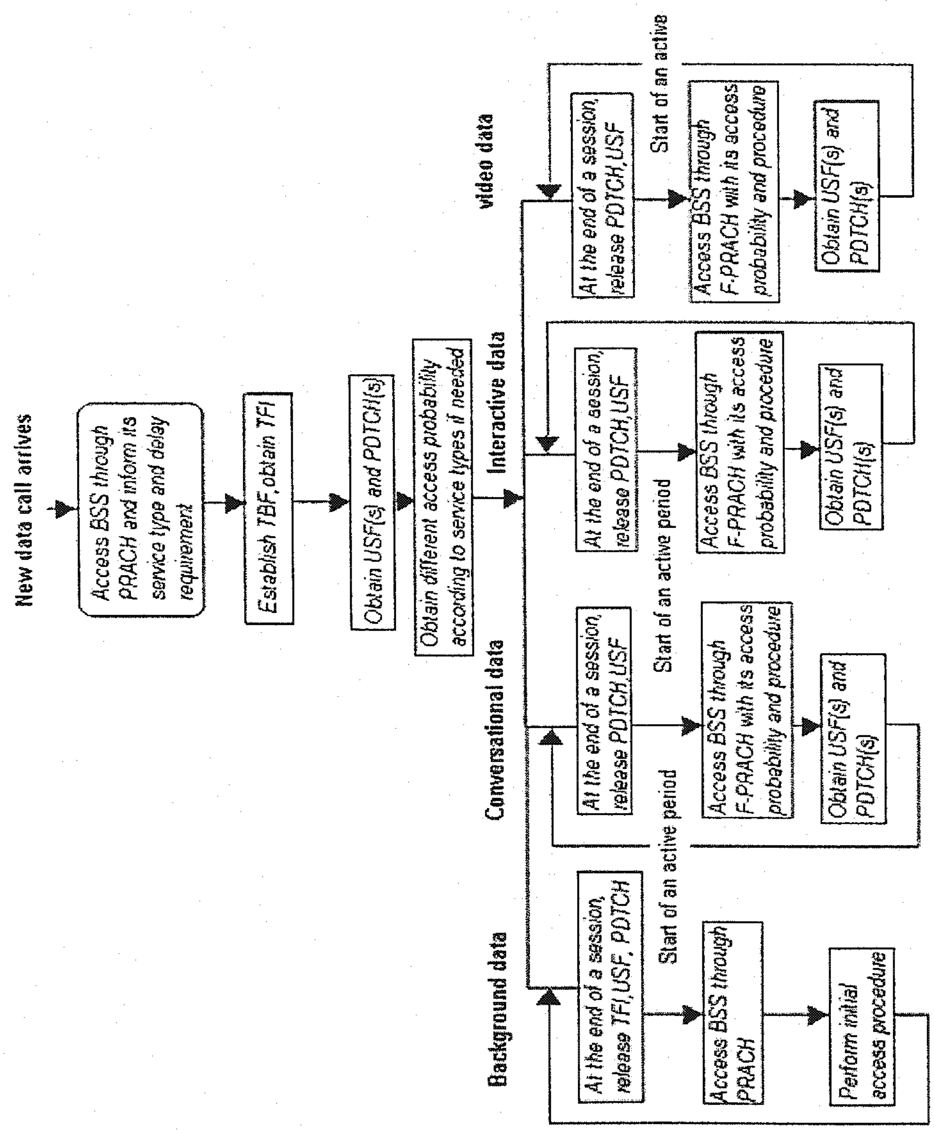


Figure 3.6 EGPRS Block Diagram of Traffic Flow

Chapter 4
An Adaptive Resource Assignment algorithm along with Overload Control

4.1 Introduction

The third generation (3G) of wireless communications will be strongly dominated by various services guaranteeing the Quality of Service (QoS). The required services include voice conversation, Web browse, short messages, real time video, non-real time video, and so forth. Current wireless systems with GSM/TDMA through GPRS towards EGPRS aim to accomplish the various services of third generation wireless communication. One approach is to integrate heterogeneous traffic types with robust Medium Access Control (MAC) protocol, in order to achieve higher spectrum efficiency and the requirements of QoS.

The purpose of this thesis is to investigate the system level performance of EGPRS Phase I in GSM interfering limited cellular environments, and to propose a more efficient assignment algorithms named Adaptive Resource Assignment along with Overload Control (ARAOC) by combining various services into GSM/EGPRS systems to properly guarantee the QoS for various traffic services. The proposed algorithm is designed to include a variety of users with different rates and bursty levels. Our approach is to allocate channels to users of random variable traffic rates during an active period of calls, which is called a session, while the buffer size of users is allocated according to their traffic class. A Static Resource Assignment (SRA) technique used similarly by current GSM network will also be discussed in this thesis.

4.2 Cellular Network

4.2.1 Cell model

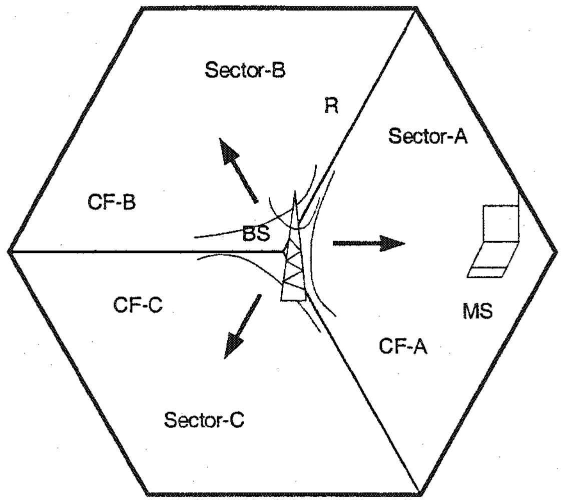


Figure 4.1 Cell model

A regular cell shape is needed for system design and performance analysis. We adopt a hexagonal cell pattern with radius R equal to 3 kilometers. The hexagon is split into 3 sectors, called A, B, and C, respectively. In Figure 4.1, a BS is located at the center of the cell. It uses three direction antennas facing on to three different directions, encompassing 120° within its sector. While users in the same sector cannot use the same channel simultaneously, users in different sectors could share the same channel at the same time. In general, a cell could use all channels in the whole spectrum though limited by the SINR of each channel. In order to lessen the communication burden of neighboring cells, as well as the burden of SINR calculation for each channel, the entire 124 carrier frequencies in a sector are being divided into three channel groups CF-A, CF-B and CF-C, as specified in Sector-A, Sector-B and Sector-C, respectively. When each sector assigns channels for an arriving call, it will first exhaust all appropriate channels within its own group. If the channels of its group are insufficient, then it turns to free channels in the other two groups, and assigns suitable channels for this call, subsequently, sends a message to nearby cells.

We can measure the average signal strength at any point following a power law of the distance of BS and MS. The average received power Pr at a distance d from the transmitting antenna is presented as [43]:

$$P_r = P_0 (\frac{d}{d_0})^{-n} \tag{4.1}$$

or

$$P_r(dBm) = P_0(dBm) - 10n\log(\frac{d}{d_0}) = P_0(dBm) - 10n\log d + 10n\log d_0 \tag{4.2}$$

where *Po* is the power received at a referenced point in the far field region of the antenna at a small distance *do* from the transmitting antenna, *d* is the T-R separation, and *n* is the path loss exponent. The value of n depends on the specific propagation environment [12]. We assume that *n* equals to 4 in shadowed urban cellular environment.

We investigate the inter-user interference on the downstream channel (from base station to users) with static resource assignment, assuming that the propagation path loss is proportional to the fourth power of the distance. To do this, we first consider the case, where a user is located on the sector boundary in the direction of the nearest interference sector (point of Q in Figure4.4), which employs the same frequency. Such a user is at a distance *Ds* from its home base station and at a distance *Di* from the near interfering base station. From formula (4.1), the ratio of interfering power level coming from an interfering sector nearby to signal power is:

$$\frac{Id}{S} = \frac{P_0(\frac{Di}{d0})^{-n}}{P_0(\frac{Ds}{d0})^{-n}} = (\frac{Di}{Ds})^{-n} \tag{4.3}$$

where *Id* denotes the interfering power at a distance *Di*, *S* denotes signal power at a distance *Ds*, *Po* denotes the power of reference point at a distance *d0*, and *n* denotes the path loss exponent, which is 4 in a shadowed urban cellular environment.

The above propagation model is valid for the line of sight transmission. In reality, the received signal level is random and distributed log-normally (dB) about the mean distance-dependent value [44]:

$$Pd = Pr + x \tag{4.4}$$

Here, *Pr* is the mean received power, and *x* is a zero mean Gaussian distributed random variable (in dB) with standard deviation σ (in dB). Usually σ is chosen from 6dB to10 dB.

4.2.2 System model

A hexagonal cell pattern is commonly used in mobile radio systems. In GSM systems, due to interference, the adjacent cells normally cannot use the same carriers, therefore the smallest number of frequency reuse is 1, and a frequency reuse pattern of 3 or 4 is widely deployed. We assume that the system model consist of 19 cells as shown in Figure4.2. In order to increase system capacity, both cell cluster size of 1 and 1/3 will be investigated.

To evaluate the interference problem for downlink, I only consider the 2 tiers interferences from 18 nearby base stations. For cell cluster size of 1, the co-channel interferences from 7 base stations of interfering sectors is considered into SIR calculation, i.e., a mobile station located in A-sector is affected only by other interfering A-sectors; similarly, a mobile station located in B-sector is affected by other interfering B-sectors, and so on. For the cell cluster size of 1/3, the co-channel interferences from 18 base stations of interfering sectors is considered into SIR calculation.

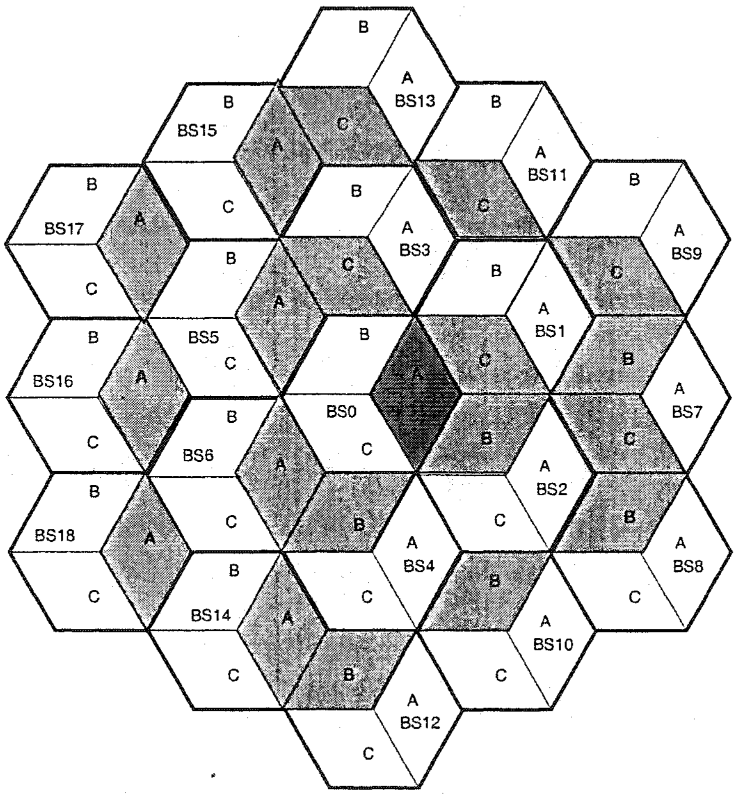


Figure 4.2 A system patterns with19 hexagonal cells split into 3 sectors

If the clocks of different cells are mutually synchronized, then the interference will be limited to users that share the common time slots. In the presence of clock frequency offset, a given user will interfere all users of the affected cells. To proceed further, we assume that all base stations transmit at the same signal level, and that the transmit power control in the upstream direction is perfect, enabling the base station to receive the same signal level from all users in its cell.

4.2.3 Co-channel interference at the worst case with cell cluster size of 1/3

We investigate a user located at the worst-case position with cell cluster size of 1/3, that means each sector using entire spectrum. If considering only 2 tiers interfering situation on the downstream channel, the inter-user interference could be from 18 interfering cells, which employ the same frequency. A user located at the point Q in Figure4.3 could have 18 co-channel interferences. The S/I ratio at the worst case with 18 co-channel interferences is:

$$CIR(18\,\text{interfering sectors}) = \frac{S}{Id} = \frac{1}{\sum_{i=1}^{18} u_i \left(\frac{r}{d_i}\right)^4} \qquad (4.5)$$

where u_i denotes the status of co-channel in interfering *i-th* sector: the status of the used channel is 1 and that of unused channel is 0; r denotes the user distance from home BS; and d_i denotes the distance from interfering *i-th* sector to the user.

From Figure4.3, reference sector-A in BS0 is largely interfered by sector-C of BS1 and sector-B of BS2, the interfering cell and distance are presented below:

BS1 & BS2: $d5=d5'=R$ *(4.6)*

BS3, BS4 & BS7: $d6=d6'=d10=2R$ *(4.7)*

BS5, BS6: $d1=d1'=\sqrt{(\sqrt{3}R/2)^2+(5R/2)^2}=\sqrt{7}R$ *(4.8)*

BS9, BS8: $d9=d9'=\sqrt{(\sqrt{3}R)^2+(2R)^2}=\sqrt{7}R$ *(4.9)*

BS10, BS11: $d8=d8'=\sqrt{(3\sqrt{3}R/2)^2+(R/2)^2}=\sqrt{7}R$ *(4.10)*

BS12, BS13: $d2=d2'=\sqrt{(3\sqrt{3}R/2)^2+(5R/2)^2}=\sqrt{13}R$ *(4.11)*

BS14, BS15: $d7=d7'=\sqrt{(2\sqrt{3}R)^2+(R)^2}=\sqrt{13}R$ *(4.12)*

BS16: $d4=$ 4R *(4.13)*

BS17, BS18: $d3=d3'=\sqrt{(\sqrt{3}R)^2+(4R)^2}=\sqrt{19}R$ *(4.14)*

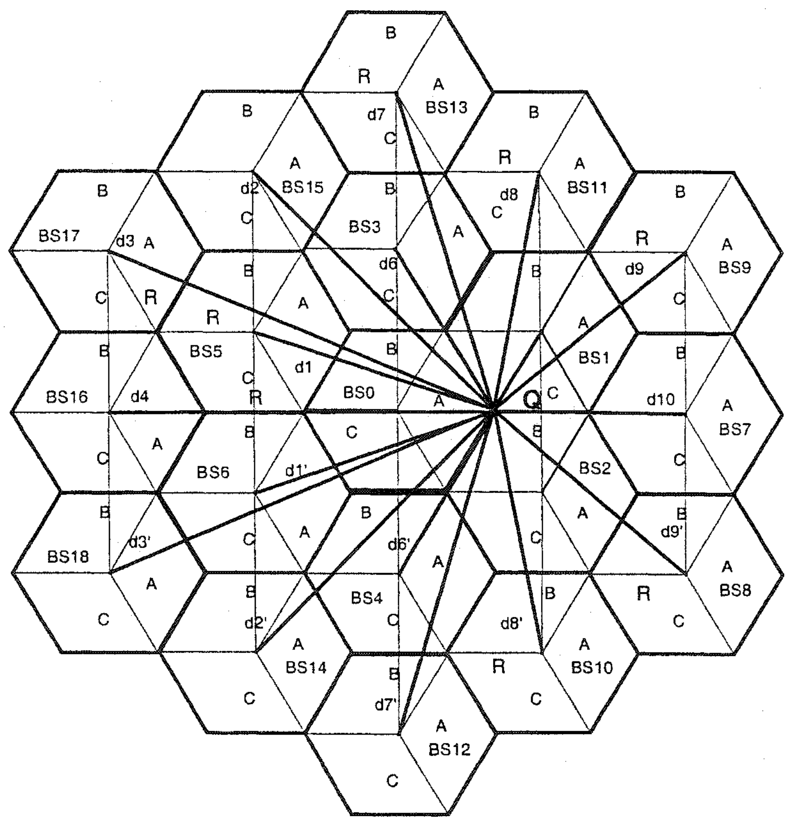


Figure 4.3 a user at the worst case with cell cluster size of 1/3

Taking (4.6)-(4.14) into (4.5), we obtain

$$\frac{Id}{S}=\sum_{i=1}^{18}u_i\left(\frac{R}{di}\right)^n=\left[\frac{R}{d1}\right]^4+\left[\frac{R}{d2}\right]^4+\left[\frac{R}{d3}\right]^4+\left[\frac{R}{d4}\right]^4+\left[\frac{R}{d5}\right]^4+\left[\frac{R}{d6}\right]^4+\left[\frac{R}{d7}\right]^4+\left[\frac{R}{d8}\right]^4+\left[\frac{R}{d9}\right]^4$$

$$+\left[\frac{R}{d1'}\right]^4+\left[\frac{R}{d2'}\right]^4+\left[\frac{R}{d3'}\right]^4+\left[\frac{R}{d5'}\right]^4+2\left[\frac{R}{d6'}\right]^4+\left[\frac{R}{d7'}\right]^4+\left[\frac{R}{d8'}\right]^4+\left[\frac{R}{d9'}\right]^4+\left[\frac{R}{d10}\right]^4$$

$$=2\left[\frac{R}{d1}\right]^4+2\left[\frac{R}{d2}\right]^4+2\left[\frac{R}{d3}\right]^4+\left[\frac{R}{d4}\right]^4+2\left[\frac{R}{d5}\right]^4+2\left[\frac{R}{d6}\right]^4+2\left[\frac{R}{d7}\right]^4+2\left[\frac{R}{d8}\right]^4+2\left[\frac{R}{d9}\right]^4+\left[\frac{R}{d10}\right]^4$$

$$=2\left[\frac{1}{7}\right]^2+2\left[\frac{1}{13}\right]^2+2\left[\frac{1}{19}\right]^2+\left[\frac{1}{4}\right]^2+2\left[\frac{1}{1}\right]^2+2\left[\frac{1}{4}\right]^2+2\left[\frac{1}{13}\right]^2+2\left[\frac{1}{7}\right]^2+2\left[\frac{1}{7}\right]^2+\left[\frac{1}{4}\right]^2$$

$$=2.343 \qquad (4.15)$$

where *Ui equals to 1 for* 7 interfering sectors which use co-channel.

Thus, the S/I ratio with 18 co-channel interferences at the worst case is concluded to be 0.4268 or –3.7dB. The channel with this S/I ratio is not useful. Since higher interferences exist from BS1-BS4, and BS10, in order to lessen these interferences, the channels without co-channel interferences from BS1-BS4 and BS10 are used. Thus, from (4.15), the SIR without co-channel interferences from BS1-BS4 and BS10, i.e., not using the corresponding carriers by current user, is:

$$SIR=\frac{S}{Id}=\frac{1}{2.343-2-3*0.0625}=6.43\rightarrow 8.08\,dB \qquad (4.16)$$

4.2.4 Co-channel interference at the worst case with cell cluster size of 1

In order to lessen the calculation and communication burden of neighbouring cells, we consider a user located at the worst case with cell cluster size of 1. There is a maximum interfering power only from all 7 capable interference sectors to a user, situated on point Q in Figure4.4.

The interference power (normalized by the useful signal power S) from this particular sector in the worst case is:

$$\frac{I_d}{S} = \sum_{i=1}^{7} u_i \left[\frac{R}{d_i}\right]^4 = 2\left[\frac{R}{d1}\right]^4 + 2\left[\frac{R}{d2}\right]^4 + 2\left[\frac{R}{d3}\right]^4 + \left[\frac{R}{d4}\right]^4$$

$$= 2\left[\frac{1}{7}\right]^2 + 2\left[\frac{1}{13}\right]^2 + 2\left[\frac{1}{19}\right]^2 + \left[\frac{1}{4}\right]^2 = 0.0621 \qquad (4.17)$$

where Ui *equals to 1 for* 7 interfering sectors which use co-channel.

Thus, SIR=S/Id =16.1 or 12.1dB.

Note that the foregoing analysis holds when all the interfering sectors are using co-channel with a cell frequency reuse factor of 1. The downlink S/I ratio of 12.1dB will be valid only when the time slot assigned to the user is also simultaneously used in all of the 7 interfering sectors that employ the same frequency. Otherwise, the S/I ratio will be larger than 12.1dB.

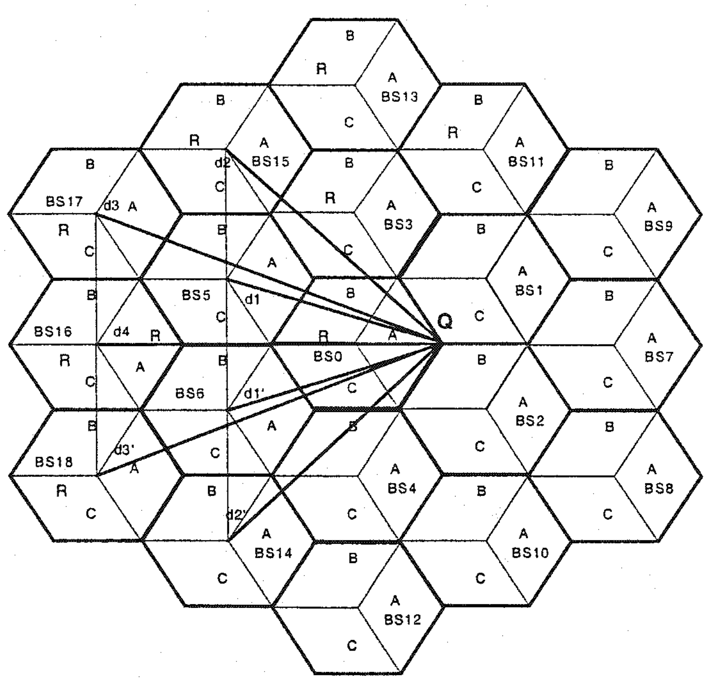


Figure 4.4 a user at the worst case with cell cluster size of 1

4.2.5 Co-channel interference at the average case with cell cluster size of 1/3

From the previous analysis, the S/I ratio of some users may be decreased to 12.1dB under the static resource assignment parameters. Here, we investigate the inter-user interference on the downstream channel with dynamic resource assignment. Since BSS will reassign channels at the end of four multi-frames, this function is equivalent to the application of frequency hopping at every four multi-frames. Thus, users can avoid the worst case. To investigate this, we focus on the location suffered from average interference, when users are located in average distance point of M in Figure4.5.

Similarly, assuming the worst case with 18 co-channel interferences, we were always able to find out channels with an average SIR to assign to users. The interference powers (normalized by the useful signal power S) from this particular sector for the user at the averaging interfering location is:

$$\frac{Id}{S}=\sum_{i=1}^{18}u_i\left(\frac{R}{di}\right)^n=2\left[\frac{R/2}{d1}\right]^4+2\left[\frac{R/2}{d2}\right]^4+2\left[\frac{R/2}{d3}\right]^4+\left[\frac{R/2}{d4}\right]^4+2\left[\frac{R/2}{d5}\right]^4+2\left[\frac{R/2}{d6}\right]^4$$
$$+2\left[\frac{R/2}{d7}\right]^4+2\left[\frac{R/2}{d8}\right]^4+2\left[\frac{R/2}{d9}\right]^4+\left[\frac{R/2}{d10}\right]^4 \qquad (4.18,$$

where *Ui equals to 1 for* 7 interfering sectors which use co-channel.

Where $d1=d1'=\sqrt{(\sqrt{3}R/2)^2+(2R)^2}=\sqrt{19}R/2$

$d3=d3'=\sqrt{(\sqrt{3}R)^2+(7R/2)^2}=\sqrt{61}R/2$

$d2=d2'=\sqrt{(3\sqrt{3}R/2)^2+(2R)^2}=\sqrt{43}R/2$

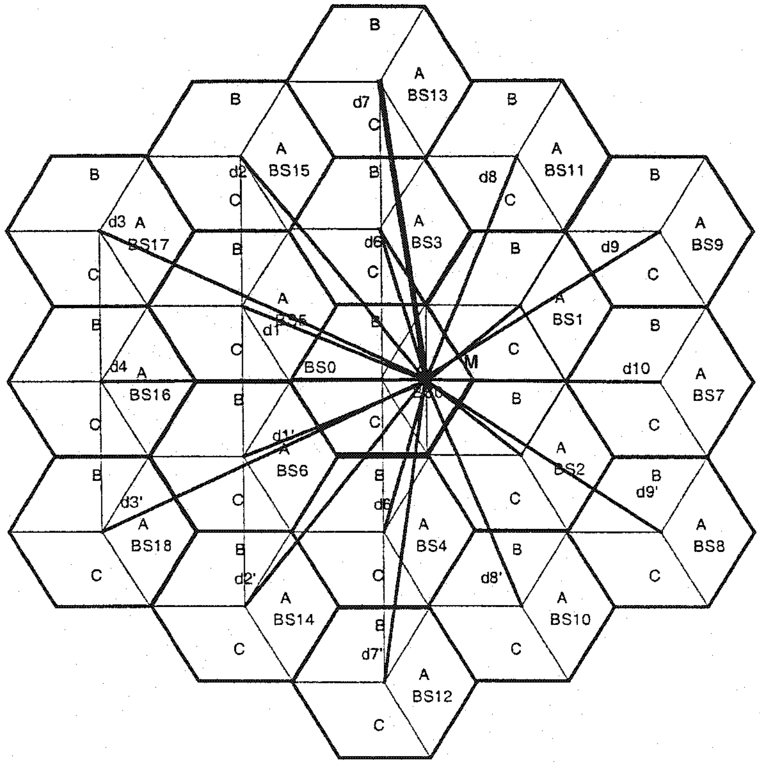


Figure 4.5 a user at the average case with cell cluster size of 1/3

$d4 = 7R/2,\ d10 = 5R/2$

$$d5 = d5' = \sqrt{(\sqrt{3}R)^2 + (R/2)^2} = \sqrt{13}R/2$$

$$d6 = d6' = \sqrt{(2\sqrt{3}R)^2 + (R/2)^2} = \sqrt{49}R/2$$

$$d7 = d7' = \sqrt{(3\sqrt{3}R/2)^2 + (R)^2} = \sqrt{31}R/2$$

$$d8 = d8' = \sqrt{(\sqrt{3}R/2)^2 + (R)^2} = \sqrt{7}R/2$$

$$d9 = d9' = \sqrt{(\sqrt{3}R)^2 + (5R/2)^2} = \sqrt{37}R/2$$

Thus, (4.18) is changed to:

$$\frac{Id}{S}=\sum_{i=1}^{18}u_i(\frac{R}{di})^n=2\left[\frac{1}{19}\right]^2+2\left[\frac{1}{43}\right]^2+2\left[\frac{1}{61}\right]^2+\left[\frac{1}{49}\right]^2+2\left[\frac{1}{13}\right]^2+2\left[\frac{1}{49}\right]^2+2\left[\frac{1}{31}\right]^2+2\left[\frac{1}{7}\right]^2$$

$$+2\left[\frac{1}{37}\right]^2+\left[\frac{1}{25}\right]^2=0.007576+0.0586=0.0662 \qquad (4.19)$$

where Ui *equals to 1 for* 7 interfering sectors which use co-channel.

Thus the average SIR under 18 co-channel interferences with cell cluster size of 1/3 is:

Average SIR (18 co-channel interferences) = S/Id = 1/0.0662 = 15.1 or 11.8dB

4.2.6 Co-channel interference at the average case with cell cluster size of 1

In order to improve SIR and hence probability of bit errors, we investigate the inter-user interference which is from 7 interfering sector-As on the downstream channel with frequency reuse factor 1, and the intended user is located on point M of Figure4.6.

From (4.18) and (4.19), The S/I ratio with 7 co-channel interferences is shown as:

$$\frac{I_d}{S}=\sum_{i=1}^{7}u_i(\frac{R/2}{di})^n=2\left[\frac{R/2}{d1}\right]^4+2\left[\frac{R/2}{d2}\right]^4+2\left[\frac{R/2}{d3}\right]^4+\left[\frac{R/2}{d4}\right]^4=0.00757 \qquad (4.20)$$

where Ui *equals to 1 for* 7 interfering sectors which use co-channel.

Thus, the average S/I ratio for the downlink is

$$\text{SIR}=S/Id = 132\text{, or } 21.1\text{ dB.} \qquad (4.21)$$

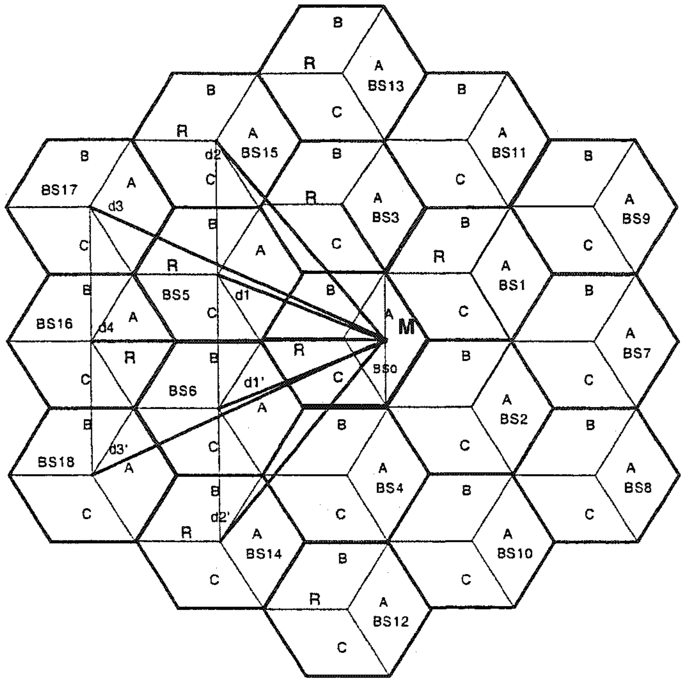


Figure 4.6 A user in the average case with cell cluster size of 1

4.3 Description of System Parameters

4.3.1 Traffic Model

In the simulation, we consider five traffic classes according to the requirements of different services. These traffic classes are background, conversational, interactive, and streaming classes for real-time and non real-time services respectively. They can also be divided into two main services: Real Time services (e.g., voice, video conferencing, etc.), and Non Real Time services (e.g., database applications, web browsing, email, streaming video and sound, etc.).

Table 4.1 describes each of these classes along with their characteristics:

We first define the rate range for each class according to the traffic character of service types. Each traffic class is characterized by a set of QoS requirements that need to be satisfied in an end-to-end mode. Both the wireless part and the fixed subsystems of a mobile communication system are responsible for providing and maintaining the required QoS.

Table 4.1 The Five Traffic Classes in our simulation

Traffic class	Background class	Conversational class	Interactive class	Streaming class (real time)	Streaming class (Non-real time)
Rate range	2.5~10Kbit/s	5~20Kbit/s	10~40Kbit/s	20~80Kbit/s	40~120Kbit/s
Variable Rate	Varies randomly with uniform distribution	Varies randomly with uniform distribution	Varies randomly with uniform distribution	Varies randomly with uniform distribution	Varies randomly with uniform distribution
Application	-Background download of e-mail -File transfer	-Voice over IP -Real-time	-Web browsing -Database retrieval	-Real-time video -Video conferencing	-Streaming video and sound -Non real time

4.3.2 Classic Quality of Service (QoS) attributes:

- The characteristics specified in term of bandwidth:
  - The peak rate (bit/sec),
  - The minimum acceptable rate (bit/sec),
  - The average rate (bit/sec),
  - The maximum burst size (the maximum number of consecutive bits sent at the peak rate)
- The reliability requirements of the connection:
  - The Bit Error Rate (BER) or Frame Error Rate (FER),
  - The maximum loss ratio (the proportion of received packets to undelivered packets),

➢ The delay requirements:

- The maximum tolerated delay (ms),
- The maximum tolerated jitter (ms) (the variation in delay),

4.3.3 Description of Input Parameters

There are many input parameters that can be varied in order to analyse the performance under various scenarios. As described above about Quality of Service, the following input parameters are used in our simulation:

- Data rate for each time slot

To compare various performances of our simulated network with the existing GSM network, we assume each channel data rate is 13.0 kbps, the same as GSM system. To support higher data rates, the channel data rate can be increased when higher-level modulation, i.e., 8-phase shift keying (8-PSK), is used.

- Normalized sector rate load (ρ):

It is defined as the ratio of the summation of average rate of all calls accepted over system maximum transmission data rate in a sector.

$$\rho = \frac{\sum_i \textit{Average rate of traffic classes of accepted call(i)}}{\textit{41 carriers * 8 slots * 13.0 Kbits/s slot rate}} \qquad (4.22)$$

Since the system maximum transfer data rate with 328 channels includes 21 channels for signalling transfer, the ρ equalling to 93.6% means that the whole sector channels have averagely reached the full rate transmission.

- Sector rate load threshold (SLT) for ARAOC:

We should set a sector load threshold (SLT) for dynamic resource assignment. When the normalized sector rate load (ρ) is over a fixed value of SLT, BSS only holds the existing calls in the sector. All new arriving calls cannot have access to this sector. As signalling data transmission uses 21 of 328 channels in each sector, the value of SLT is set to 90%.

- Sector rate load limit for SRA: It is set as call dropping probability of 2%, which is the requirements of GSM systems.
- Channel buffer size [packets]: Different buffer size of 4, 8, 16, 32 or 64 packets is used to compare its influence on the performance. A fixed buffer size of 16 packets is used in other performance simulations to find the influence of other parameters.
- User's Buffer size [packets]:

  It is equal to channel buffer size multiplied by assigned channel number of user.
- Static assignment parameter: it takes a value from 0.0, 0.25, 0.5, 0.75 and 1.0.
- Resource allocation type: The allocation is dynamic. The resource is released if a user's packets transmission is finished.
- Scheduling transmission algorithm: First in and first out.
- All base stations and mobile stations transmit with the same power. Power control is not considered.

4.3.4 Description of Output Parameters

There are many output parameters generated by the simulator. The parameters used in this paper to analyse the performance under various scenarios are:

- User Throughput

The average network throughput is defined as the ratio of the number of packets successfully transmitted in a long interval to the maximum number of packets that can be transmitted continually on the channels [9]. The user throughput is defined as:

$$User\ Throughput(\eta_i) = \frac{Total\ Number\ of\ RLC\ Block\ delivered\ by\ user\,(i)}{Total\ Number\ of\ RLC\ block\ could\ be\ delivered\ by\ user\,(i)} \qquad (4.23)$$

$$Average User Throughput : \overline{\eta} = \frac{1}{N}\sum_{i=1}^{N}\eta_i \qquad (4.24)$$

Variance of average user throughput:

$$\sigma_{\eta}^{2} = \frac{\sum_{i=1}^{N}(\eta_i - \overline{\eta})^2}{N-1} \qquad (4.25)$$

Where N denote the number of user (i) existing in system.

- Queuing Delay:

The queuing delay is defined as the duration from the instant of the packets getting into user's buffer to the instant of these packets successfully getting out. The average queuing delay is defined as the total number of packets in the buffer of all users during certain iterations divided by the number of iterations, and by the number of accepted users. By simulation model, we can write the queuing delay per user as:

$$Di = \frac{\sum_{j=1}^{M} buffer\ content\ of\ user(i)\ at\ iteration(j)}{M} \qquad (4.25)$$

where M denotes the number of iterations for each user.

Average queuing delay:

$$\overline{D} = \frac{\sum_{i=1}^{N} D_i}{N} \qquad (4.26)$$

Variance of average queuing delay:

$$\sigma_D^2 = \frac{\sum_{i=1}^{N} (D_i - \overline{D})^2}{N-1} \qquad (4.27)$$

where N denotes the number of accepted user (i) existing in system.

- Buffer overflow:

I define the buffer overflow as the number of packets dropped when a given threshold (Maximum buffer size) is exceeded. The average buffer overflow of user ***i*** is defined as the total number of times of user's buffer size overflow divided by the number of users (N), and by the number of iterations (M), i.e.

$$BO_i = \frac{\sum_{j=1}^{M} Number\ \ of\ \ blocked\ \ packets\ \ of\ \ user(i)\ \ at\ \ iteration(j)}{M} \qquad (4.28)$$

where M denotes the number of iterations for each user.

Average Buffer Overflow:

$$\overline{BO} = \frac{\sum_{i=1}^{N} \sum_{j=1}^{M} Number\ \ of\ \ blocked\ \ packets\ \ of\ \ user(i)\ \ at\ \ iteration(j)}{N * M} \qquad (4.29)$$

Variance of average buffer overflow:

$$\sigma_{BO}^2 = \frac{\sum_{i=1}^{N} (BO_i - \overline{BO})^2}{N-1} \qquad (4.30)$$

where N denotes the number of accepted users (i) existing in the system.

- Network Utilization Efficiency:

The network utilization efficiency is used to present the system capacity for packet data traffic. To evaluate the system performance under different carrying load, its definition is the ratio of the sum of all transmission rates used to the total of all full traffic channels rates. To simplify, we assume 13.0 Kbit per second as the transmission rate for each time slot. Usually, each sector uses 41 carriers, and each carrier has 8 time slots. Total available time slots for each sector are 41*8=328, including 21 time slots for signalling transmission. These 21 time slots are not used by user traffic. The network utilization efficiency NE per sector is represented by:

$$NE = \frac{\sum_{i=1}^{N} transmission\ rate\ of\ user(i)}{Total\ rate\ of\ all\ channels\ per\ sector}$$

$$= \frac{\sum_{i=1}^{N} transmission\ rate\ of\ user(i)}{41\ carriers * 8\ slots * 13.0\ Kbitps\ per\ slot} \qquad (4.31)$$

When the network in the sector is fully loaded, NE only equals to 93.6%.

- Call dropping/blocking probability:

The call dropping probability is defined as the ratio of the number of forced terminated handoff calls to the number of all arriving handoff calls in this sector. The call dropping probability per cell is represented as:

$$P_{drop} = \frac{Number\ of\ calls\ dropped}{Total\ number\ of\ handoff\ calls\ arrived} \qquad (4.32)$$

The call blocking probability is defined as the ratio of the number of new calls blocked to the total number of all arriving calls in this sector at a certain interval. The call

$$P_{block} = \frac{Number\ of\ new\ calls\ blocked}{Total\ number\ of\ arrival\ calls} \qquad (4.33)$$

blocking probability per cell is represented as:

4.4 Description of the SRA Algorithms

4.4.1 Static Assignment parameter

The admission controller decides which channel resources meet QoS constraints thus can be accepted into the available channel groups A connection request is accepted only when sufficient resources are available to establish the call at its required QoS and maintain the agreed QoS of existing calls. Thus, an important parameter, Assignment Parameter (AP) greatly affects the performance of the system and users. It is introduced into our simulation. AP varies within the scope from 0.0 to 1.0. When AP is increased, the assigned time slots and transmission rate will also increase for all users.

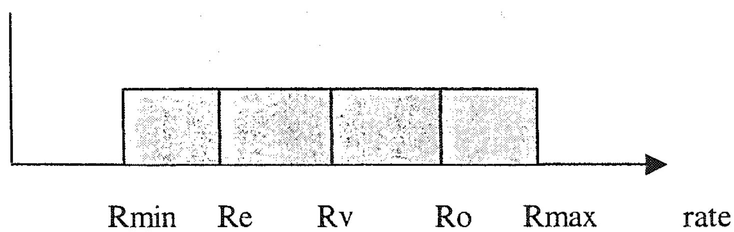


Figure 4.7 illustrate the rate of traffic class with uniform distribution

To simplify in Figure4.7, we define five assignment parameters in the following:

1) Minimum assignment parameter: AP = 0.0, with min transmission rate (Ri);
2) Less assignment parameter: AP = 1/4 = 0.25, with less transmission rate (Re);
3) Average assignment parameter: AP = 0.5, with average transmission rate (Rv);
4) Most assignment parameter: AP = 1/4 = 0.75, with most transmission rate (Ro);
5) Maximum assignment parameter: AP = 1.0 with max transmission rate (Ra).

4.4.2 Instant transmission rate

In 3.3.1 section, five traffic classes have been defined. Their rates vary randomly with uniform distribution in their variable scope. Thus, the instantaneous transmission rates are [35]:

$$R = Rmin + U(0,1) * (Rmax - Rmin) \tag{4.34}$$

While Rmin denotes the minimum capable transmission rate for each traffic class, Rmax expresses the maximum transmission rate for each traffic class, and U (0,1) indicates variable random from 0.0 to 1.0 according to uniform distribution.

4.4.3 Statically assigned transmission rate

As we have known, the transmission rates are random variables distributed uniformly in their variable scopes. To evaluate the performance of system and users, we use the above-mentioned five assignment parameters to allocate the transmission rates.

A general assignment rate is:

$$R = Rmin + AP * (Rmax - Rmin) \tag{4.35}$$

While *Rmin* denotes the minimum transmission rate for the traffic class, *Rmax* expresses the maximum transmission for the traffic class, and AP indicates the value of the assignment parameter.

For different assignment parameters, possible assignment rates respectively are:

1) Ri = Rmin;
2) Re = Rmin + (Rmax – Rmin) / 4;
3) Rv = Rmin + (Rmax – Rmin) / 2;
4) Ro = Rmin + (Rmax – Rmin) * 3 / 4;
5) Ra = Rmax.

4.4.4 Static buffer size

Another important parameter that affects the QoS of users is the buffer size of users. The larger the user buffer size is, the buffer overflow will be smaller, but the delay would be longer. Due to different traffic classes, which have different transmission rates and large rate variation, the buffer size should be different according to traffic classes. In this simulation, we consider making a fixed channel buffer size for each time slot assigned to a certain user. The more time slots are assigned, the larger is the buffer size. The user's buffer size equals to:

User buffer size = channel buffer size * number of assigned channel of user

We should find out a suitable channel buffer size to balance the buffer overflow and queuing delay in our simulation.

4.4.5 Loading limit

To improve the performance of system, we use an average call dropping probability (CDP) to control call admission, and to balance call dropping probability and call blocking probability. When the CDP is over the system requirement of 2%, a new call admission will be controlled, and handoff call is not limited. The details are as follows:

- When the call dropping probability is less than 2%, BSS will look for suitable channels in resource pool to assign to arriving calls;
- When the call dropping probability is larger than 2%, BSS first finds suitable channel(s) in the whole channel group of the sector, which may accept user request, then in other two channel groups only for an arriving handoff calls or a new call, who require only less than 3 time slots;

4.5 Adaptive Resource Assignment along with Overload Control

4.5.1 Introduction

With the increase in the demand on wireless/mobile communications and the emergence of bandwidth-intensive multimedia applications, the link bandwidth of wireless/mobile networks is becoming a bottleneck. The scarcity in wireless resources motivates us to research on the adaptive channel allocation and SIR calculation algorithm that can operate over a wide range of available bandwidth with small cell cluster size (see the section of 4.2.1). In this algorithm, a hexagonal cell is split into 3 sectors, each with its own channel group. The cell cluster size of 1 or 1/3 is used for each sector. A dynamic co-channel interference algorithm can calculate the SIR of candidate channels depending on cell cluster size of 1/3. However, BSS first assigns candidate free channels based on cell cluster size of 1. In this case, each cell uses the entire spectrum bandwidth; otherwise, only 1/3 was used in each sector with its own channel group. When the sector does not have adequate free channels for an arriving call, the BSS will search from 2 other channel groups. On the other hand, a new congestion control scheme is needed to guarantee that the system is properly operated and the requirements for QoS in various data services under overloading situation is satisfied. Therefore, we develop an Adaptive Resource Assignment Algorithms along with Overload Control (ARAOC), which combines the advantages of static and dynamic schemes to improve spectrum utilization efficiency, the system performance, and to guarantee the requirements of QoS.

4.5.2 Call Admission Control Schemes of ARAOC (see Figure4.9)

a. Updating the resources condition and calculating C/I ratio for every 4 multi-frames:

- First, there are two tables of resource and user information built up for each sector. One holds user information and traffic status (like later Table 4.2). Another holds the resource information of local sector (see later Table 4.3), and 18 nearby interfering sectors with channels used. The information is updated at every iteration interval for resource table, at every four multi-frames (104 iterations) for user table.

- To calculate the ratio of the signal power to interfering powers (S/I ratio) for each time slot, we search for co-channels interferences from the 18 nearby interfering sectors (see Figure4.2 in section 4.2.3). The interfering powers from 18 nearby interfering sectors should be calculated and updated into an interfering state table.
- Assume every BSS transmits at the same power. The S/I ratio at a channel is:

$$CIR(18\,\text{interfering sectors}) = \frac{1}{\sum_{i=1}^{18} u_i \left(\frac{r}{d_i}\right)^4}$$

Here, r is the distance from BS of the reference sector to the user, d_i is the distance from the BS of interfering sector i to the user, and u_i is a variable which takes the value of 1 when a co-channel is used in an interfering sector, and 0 otherwise. The detail of calculating SIR was shown on the formula (4.18) – (4.21).

b. Dynamic resource allocation in each sector: The entire 124 frequency carriers (CF) are being divided into three channel groups, called CF-A, CF-B, CF-C, which corresponds to sector A, B, C, respectively.

These three channel groups are shown as:

The number of carriers in CF-A: 3* N (N=0, 1, 2, ... 40)

The number of carriers in CF-B: 3* N+1 (N=0, 1, 2, ... 40)

The number of carriers in CF-C: 3* N+2 (N=0, 1, 2, ... 40)

I. We first assign free channel(s) from the sector's own channel group, and then from the other two channel groups. Assigned number of channel(s) for each user depends on the instant transmission rate at every 4 multi-frames. The channel numbers in a frequency carrier can reach up to 8 timeslots.

II. When a sector is loaded under a designated threshold (DT), the BSS only searches in its own channel group of the sector.

III. When a sector is overloaded, the BSS first searches suitable free channels in its own entire channel group. If free channels are not found, the BSS will search other channel groups. For instance, when a user accesses to the cellular network from sector A, BSS of the sector A first searches for suitable free channels in CF-A. If CF-A is not available for this user, the BSS will look for available channels in CF-B and CF-C. If there are unavailable channels in resource pool for this user, the BSS will block this call. Needless to say, a new call that will cause the SIR of ongoing calls below threshold will be blocked.

IV. Adaptive Channel Assignment Strategy:

- First, network average loading factor (ρ) and average dropping probability (P_{drop}) will be calculated at each iteration.
- When an active period of an existing call with TFI along with identifying applicable information requires fast access channels path, BSS looks for a suitable assigned channel from the pool.
- When the loading factor (ρ) in a sector is less than the designated threshold (see section 4.3.2), the BSS uses dynamic assignment schemes in the sector, by allotting time slots as much as possible according to the user's instantaneous rate

of each four multi-frames, and providing the best effort service. On the other hand, when the loading factor (ρ) in a sector is larger than a loading threshold, the BSS resorts to static assignment schemes. It assigns fixed time slots for each user according to its traffic class and assignment parameter (such as: select AP=0.5). When AP corresponds to satisfying the basic service for each traffic class. If the loading factor ρ is less than loading threshold again, the BSS resorts to dynamic assignment schemes.

- When the loading factor is less than the designated threshold, and the average dropping probability is less than 2%, the BSS will look for available channels from the resource pool, and allocate channels to arriving calls. It still uses the dynamic assignment scheme.
- When the loading factor is less than the designated threshold, and average dropping probability of a sector is larger than 2%, an arriving handoff calls, or a new call, which only needs less than 3 time slots, is allowed to look for adequate available channel(s) from the resource pool.
- When the loading factor is over the designated threshold, the BSS changes to static assignment policy, which allocates fixed time slots to an arriving call according to the traffic class and the system assignment policy. The BSS provides acceptable basic rate. In the process, the BSS first searches suitable free channels from its own channel group; if no free channel available, it continues searching from the 2 other channel groups; finally, the BSS allots the channels chosen for this arriving call. However, the BSS will reject all of the handoff and new calls if all 3 channel groups are not available.

c. Dynamic Borrow Resource Policy: As per the previous description of channel groups in the section of 4.2.1, a hexagonal cell is split into 3 sectors. When a cell cluster size is 1, the entire 124 carrier frequencies in a sector are being divided into three channel groups CF-A, CF-B and CF-C, as specified in Sector-A, Sector-B and Sector-C, respectively. When a sector assigns channels for an arriving call, it will first exhaust all appropriate channels within its own group. If its channel group proves to be insufficient, then it turns to the other two channel groups. If the channels in the other two channel groups are available, the BSS will borrow the channels from other channel groups for this arriving call. For each four multi-frames, when its own channel group is available for the active calls, the borrowed channels will be returned back; when its own group is unavailable for active calls, the borrowed channels will be still used for the user. If the S/I ratio of some resources satisfy the requirement of this call. BSS can assign the time slot(s) to the user, and inform nearby interfering sectors. This means that BSS borrows the resources from nearby sectors, and also implies full communications and exchange of the reservation tables among BSSs.

4.5.3 Congestion Control Schemes

To improve the network utilization efficiency, we assume that users do not operate at their peak rate values simultaneously. Since the traffic demands are stochastic and unpredictable, congestion is inevitable. When congestion occurs, the performance of queue delay, buffer overflow and packet loss may become very large in a short time. To secure the Qos of users, thus, congestion control is necessary, and the following scheme is used:

a. Set two thresholds to detect congestion:

This new scheme is based on monitoring the user buffer occupancy. We set two thresholds: assault threshold (Ta) and reduction threshold (Tr). When the user buffer occupancy exceeds the assault threshold, a congestion assault message is sent to the user via control channel. On the other hand, when the buffer occupancy goes below another threshold (Tr), a congestion reduction message will be sent to the user via control channel. Upon receipt of the congestion assault message, the users are expected to reduce the traffic rate so as to yield speedy recovery from congestion. In contrast, upon receipt of the reduction message, the user traffic will be restored to pre-congestion levels.

As mentioned above in the section 3.9.2, the Signalling exchange between the BSS and a MS uses two logical channels in EGPRS: a Fast Packet Access Channel (F-PACH), a Fast Packet Control Channel (F-PCCH). In uplink packet access, the fast packet channel requesting message carried in F-PACH) contains information on the specific TBF being referenced (i.e., the uplink TFI assigned to the MS and other relevant information). Based on this information, the BSS can uniquely identify the MS and its specific application, and therefore quickly assign the necessary uplink resource. The F-PACH with individual bursts messages is transmitted into special physical channel (e.g., time slot 0) of some selected carriers. In downlink channel assignment, the BSS transmits the access grant and control messages to specific MS via F-PCCH. The F-PCCH is split into two logical channels: a Fast packet access grant channel (F-PAGCH) and a Fast Packet Polling Channel (F-PPCH). These two channels can be time multiplexed on the same physical channel located on specific time slots (e.g.. time slot 0) of some selected carriers. The F-PAGCH is used to respond to access requests received on the F-PACH. This response is a typical assignment message that specifies the channels, USFs, and other parameters for a set of MSs. F-PPCH is used to poll different mobiles. If an MS has an ongoing downlink data transfer, it is possible to use the PACCH to transmit control messages to the MS. Otherwise; the BBS can use the downlink F-PCCH to communicate with the MS. Each pair of F-PACH and F-PAGCH/F-PPCH may carry the fast uplink access request, access grant, polling, and polling response messages for a set of carrier frequencies. In this thesis, we consider the congestion only on the wireless uplink and down link, and impossible congestion in the associated terrestrial Internet.

b. Automatic Adjust Assignment parameter:

When the loading factor is less than its loading threshold (light load), the BSS will use the dynamic assignment scheme to assign channels as much as possible according to user requirements. In this case, sectored system does not face traffic congestion.

When a sector load is larger than its threshold (overloaded), traffic congestion will develop. In order to control traffic congestion at the system level, dynamic channel allocation is adapted to static channel assignment. The BSS will limit the transmission rate of all existing calls at the basic fixed rate, which depends on static assignment parameter (AP) and the type of traffic class. When a sector load becomes light, the traffic congestion will be eliminated. Static channel allocation is adapted to the dynamic channel assignment; channels are assigned as many as possible to meet the user's request adequately.

4.6 Description of Simulations

The system level simulator, used to produce the results presented here, models the transfer of data between the BSS and MS in the designated cell. This is achieved by modelling comprehensively all the aspects of the data transactions as specified in the RLC/MAC GSM specification. The following sub-sections remark the overall description, simulation description of SRA and ARAOC presented in this thesis.

4.6.1 Overall Simulation Description

A cellular network described in Chapter 3.2 with 19 hexagonal cells, which are being split into 3 sectors A, B, C, is simulated as the subject network model. Each sector has a resource pool, including three channel groups: CF-A, CF-B, CF-C, used in sectors A, B, C, respectively. Each channel group has 41 carriers, each with 8 time slots. Thus, each sector has 307 channels for traffic channels, and 21 channels for signalling channels. However, in ARAOC algorithms, when the channels in local sector group are not available to a call, it will try to borrow channels from the two other channel groups.

The base time unit in the simulator is a radio link control (RLC) block; each RLC block period is referred to as one program iteration. During the operation of the system, a call is randomly generated with a handoff call probability of 30% and a new call generation probability of 70% every 10 iterations. BSS also randomly releases a call every 10 iterations with a uniform distribution because the user is being switched to other cells, or being terminated.

We first build up 2 tables about the user and resource information for each sector. The user information table stores the number of user, user ID, instantaneous data rate, rate type, as well as an amount of assigned time-slots with required SIR and relative parameters (i.e., Table 4.2). The resource table stores the number of channels, its state (used or unused) and situation (SIR), as well as their states of 18 nearby co-channel interferences with user ID (i.e., Table 4.3).

Most recent mobile systems (such as IS-136 and GSM) provide voice service with tolerable call dropping probabilities around 2%, speech rate at 13 kb/s, data rate up to 9.6 kb/s. To compare these integrated data services with GSM, we assume the characteristics and environment as in the GSM system. For example: channel data rate of 13.0kb/s, modulation of 0.3 GMSK and call dropping probability of 2%, etc. Then, we investigate the system capacity and performance, and introduce various services in both SRA and ARAOC algorithms. System using different modulations (GMSK and 8-PSK) and channel data rates can support different maximum user rate and system capacity.

There are six major parts in the simulations of both SRA and ARAOC algorithms. The first one is the initialization part, which sets various input parameters. The second one models calls and packet generator, which yields the call model with its service type and

Table 4.2 A part of User Information Table for each sector

User No.	UserID No.	Data rate (Kbps)	Rate type	TS No.	AP	SIR (dB)	Run times	No. Average sentPicket	No. average blockPicket
0	7206	96.4657	5	6	0.5	21	254	5.76402	0.306248
1	10893	99.8062	5	6	0.5	21	798	5.9373	0.108037
2	25208	83.5334	5	6	0.5	21	201	5.39063	0.36
3	9339	105.377	5	6	0.5	21	484	5.63179	0.180602
4	9789	111.757	5	6	0.5	21	619	5.87041	0.118668
5	3997	57.7391	5	6	0.5	21	493	5.7921	0.161507
6	28934	100.986	5	6	0.5	21	308	5.54927	0.317339
7	3087	77.9824	5	6	0.5	21	119	5.41037	0.468791
8	13385	89.8613	5	6	0.5	21	520	5.86049	0.16839
9	17794	58.0911	5	6	0.5	21	511	5.74059	0.189253
10	16679	97.0341	5	6	0.5	21	130	5.14863	0.412791
11	23645	51.5393	5	6	0.5	21	425	5.6331	0.183237
12	32246	61.0692	5	6	0.5	21	879	5.93028	0.095453
13	17690	69.9976	5	6	0.5	21	624	5.81521	0.156007
14	24458	50.2435	5	6	0.5	21	303	5.62248	0.22195
15	27885	43.0284	5	6	0.5	21	101	5.2751	0.542115
16	25077	58.5283	5	6	0.5	21	476	5.75128	0.176498
17	943	84.4734	5	6	0.5	21	824	5.84997	0.119859
18	5613	63.6597	5	6	0.5	21	200	5.43656	0.392443
19	14160	88.9357	5	6	0.5	21	753	5.84435	0.107738
20	5109	102.288	5	6	0.5	21	419	5.71612	0.177425
21	24535	116.223	5	6	0.5	21	600	5.7674	0.167555
22	27008	73.3247	5	6	0.5	21	599	5.86207	0.136738
23	31305	47.6803	5	6	0.5	21	598	5.81022	0.171772
24	13520	82.3117	5	6	0.5	21	685	5.82331	0.136232
25	9725	53.519	5	6	0.5	21	142	5.19592	0.446058
26	28272	53.9252	5	6	0.5	21	597	5.86209	0.129178
27	20475	80.2472	5	6	0.5	21	500	5.74709	0.183637
28	14846	75.0517	5	6	0.5	21	213	5.48077	0.334225
29	15954	113.661	5	6	0.5	21	556	5.64078	0.151611
30	14054	58.6473	5	6	0.5	21	555	5.75373	0.187326

Table 4.3 A part of Resource State Table for each sector

ChNo	Rate	stat	SIR	UserId	IS-0	IS-1	IS-4	IS-5	IS-12	IS-13	IS-16	IS-17	IS-18
0	13	1	21.206	1	0	0	20710	14958	15316	672	4139	26820	22457
1	5.047	1	21.206	17149	0	0	20710	14958	15316	672	4139	26820	22457
2	2.57	1	21.206	8068	0	0	20710	14958	15316	672	4139	26820	22457
3	4.325	1	21.206	16090	0	0	26990	28072	20896	6785	26431	27834	22726
4	6.345	1	21.206	20838	0	0	11844	24991	9162	1318	31930	17881	23285
5	7.553	1	21.206	20373	0	0	11844	24991	9162	1318	31930	17881	23285
6	9.963	1	21.206	9161	0	0	21879	12705	28392	21939	7616	22891	6808
7	3.509	1	21.206	5046	0	0	21879	12705	28392	21939	7616	22891	6808
8	0	0	4.5358	0	0	1440	0	0	0	0	0	0	0
9	0	0	4.5358	0	0	1440	0	0	0	0	0	0	0
10	0	0	4.5358	0	0	1440	0	0	0	0	0	0	0
11	0	0	4.5358	0	0	22261	0	0	0	0	0	0	0
12	0	0	4.5358	0	0	1267	0	0	0	0	0	0	0
13	0	0	4.5358	0	0	1267	0	0	0	0	0	0	0
14	0	0	4.5358	0	0	4389	0	0	0	0	0	0	0
15	0	0	4.5358	0	0	4389	0	0	0	0	0	0	0
16	0	0	5.3387	0	31180	0	0	0	0	0	0	0	0
17	0	0	5.3387	0	24492	0	0	0	0	0	0	0	0
18	0	0	5.3387	0	28809	0	0	0	0	0	0	0	0
19	0	0	5.3387	0	7703	0	0	0	0	0	0	0	0
20	0	0	5.3387	0	28809	0	0	0	0	0	0	0	0
21	0	0	5.3387	0	28809	0	0	0	0	0	0	0	0
22	0	0	5.3387	0	21972	0	0	0	0	0	0	0	0
23	0	0	5.3387	0	21972	0	0	0	0	0	0	0	0
24	9.537	1	21.206	5961	0	0	15257	11068	24348	27416	24801	16321	2467
25	2.852	1	21.206	32189	0	0	15257	11068	24348	27416	24801	16321	2467
26	4.183	1	21.206	517	0	0	15257	11068	24348	27416	24801	16321	2467
27	7.289	1	21.206	20307	0	0	15257	11068	24348	27416	24801	16321	2467
28	2.948	1	21.206	14907	0	0	15257	11068	24348	27416	24801	16321	2467
29	4.606	1	21.206	5396	0	0	15257	11068	24348	27416	24801	16321	2467
30	5.132	1	21.206	12277	0	0	15257	11068	24348	27416	24801	16321	2467
31	8.161	1	21.206	5825	0	0	15257	11068	24348	27416	24801	16321	2467
32	0	0	4.5358	0	0	19749	0	0	0	0	0	0	0
33	0	0	4.5358	0	0	19749	0	0	0	0	0	0	0
34	0	0	4.5358	0	0	19749	0	0	0	0	0	0	0
35	0	0	4.5358	0	0	19749	0	0	0	0	0	0	0
36	0	0	4.5358	0	0	19749	0	0	0	0	0	0	0
37	0	0	4.5358	0	0	19749	0	0	0	0	0	0	0
38	0	0	4.5358	0	0	19749	0	0	0	0	0	0	0
39	0	0	4.5358	0	0	19749	0	0	0	0	0	0	0
40	0	0	5.3387	0	19339	0	0	0	0	0	0	0	0
41	0	0	5.3387	0	19339	0	0	0	0	0	0	0	0
42	0	0	5.3387	0	19339	0	0	0	0	0	0	0	0

various requirements, as well as packet model with transmission rate and packet length. The third one models the radio resource assignment algorithms, which decide call access and / or renew an active period for an existing call. The fourth models the data transmission part. The fifth part models the release of the TFI and PDTCH(s) of the MS, which has either completed a call or has terminated a call by force. The final part is for the computation of the performance statistics of various data, and print out some major output parameters, such as: call dropping/blocking probability, network utilization efficiency, average buffer overflow, average queuing delay, and etc.

There are three types of results to compare major performance in both SRA and ARAOC algorithms. The first type of results is obtained by assuming assignment parameter (AP) of 0.5, random selection for traffic class 1-5, and the variable values of channel buffer sizes (CBS) of 4, 8, 16, 32 and 64 under the following assumptions. It is used to analyze the influence on the performance of different channel buffer sizes, and to find out the best values of channel buffer size for our simulation. Then, the second type of results is attained by assuming a channel buffer size of 16, random selection for traffic class 1-5, and the values of assignment parameter of 0.0, 0.25, 0.5, 0.75 and 1.0 with some assumptions described afterwards. The final type of results is produced by assuming a channel buffer size of 16, random selection for traffic class 1-5, different traffic classes, and with the following assumption.

The following assumptions are made in both simulations:

- Data transfer: Downlink only
- Resource traffic type: data transmission to an MS on a packet-by-packet basis
- C/(I+N) variation: Fixed during four multi-frames for each carrier

- Mobile motion: Static during four multi-frames
- Time slot transmission rate: 13.0 Kbps, which is currently used in GSM.
- Normalized sector rate load (ρ): We have defined the ρ in formula 4.22 in section 4.3.3. It is the ratio of summing average rate of all calls accepted with entire sector maximum transmission data rate. If ρ is equal to 1, it means that all channels of sector have reached to full rate transmission.
- Scheduling algorithm: packet-based first in first out (FIFO) fashion
- Traffic Class (TC): 5
- All base stations transmit with the same power. Power control is assumed ideal.
- There is perfect filtering between adjacent channels, so adjacent channels interference can be neglected.

4.6.2 SRA algorithm

In this simulation, there are six major parts (see the section of 3.6.1). The third part for resource assignment algorithms uses a static assignment algorithm, which assigns some fixed number of channels for a user according to the assignment parameter (AP) and user's traffic class during whole call. We focus on the description of resource allocation.

Figure 4.8 is the flow diagram of the channel allocation process in SRA algorithm. The brief description is as follows:

- When a new call or handoff call arrives through PRACH to access a cell, after the initial access procedure, the MS sends a detailed resource request with temporary logical link identifier (TLLI) and the requested service over a packet associated control channel (PACCH) to the BSS of this cell.

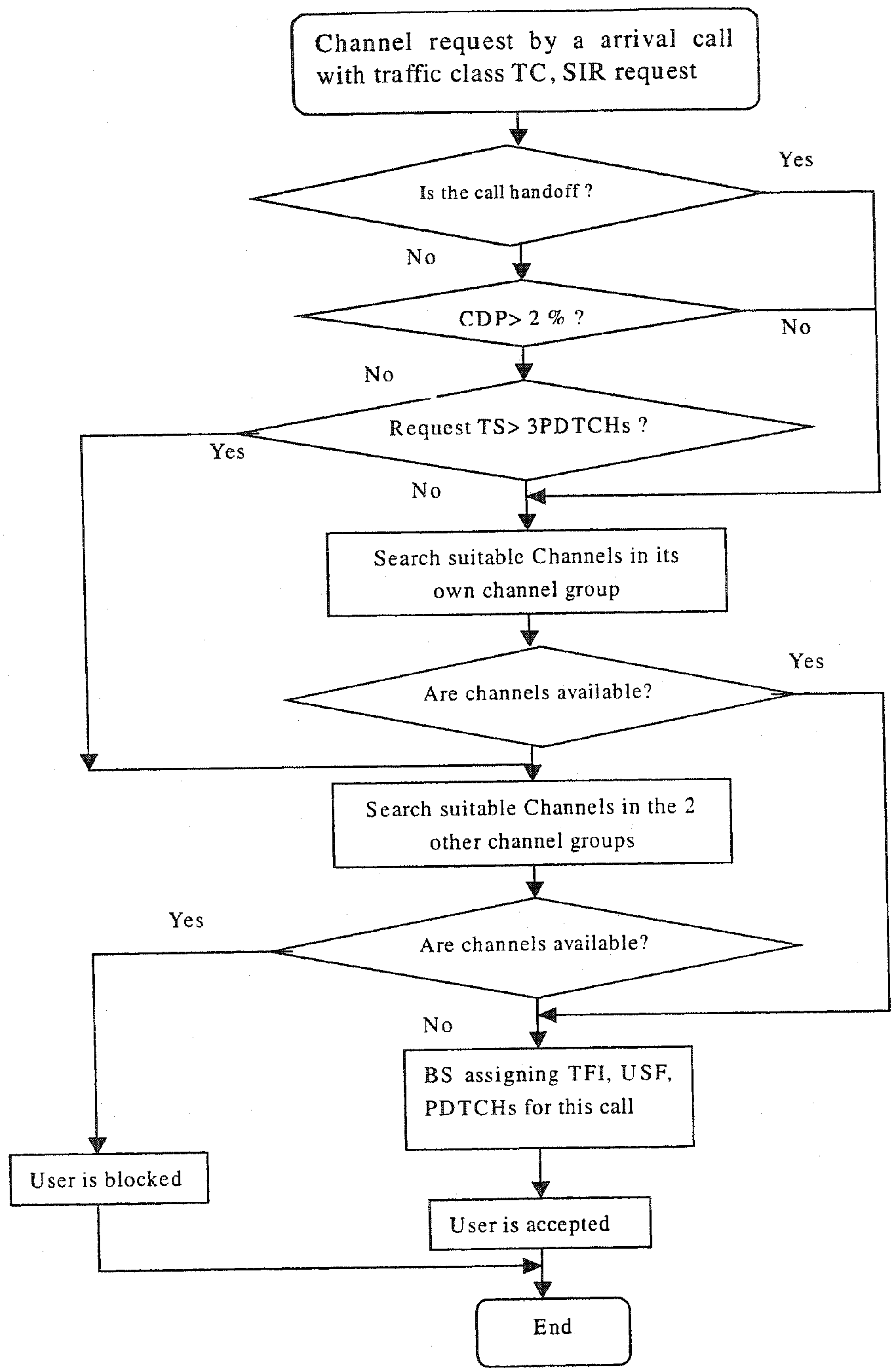


Figure 4.8 Block Diagram of SRA Channel Allocation

- If an arriving call is a new call, BSS first checks the dropping probability of handoff call (CDP) of this sector whether it is less than 2%. If it is positive, the BSS will try to find out an adequate free PDTCH(s) from its entire channel group of local sector. If it fails, it will continue searching from two other channel groups. The BSS assigns available PDTCH(s) for an arriving new call.

  However, if it is negative, the BSS will look for and assign free PDTCH(s) with less than 3 PDTCH(s) in the same way as above.
- BSS will reject a handoff call or new call if the channels in resource pool are not available for an arriving call.
- After BSS assigns certain channels to an accepted user, this user uses only those channels to transmit data.

4.6.3 Simulation description of ARAOC

In this simulation, there are also six major parts (similar with SRA in the section of 4.6.1). Except that the third part for radio resource assignment algorithms is very different, the other parts are very similar with SRA algorithms. The major differences in both SRA and ARAOC are static assignment algorithm for SRA, and adaptive assignment algorithms for ARAOC. Therefore, We focus on the description of resource allocation algorithms. The ARAOC algorithms provide the best effort service for users. It adapts between dynamic resource assignment and static resource assignment depending on the sector load factor. The BSS assigns channels according to the instantaneous transmission rate of users every four multi-frames. A session is associated with a single MS. Both the inter-arrival time and the active period follow exponential distributions.

Upon generating a new session with TFI, a search for suitable resource is made in accordance with the scheduling algorithm in operation. If the resource is found, the new session is assigned to this resource. If the resource is not found, the session is simply added to the session stack to wait for processing. In the following part, we will describe the channel allocation process with ARAOC algorithms and data transmission.

A. First, we will describe the channel allocation process with ARAOC algorithms, of which the flow diagram is shown in Figure 4.9:

- When a new call or handoff call arrives through PRACH to access a cell. After the initial access procedure, the MS sends a detailed resource request with TLLI and the requested service over a packet associated control channel (PACCH) to the BSS of this cell.
- When BSS receives a request, it first checks whether the loading factor (ρ) of the local sector is less than the designated threshold. When the factor (ρ) is less than the designated threshold (e.g., 0.9), the BSS takes a loose assignment parameter, which is called dynamic assignment policy. It establishes a temporary block flow (TBF) and distributes a TFI, PDTCH(s), and packet uplink assignment message over the packet access grant channel (PAGCH) to the BSS. The PDTCH(s) is, as much as possible, and according to user instant rate required, provide the best effort service to an arriving call.
- When the sector dropping probability of handoff call (CDP) is larger than 2%, the BSS will find the available PDTCH(s) in its entire channel group; if it is unsuccessful, it will continue searching from other two channel groups for a handoff call, or a new call with only less than 3 PDTCH(s). However, when the CDP is less than 2%, the BSS will find a requested PDTCH(s) from resource

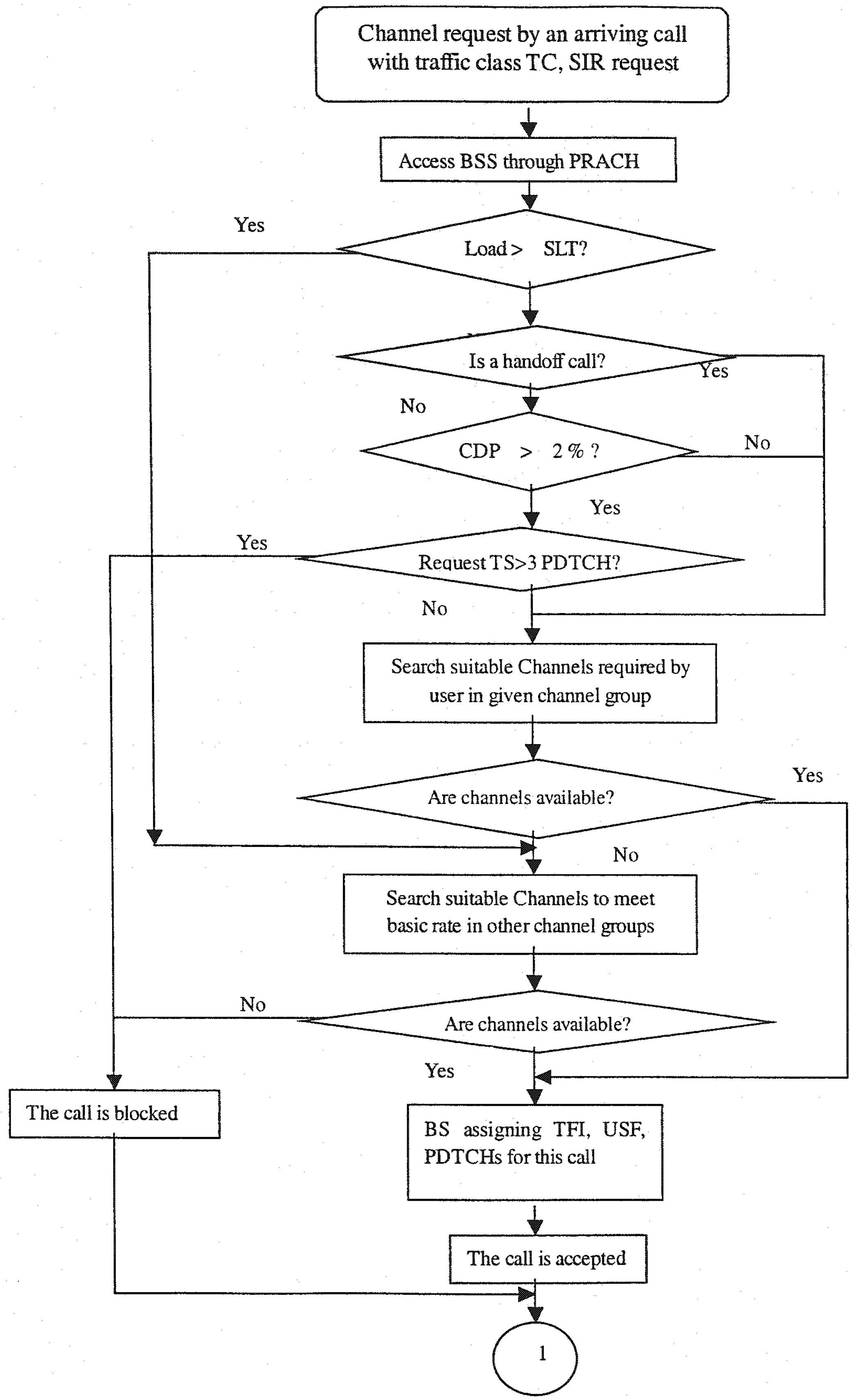
pool for all arriving calls, including handoff calls and new calls.

- When the factor ρ is larger than the designated threshold (i.e., 0.9), the BSS adapts to a tight assignment policy, and uses static assignment algorithm. It assigns the PDTCH(s) only according to the traffic class and the tight system assignment parameter (i.e., AP=0.5), so that it only guarantees the basic rate for all users. On the other hand, because BSS will calculate co-channel interfering powers from the18 nearest interfering sectors, the BSS can find suitable free channels from two other channel groups for this arriving call. If adequate free channels in a carrier can satisfy the SIR requirement of the user, the BSS will borrow and assign these channels.
- BSS will reject all of the handoff calls and new calls if the three channel groups in a resource pool are not available for them.

B. At the end of each active period (i.e., no more data to send): The BSS will detect the content of every user buffer at each repetition. When the content is empty for 3 iterations consecutively, the MS releases only its PDTCH(s) and maintains its TFI. In addition, when it does not have an ongoing downlink data transfer, the MS only uses the fast downlink control channel.

C. At the beginning of each new period of activity: The MS accesses the system using its TFI over F-PRACH along with the specific access probability and procedure. If the BSS has a conditional ongoing downlink data transfer, it may receive a PDTCH distribution via an assignment message sent on either the F-PAGCH or a PACCH. In our simulation, we have assumed that all the above control channels are perfect.

D. During the data transmission in progress, each MS, already assigned some PDTCH(s) and a unique TFI, listens to its set of assigned downlink channels and only accepts RLC blocks with its TFI. A packet generator can produce two types of packets, bursty and streaming. When a set of packets is generated, the BSS puts the packets into the user's buffer. When the number of packets exceeds the user's buffer size, the value of buffer overflow is incremented by 1. When the buffer occupancy exceeds an assault threshold (Ta), a congestion assault message is being sent to the user, and subsequently, the BSS shuts down its transmission rate. When the buffer occupancy declines below the reduction threshold (Tr), a congestion reduction message is being sent to the user, and consequently, the BSS increases its transmission rate. The BSS transmits packets to the MS through the assigned time slots. The number of packets successfully sent will be counted as the user's throughput; otherwise, the packets that remain in buffer will be counted as its queuing delay. Therefore, the BSS can communicate with a given MS on any of the channel assigned.


Channel request by an arriving call with traffic class TC, SIR request
Access BSS through PRACH
Yes
Load > SLT?
Is a handoff call?
Yes
No
CDP > 2 % ?
No
Yes
Yes
Request TS>3 PDTCH?
No
Search suitable Channels required by user in given channel group
Yes
Are channels available?
No
Search suitable Channels to meet basic rate in other channel groups
No
Are channels available?
Yes
The call is blocked
BS assigning TFI, USF, PDTCHs for this call
The call is accepted
1

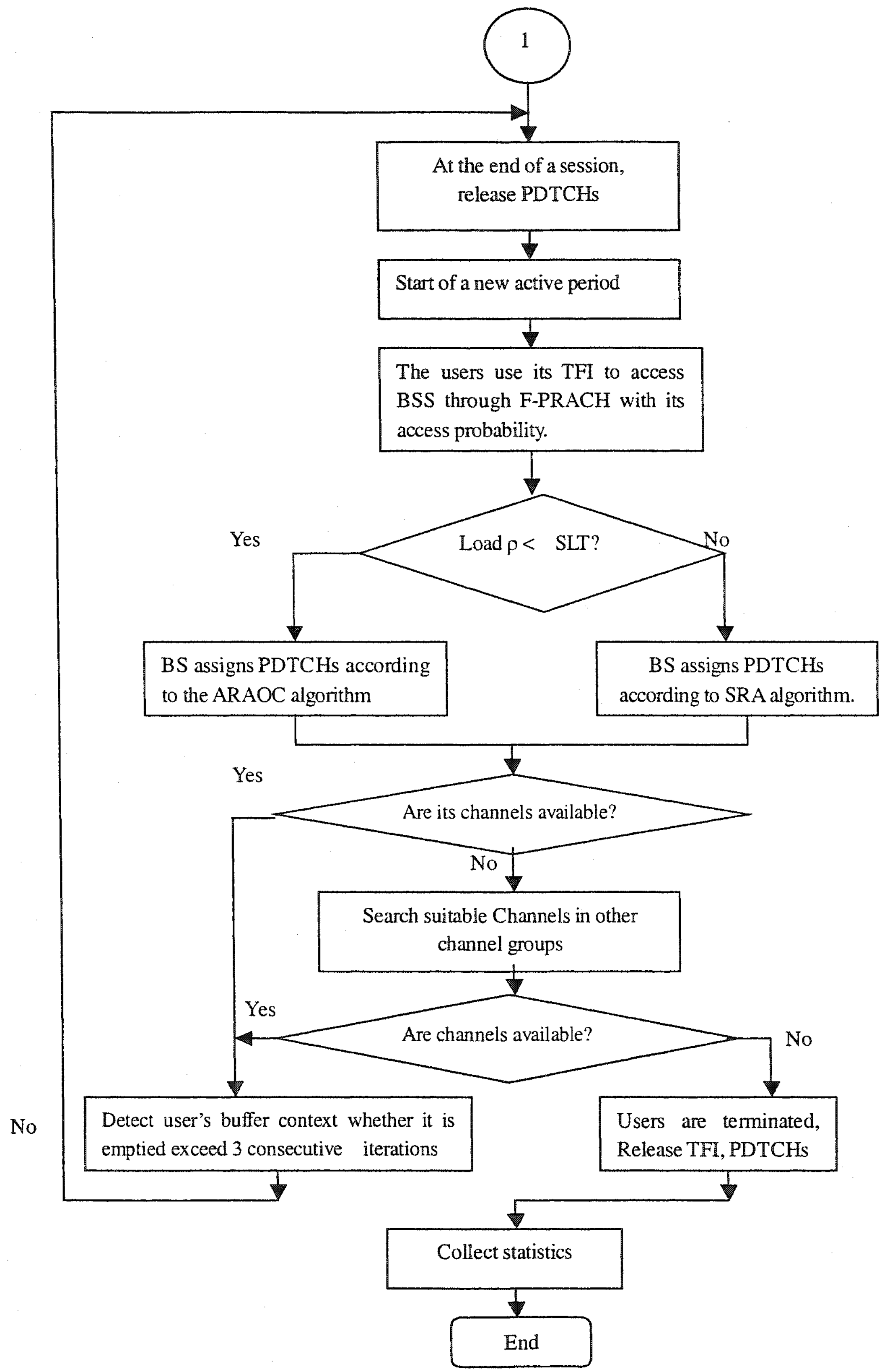


Figure 4 .9 Block Diagram of ARAAOC Channel Allocation

4.7 Discussion on Simulation Results in SRA and ARAOC algorithms

To investigate the system capacity and performance as well as the QoS of users, we will now discuss the simulation results in different situations. In doing so, our simulation is being divided into 3 parts. The first part is to analyze the influence brought upon by different channel buffer sizes; the second part is to analyze the effect from different assignment parameters; the last part is to discuss the influence of various traffic classes.

In all simulations, some fixed parameters are assumed: the channel data rate (**CDR**) is 13.0 Kbps; the call handoff rate (**CHR**) in all arriving calls is 30%; each sector rate load threshold (**SLT**) for ARAOC is set at 90%; each sector load for SRA (**SLL**) is limited by a call dropping probability (**CDP**) of 2%; For two thresholds of buffer occupancy: assault threshold (**Ta**) is set at 80%, and reduction threshold (**Tr**) is set at 20%; the number of the iteration program (**NIP**) is 104000 times; Required Minimum SIR (**RMS**) for SRA reach above 12 dB; Capable Average SIR (**CAS**) for ARAOC can reach 21 dB; and each sector (**CFN**) consists of 41 carriers.

4.7.1 Analysis of the influence of Channel Buffer Sizes on performance

In this simulation, there are 3 assumptions: assignment parameter is AP = 0.5; a traffic class (**Rtype**) is generated randomly with uniform distribution. The influence of different channel buffer sizes is discussed below:

Figure4.11 and Figure4.10 show that the Average Buffer Overflow (ABO) relates to different channel buffer sizes and load. From observation of the above two Figures, the bigger channel buffer size is assigned, the lesser is the ABO. The ABO in Figure4.11 is under 2.5% when traffic load is less than 80%, whereas it is less than 4.5% when loading at 90% for all channel buffer sizes. Especially, when a channel buffer size is equal to 16, ABO is less than 1.5% under 80% load, and less than 4% at 90% load. On the other hand, Figure4.10 shows that ABO is large and of different value from 6.5% to 12% with different channel buffer sizes. In particular, ABO is about 10% at a channel buffer size of 16.

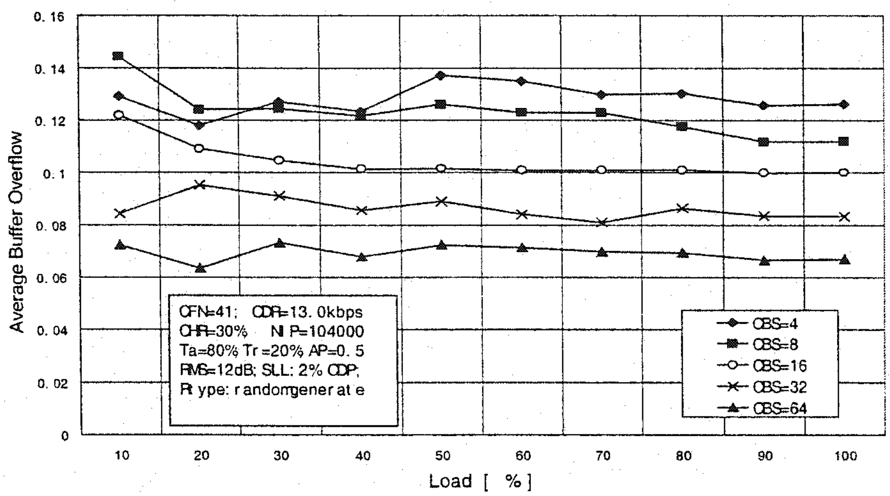


Figure 4.10 Average Buffer Overflow in SRA vs. Load with Channel Buffer Sizes (CBS)

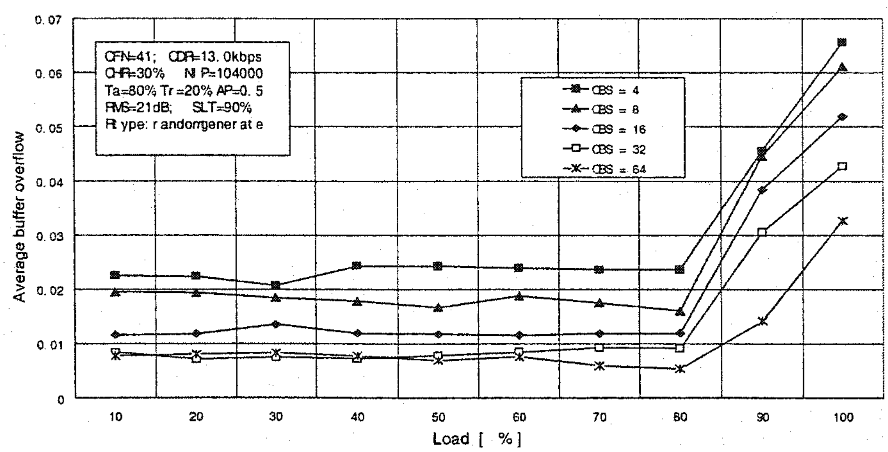


Figure 4.11 Average Buffer Overflow in ARAOC vs. Load with CBS

Figure4.12 and Figure4.13 illustrate that Average Queuing Delay (AQD) increases as channel buffer size increases. Figure4.12 shows that AQD changes widely due to channel buffer size over 16 (i.e., about 16 packets for a channel buffer size of 16, and about 40 packets for a channel buffer size of 32). As in the similar case in Figure4.13, AQD is about 18 packets for a channel buffer size of 16, and 40 packets for a channel buffer size of 32.

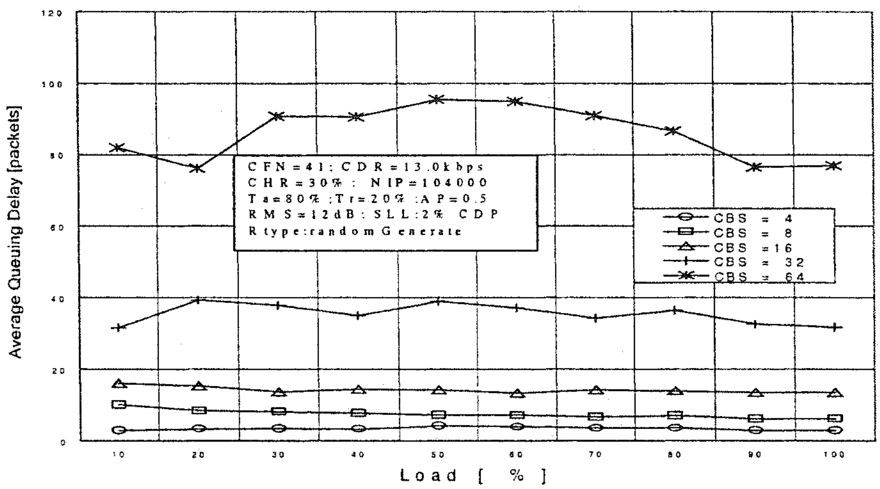


Figure 4.12 Average Queuing Delay in SRA vs. Load with CBS

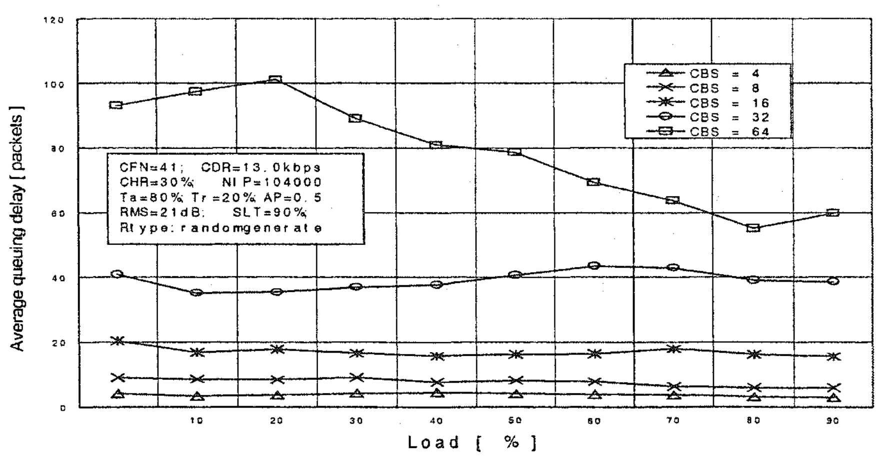


Figure 4.13 Average Queuing Delay in ARAOC vs. Load with CBS

Figure4.14 and Figure4.15 further demonstrate the relationship of average buffer overflow, channel buffer size and load. The Variance of average Buffer Overflow (VBO) in Figure4.14 shows that VBO is approximately 0.016 for the channel buffer size in the range of 4-32, and changes from 0.032 to 0.022 for a channel buffer size of 64; whereas, the VBO in Figure4.15 is very small. It is less than 0.0006 at a load of 80%, separately less than 0.0025 at 90% load for all channel buffer sizes. Especially, the VBO is equal to 0.0002 when a channel buffer size is 16 and the load is under 80%.

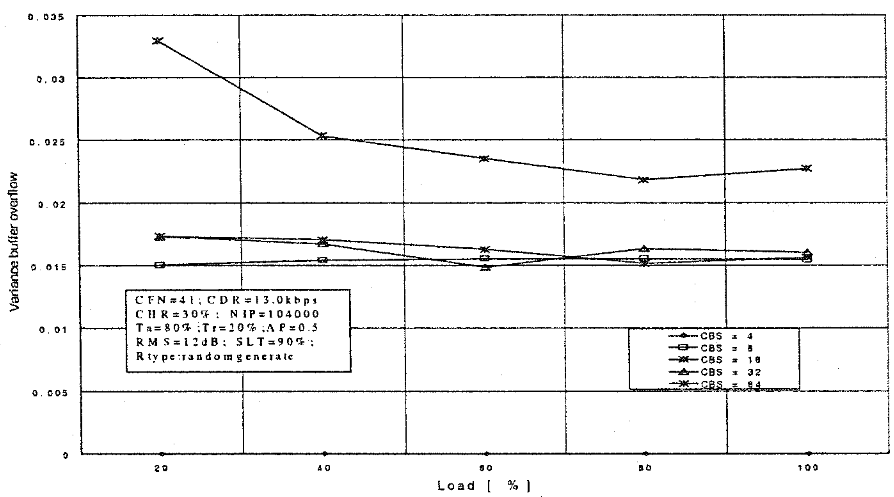


Figure 4.14 Variance Buffer Overflow in SRA vs. Load with CBS

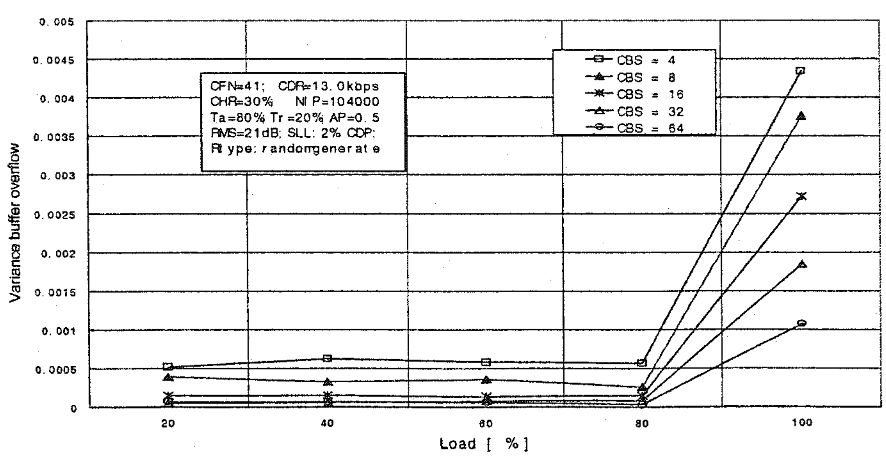


Figure 4.15 Variance Buffer Overflow in ARAOC vs. Load with CBS

In summary, from the observation of the above figures of SRA and ARAOC, the bigger channel buffer size has resulted in a smaller ABO and a larger AQD. Especially, when a channel buffer size exceeds 16, the ABO and AQD are changed quickly. Thus, we should find a balancing point between ABO and AQD to satisfy the user requirements of QoS for low packet loss rate and delay. From Figure4.10—Figure4.18, we have seen that the value of channel buffer size 16 for ABO, VBO, AQD, VQD and AUT is the best choice as a balanced point for both SRA and ARAOC algorithms. When a channel buffer size is equal to 16, the AQD and VQD for the both algorithms have fewer differences, whereas the ABO and VBO in ARAOC are much better in value than those in SRA. Therefore, a channel buffer size of 16 is the best selection for the above mentioned parameters, and the results of ARAOC are much better than those in SRA. For example: for a channel buffer size of 16 with similar AQD and VQD, the ABO in ARAOC is less than 1.5% in situation under 80% load, and less than 4% at 90% load; whereas, the ABO in SRA is around 10% for various loads.

4.7.2 Analysis of the effect of assignment policy on performance

From the above analysing results for channel buffer size, we assume that channel buffer size (**CBS**) is 16, and a rate type (**Rtype**) is obtained randomly by calling a uniform distribution. The influence of different assignment parameters is analyzed.

Figure4.16 shows the dropping probability of handoff call (CDP) is 0.024, 0.035, 0.05, 0.055 loaded at 70%, 80%, 90%, and 100% respectively for an assignment policy of 0.75. To satisfy the system requirement of 2% CDP, a sector loads 65%, 68%, 93%, 100% for assignment parameters of 1.0, 0.75, 0.5, below 0.25, respectively. Thus, an assignment parameter of 0.5 can satisfy the required system performance.

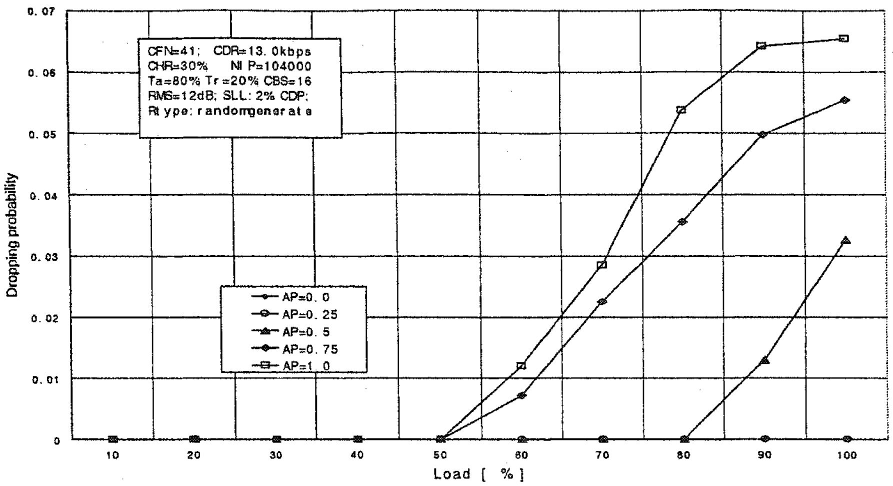


Figure 4.16 Call Dropping Probability in SRA vs. Load with AP

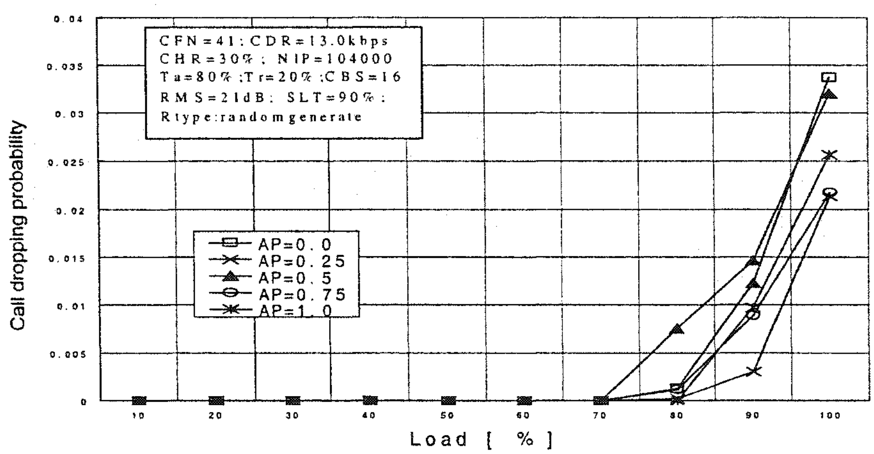


Figure 4.17 Call Dropping Probability in ARAOC vs. Load with AP

Figure4.17 shows that the CDP equals to zero for all assignment parameters if the sector load is less than 70%. The CDP is less than 1.5% at 90% load for all assignment parameters. The CDP is less than 2% at 93% load for an assignment parameter of 0.5. Since a sector system does not count the 21 channels as signaling transmission, 93 % is considered as a full load. Therefore, the ARAOC in CDP is much better than SRA.

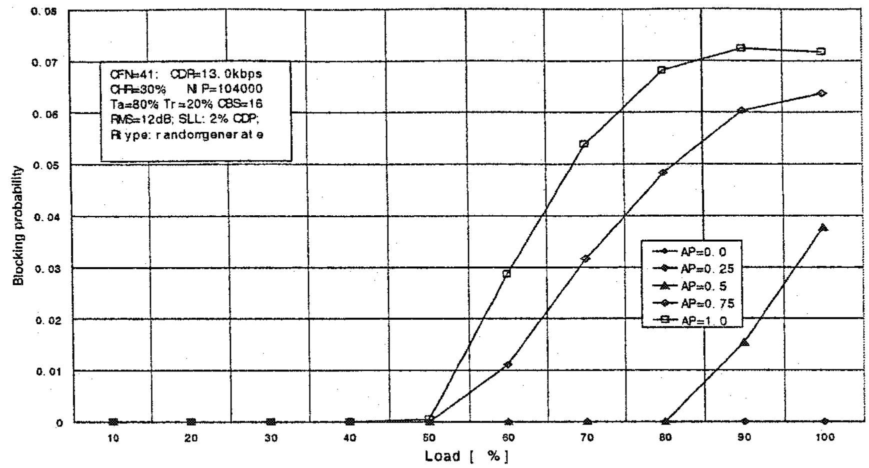


Figure 4.18 Call Blocking Probability in SRA vs. Load with AP

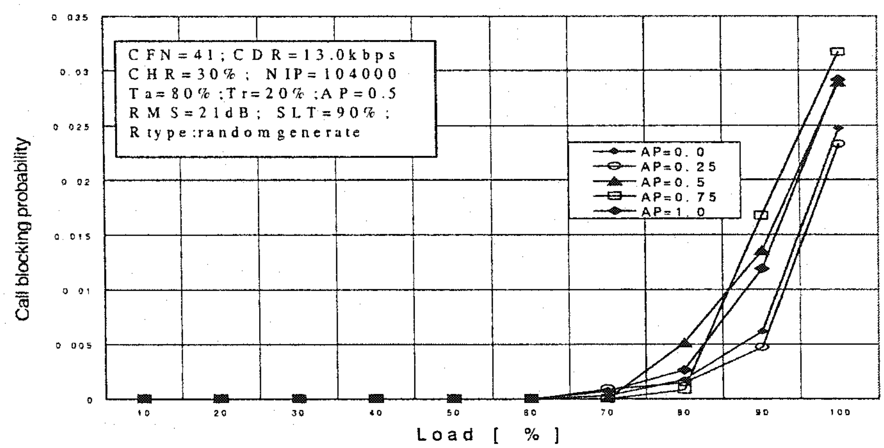


Figure 4.19 Call Blocking Probability in ARAOC vs. Load with AP

Figure4.18 and Figure4.19 illustrate that a blocking probabilities of a new call (CBP) is escalating as assignment parameters increased and the sector load reached up to a certain value. Figure4.18 shows that CBP values are 0.068, 0.048 and 0.0 at 80% load, as well as value of 0.072, 0.062, 0.02 and 0.0 at 93% load corresponding to assignment parameters of 1.0, 0.75, 0.5 and below 0.25. Thus, assignment parameters largely affect the CBP. When an assignment parameter is 0.5, the CBP and load factor can satisfy the system requirement. Figure4.19 shows that the value of CBP rises as the assignment parameter increases. The CBP for all assignment parameters are less than 0.005 at 80% load, and less than 0.022 at 93% load. The sector load limited by CBP of 2% is up to 95% for an assignment parameter of 0.5.

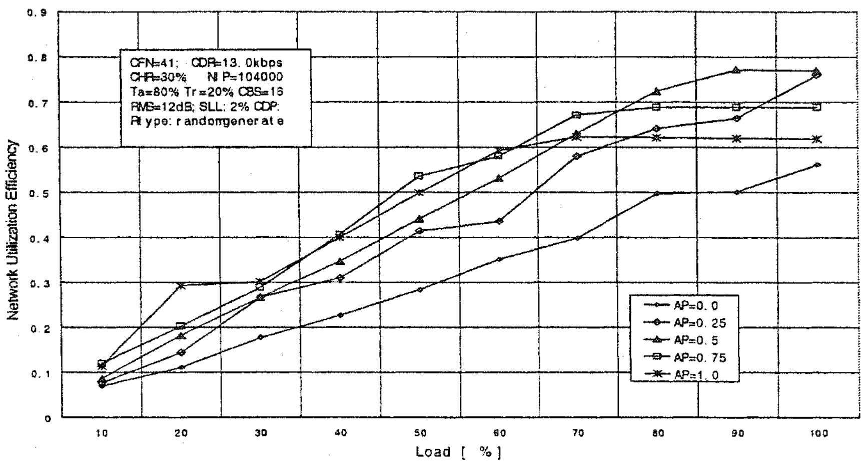


Figure 4.20 Network Utilization Efficiency in SRA vs. Load with Assignment parameter (AP)

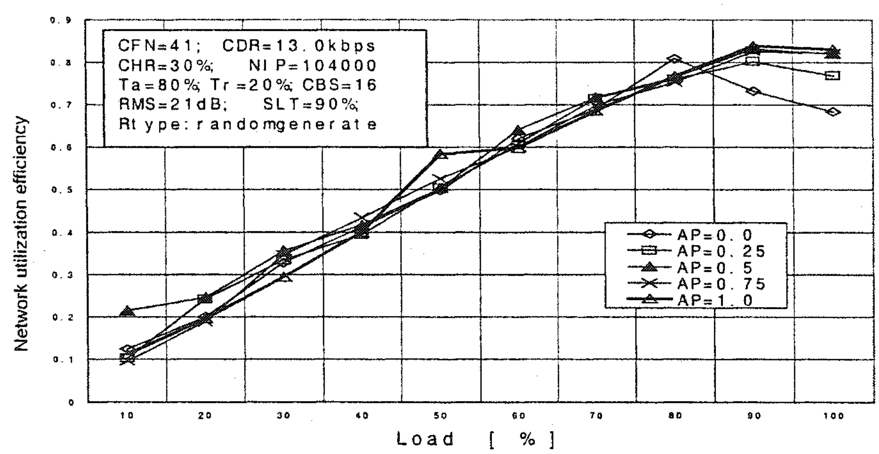


Figure 4.21 Network Utilization Efficiency in ARAOC vs. Load with AP

Figure4.20 and Figure4.21 illustrate that increasing the sector load is raising the value of Network Utilization Efficiency (NUE). Figure4.20 shows that an assignment parameter of 0.5 can obtain the best NUE (up to 78%) in all assignment parameters, and NUE rises as assignment parameter increases. The value of NUE in Figure4.21 can increase to 83%. Thus, the value of NUE in ARAOC is at least 5% higher than that of SRA.

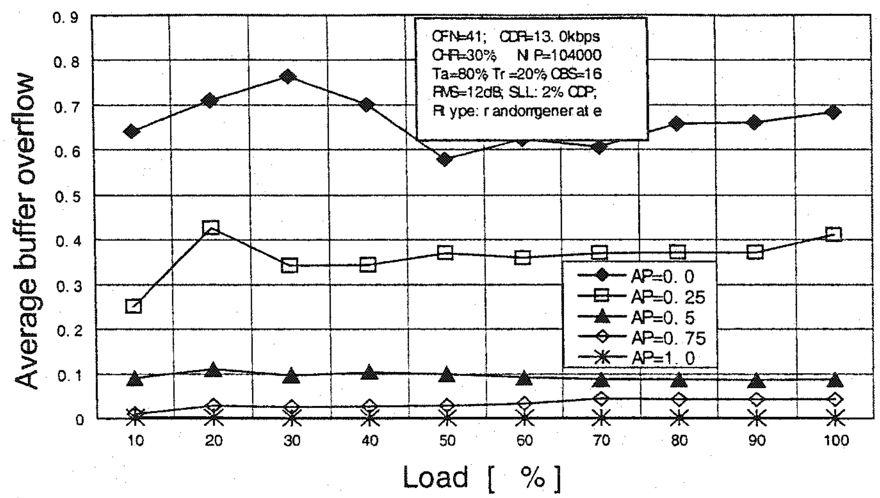


Figure 4.22 Average buffer overflow in SRA vs. Load with AP

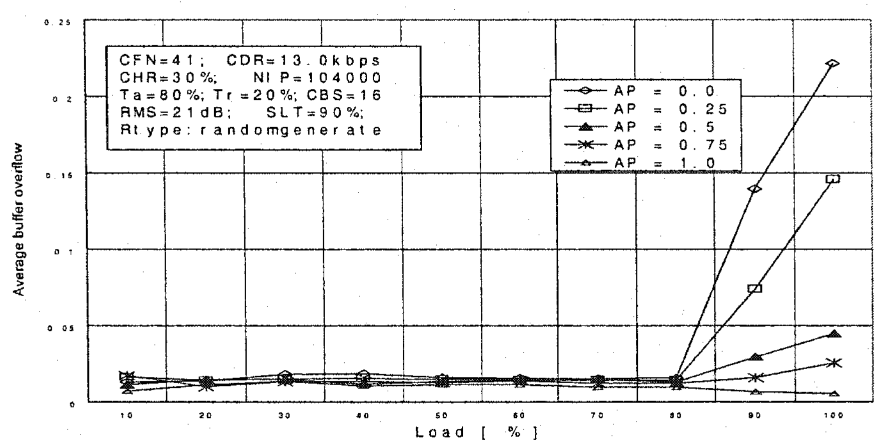


Figure 4.23 Average buffer overflow in ARAOC vs. Load with AP

Figure4.22 shows that the value of Average buffer overflow (ABO) increases from 0 to 0.76 as assignment parameters decreased. When assignment parameter is at 0.25 or 0.0, the value of ABO is very large, resulting in many packet losses or retransmissions. A value of ABO less than 10% for an assignment parameter of 0.5 is acceptable. With an assignment parameter of 0.5, the ABO, as shown in Figure4.23, is very small, and less than 0.02 under 80% load, and 0.05 at full load.

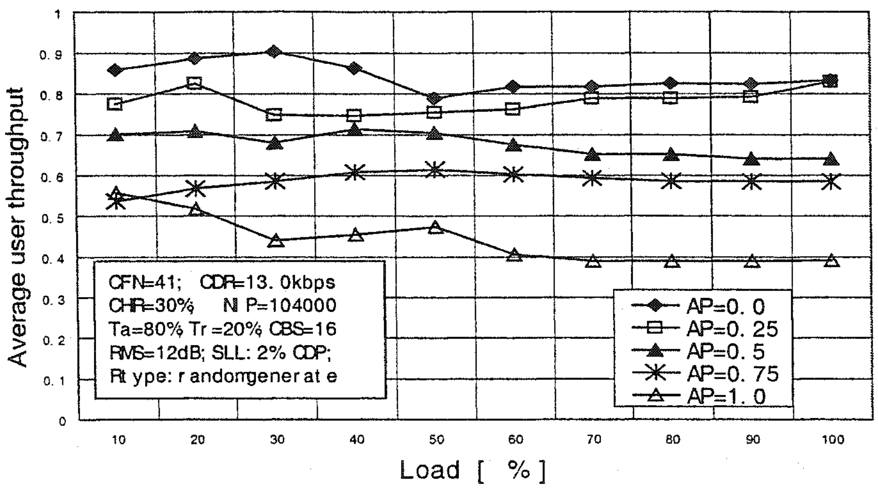


Figure 4.24 Average User Throughput in SRA vs. Load with AP

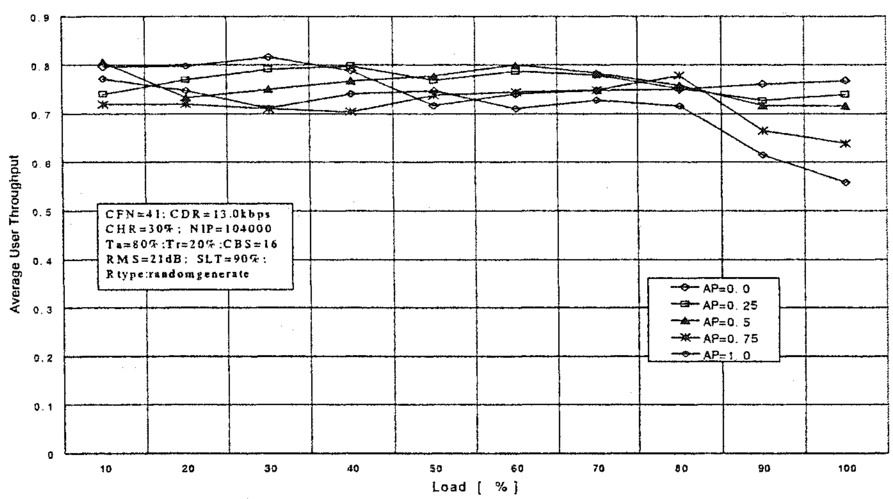


Figure 4.25 Average User Throughput in ARAOC vs. Load with AP

Figure4.24 and Figure4.25 show the relationship along with Average User Throughput (AUT), assignment parameters and load. From the above observations, the value of AUT in Figure4.24 decreases as the AP increases. It rounds to 68% at AP of 0.5. When the sector load exceeds 80%, the value of AUT in Figure4.25 rounds to 75% for all assignment parameters, it decreases as AP increases.

From Figure4.20 to Figure4.25 above, we find that assignment parameters in SRA algorithm have a significant influence on the system performances and QoS, whereas the influence on ARAOC algorithm is not obvious when loading is less than 80%.

In SRA algorithm, when the assignment parameter goes up, CDP and CBP will increase, but the ABO, AQD and AUT will decrease. In order to satisfy the system requirement for a CDP of 2%, the sector loads 65%, 68%, 93%, and 100% by an assignment parameter of 1.0, 0.75, 0.5 and below 0.25 respectively. Thus, an assignment parameter of 0.5 can bring us the following results: the sector load of 93%; a CBP of 2.5%; best network utilization efficiency (i.e., up to 78%); average buffer overflow about 10%; average queuing delay around 12 packets, and AUT about 63%.

In contrast to SRA, ARAOC has a much better system performance and QoS. Under the condition of CDP of 2%, the ARAOC, with assignment parameter 0.5, can achieve up to 93% for the sector load, 2% for the CBP, 84% for the NUE, 3.5 % for the ABO, 14 packets for the AQD, and 72% for the AUT.

In conclusion, all the above parameters excluding AQD in ARAOC are superior to those in SRA. Furthermore, in order to obtain the best results, the value of assignment parameter of 0.5 will be chosen in the following simulations.

4.7.3 Discussion of the influences of traffic classes on performances

From the analysis above, we choose channel buffer size 16, and assignment parameter 0.5 in the simulations of this part. The influence of various rate types (traffic class) is discussed below.

A. Discussion on the influence of different traffic classes to system performance

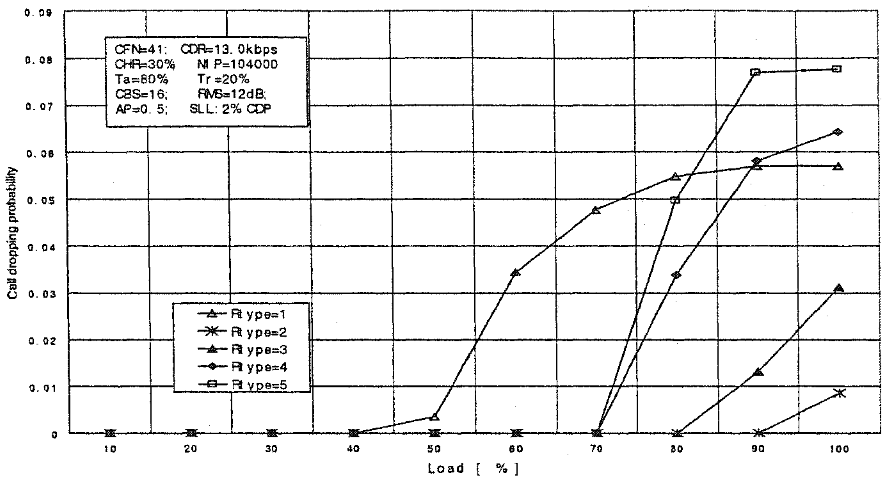


Figure 4.26 Call Dropping Probability in SRA vs. Load with Rate type (Rtype)

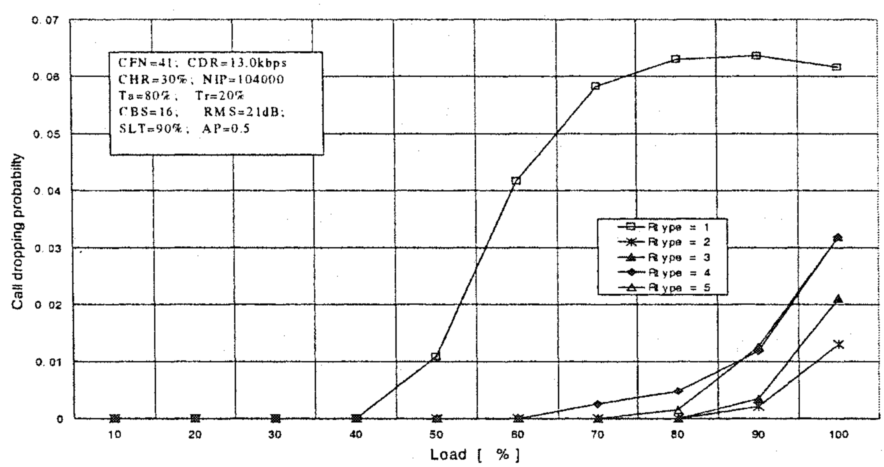


Figure 4.27 Call dropping probability in ARAOC vs. Load with Rate type (Rtype)

Figure4.26 and Figure4.27 show that the call dropping probability increases when the sector loading goes above a certain value in the two algorithms. In Figure4.26, for traffic class 1, 4 and 5, the CDP is separately 5.5%, 3.5% and 5% at 80% load, separately 5.7%, 6% and 7.8% at 94% load. We can see that the CDP of ARAOC in traffic classes of 2-5 is far less than that of SRA, and less than 2% requirement at 95% load. However, video streaming users for class 4 and 5 in SRA are dropped to 6%-8%.

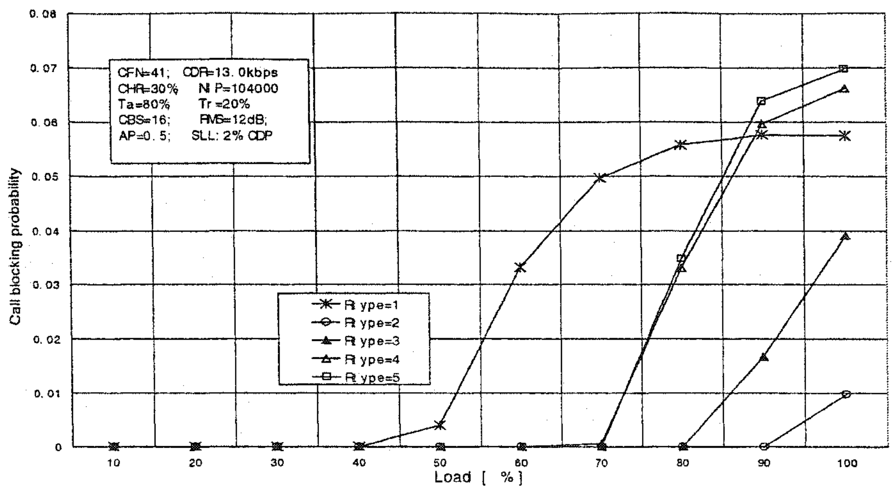


Figure 4.28 Call Blocking Probability in SRA vs. Load with Rate type (Rtype)

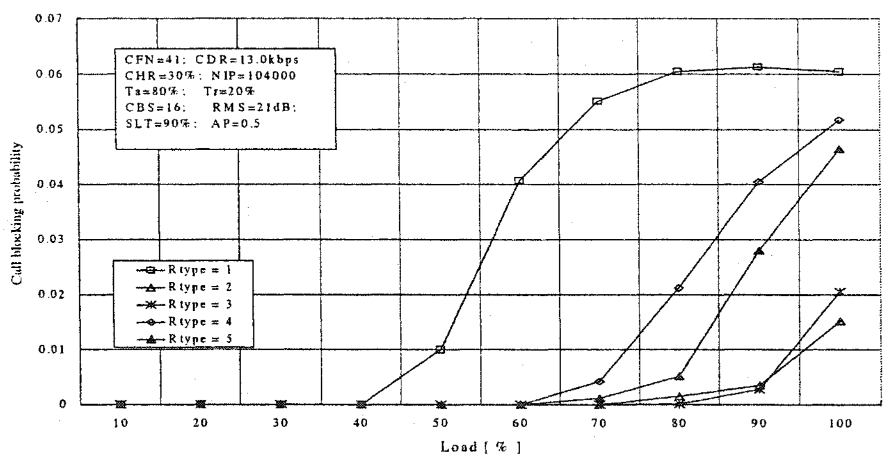


Figure 4.29 Call Blocking Probability in ARAOC vs. Load with Rate type (Rtype)

Figure 4.28 and Figure 4.29 in ARAOC indicate that the best balancing point between call dropping probability and call blocking probability. That is to satisfy the system requirement for CDP of below 2% at full load, and to make the CBP as small as possible.

In contrast, both CDP and CBP in SRA for streaming class are larger than 6%. Since a traffic class of 1 takes low transmission rate between 2.5 Kbps and 10.0 Kbps, its average rate is 6.75 Kbps lower than the channel rate (13.0Kbps). Thus, the CDP/CBP of traffic class 1 in both algorithms are large.

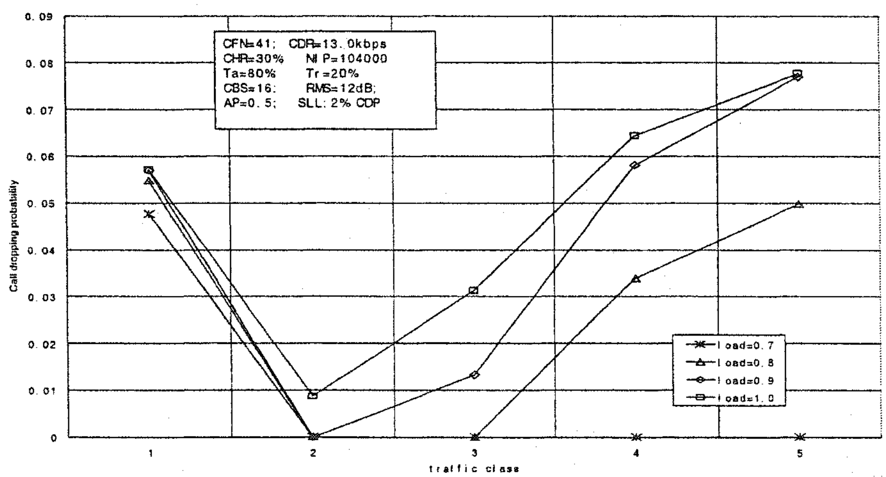


Figure 4.30 Call Dropping Probability in SRA vs. Traffic class with load

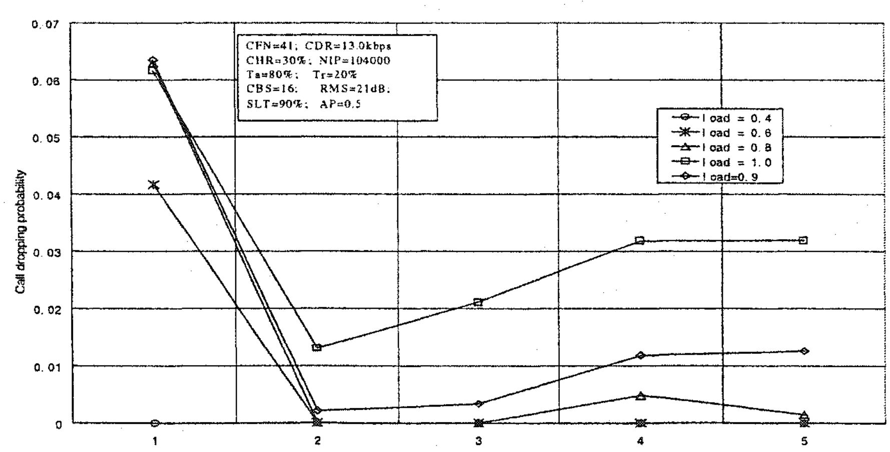


Figure 4.31 Call Dropping Probability in ARAOC vs. Traffic class with load

In addition, Figure4.30 and Figure4.31 clearly illustrate that CDP changes with the traffic class. The values of the CDP in ARAOC are much better than those in SRA.

Consequently, both CDP and CBP in two algorithms are affected greatly by the rate types of 1,4 and 5. In addition, the ARAOC can greatly lower both CDP and CBP for the traffic classes of 2-5, and improve the system capacity.

B. Discussion of the influence of traffic classes on system performance

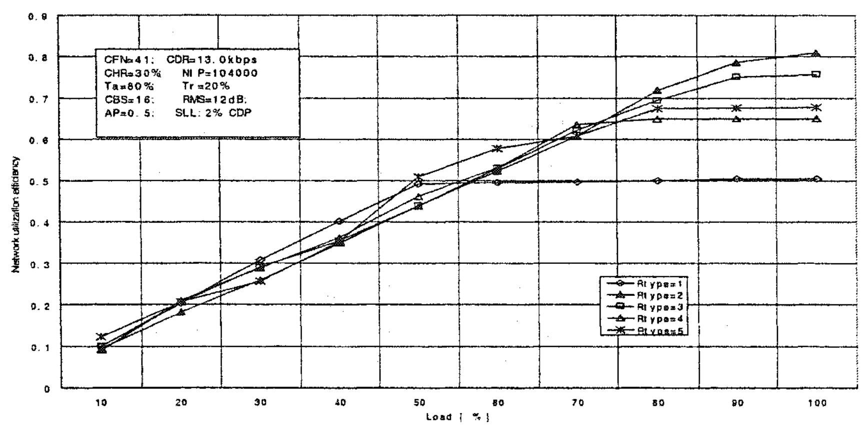


Figure 4.32 Network Utilization efficiency in SRA vs. Load with Rate type (Rtype)

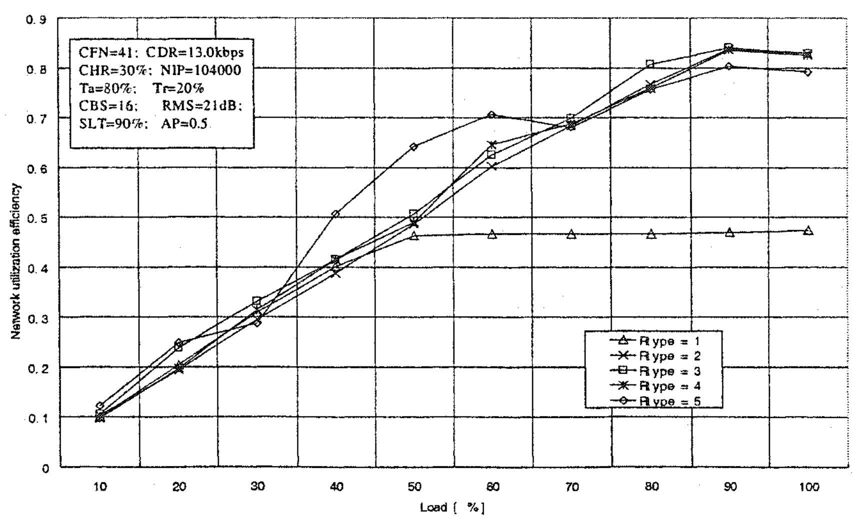


Figure 4.33 Network Utilization efficiency in ARAOC vs. Load with Rate type (Rtype)

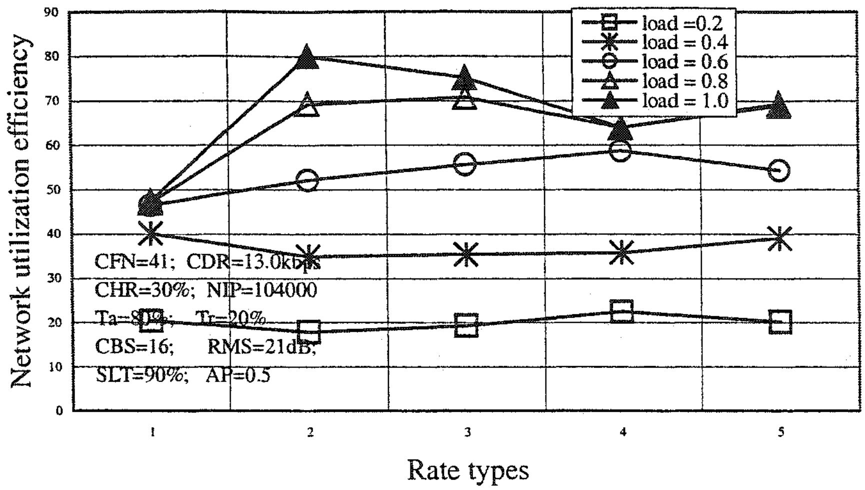


Figure 4.34 Network Utilization Efficiency in SRA vs. Traffic with Load

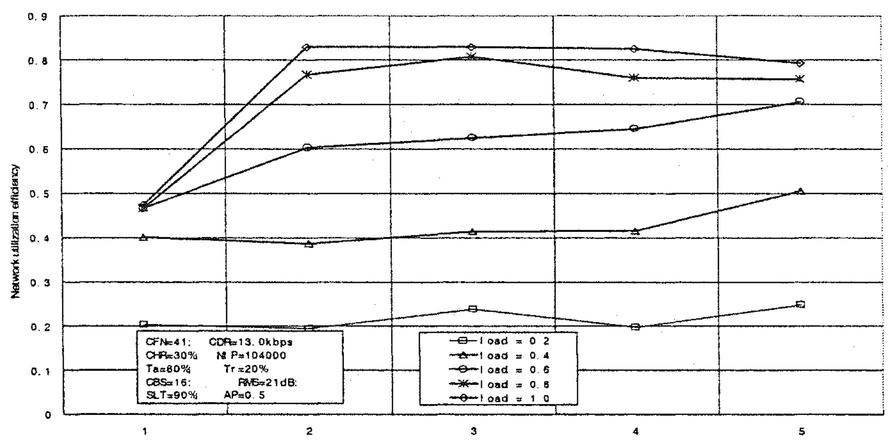


Figure 4.35 Network Utilization Efficiency in ARAOC vs. Traffic with Load

Figure4.32 to Figure4.35 show the comparison of the Network Utilization Efficiency (NUE) in SRA and ARAOC. Clearly, the values of NUE for all classes increase when the sector load increases, and the values in ARAOC are better than those in SRA. Here, a curve is obtained by loading only a traffic class. Since a traffic class of 1 takes low transmission rate between 2.5 Kbps and 10.0 Kbps, its average rate is 6.75 Kbps lower than channel rate (13.0Kbps). Thus, NUT of class 1 could not exceed 50%.

C. Comparison of the Average User Throughput (AUT) in both SRA and ARAOC

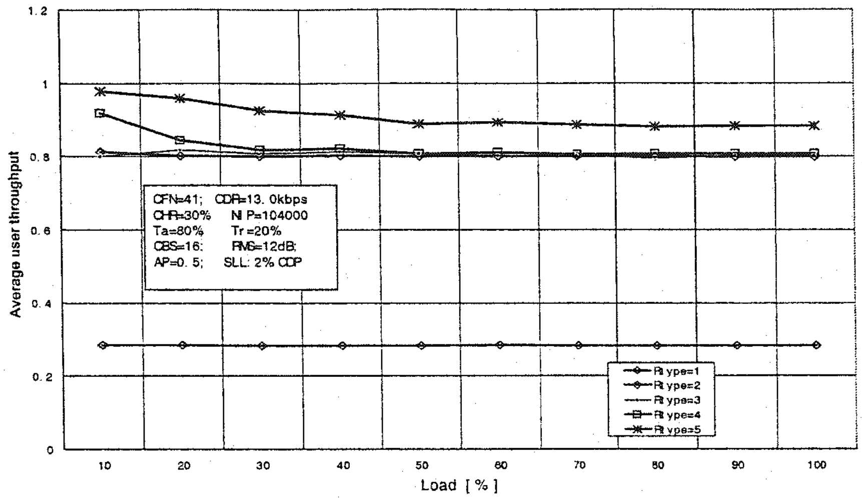


Figure 4.36 Average user throughput in SRA vs. Load with Rate type (Rtype)

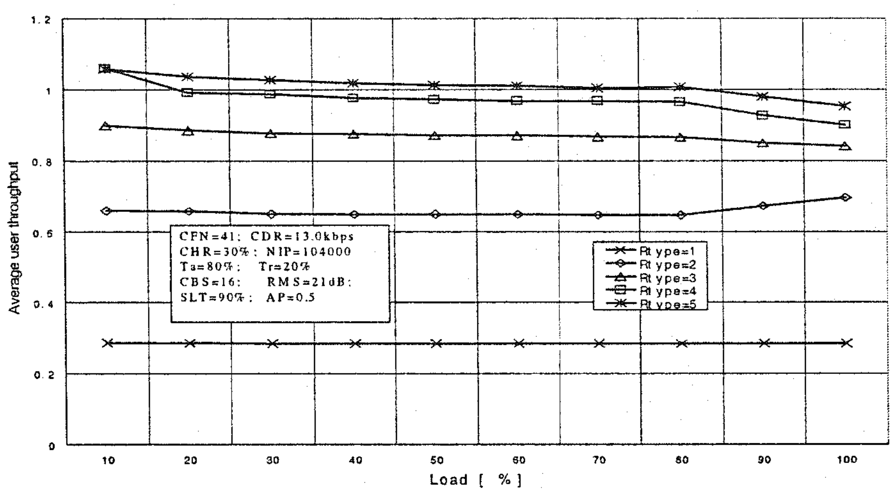


Figure 4.37 Average user throughput in ARAOC vs. Load with Rate type (Rtype)

Figure4.36 - Figure4.39 show the comparison of Average User Throughput (AUT) vs. load with various traffic classes in both SRA and ARAOC. The AUT is not influenced by loading factor, but it rises as traffic class increases. The average rate of traffic class 1 is less than the transmission rate in each time-slot; consequently, the AUT

of class 1 is low in both SRA and ARAOC. In ARAOC, since the assigned number of channels is based on the user data rate, the AUT rises as the traffic class increases. It is equal to 30%, 65%, 88%, 98% and 100% for traffic classes of 1- 5 respectively. In contrast, the AUT in SRA is fixed at 30%, 80% and 100% for traffic classes of 1, 2-4 and 5 respectively. Since a traffic class of 1 takes low transmission rate between 2.5 Kbps and 10.0 Kbps, its average rate is 6.75 Kbps lower than the channel rate (13.0Kbps). Therefore, the AUT of traffic class 1 for the both is very small.

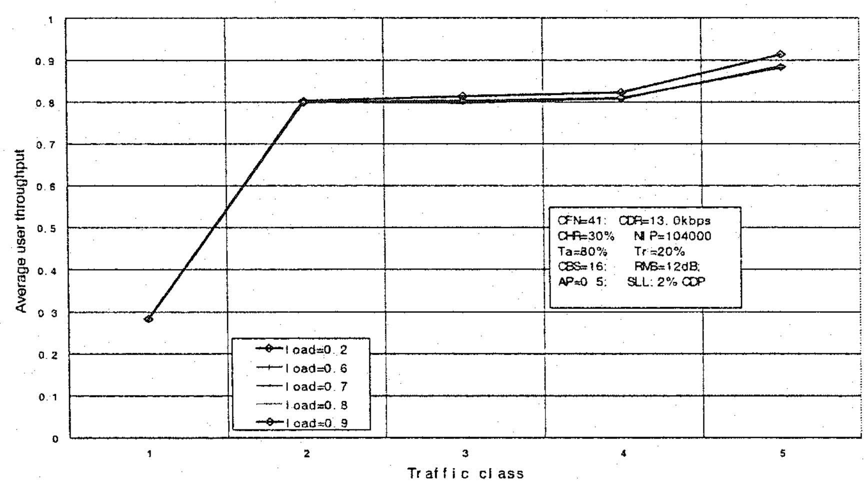


Figure 4.38 Average user throughput in SRA vs. Traffic class with Load

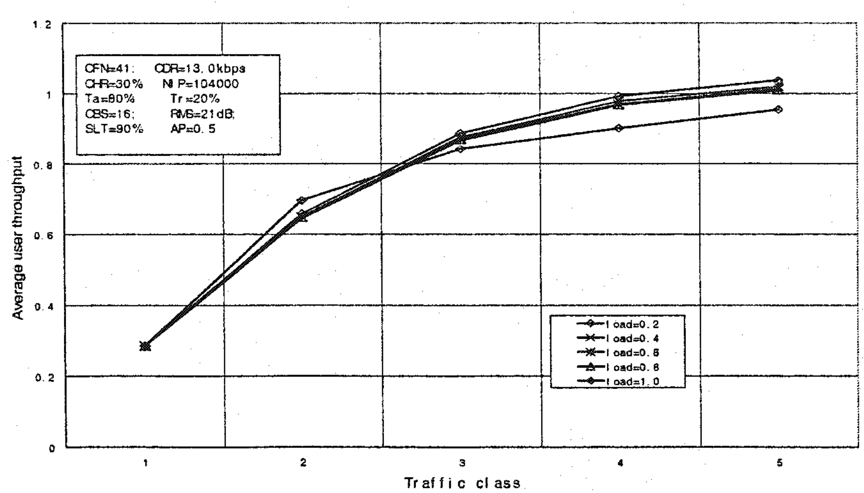


Figure 4.39 Average user throughput in ARAOC vs. Traffic class with Load

D. Discussion of the influence of traffic classes on user performance in both algorithms

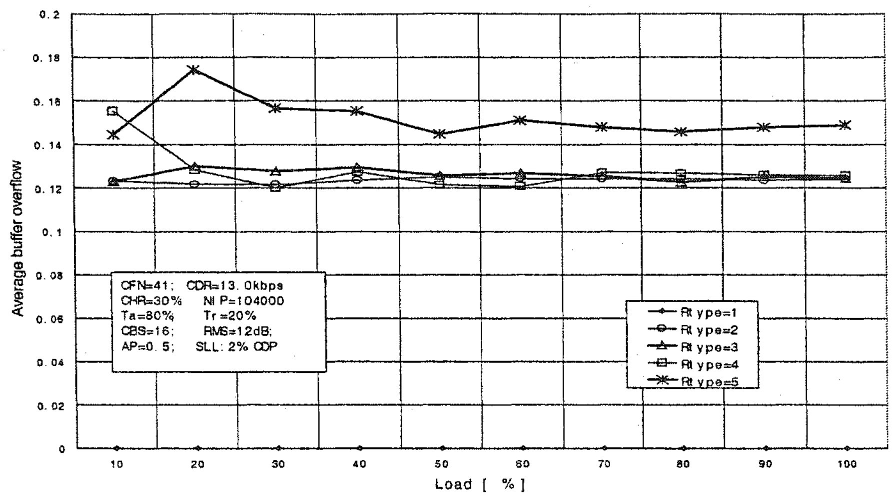


Figure 4.40 Average buffer overflow in SRA vs. Load with Traffic class

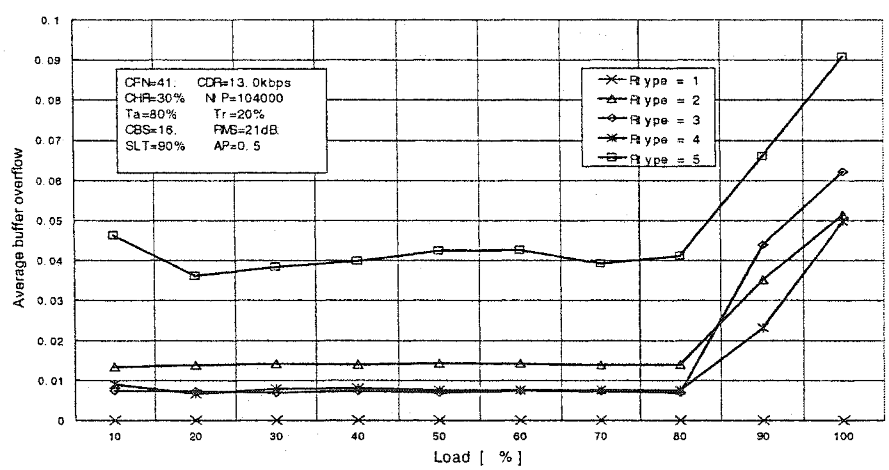


Figure 4.41 Average buffer overflow in ARAOC vs. Load with Traffic class

Figure4.40 to Figure4.41 show the comparison of Average Buffer Overflow (ABO) vs. Load in both SRA and ARAOC respectively. When ABO and VBO increase, they result both in bigger error ratio for real-time call, and in larger retransmission ratio for non real-time traffic. When the sector load increases in SRA, the ABO is around 12.5% for class 1 to 4, and around 15% for class 5. In ARAOC, when the sector load increases up to 80%, the ABO increases slightly to 1.5% for class 1 to 4, and 4% for class 5. When the sector reaches its full load (94%), the ABO is below 4%, 5%, and 7.5% for traffic class 2 & 4, 3, and 5, respectively. Therefore, ARAOC is much better than SRA in satisfying the requirements for packet loss and error rate.

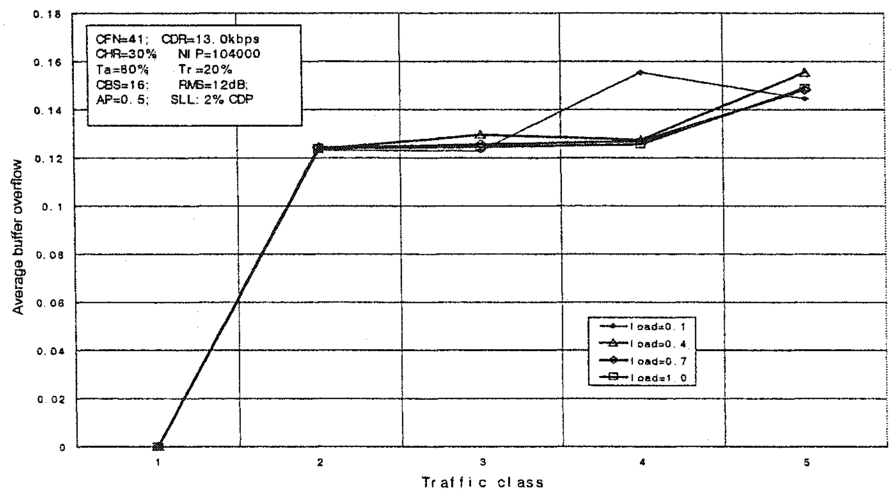


Figure 4.42 Average buffer overflow in SRA vs. Traffic class with Load

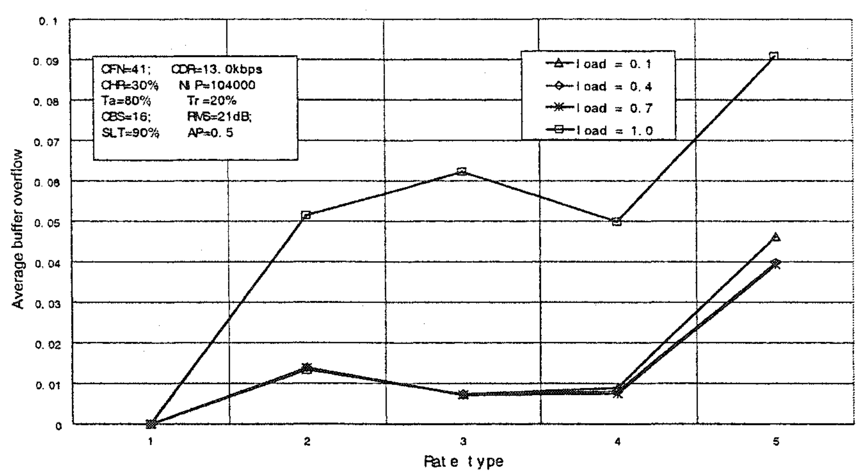


Figure 4.43 Average buffer overflow in ARAOC vs. Traffic class with Load

Figure4.42 and Figure4.43 show the comparison of Average Buffer Overflow (ABO) vs. traffic class in both SRA and ARAOC, respectively. From the observation, the

ABO of ARAOC is much better than that that of SRA. Especially, when the sector load is below 80% for ARAOC, the ABO is under 2% for traffic class 1 to 4, as well as 4.5% for the class5. The ABO can satisfy the requirements of the system performance.

E. Discussion of the influence of traffic classes on user QoS

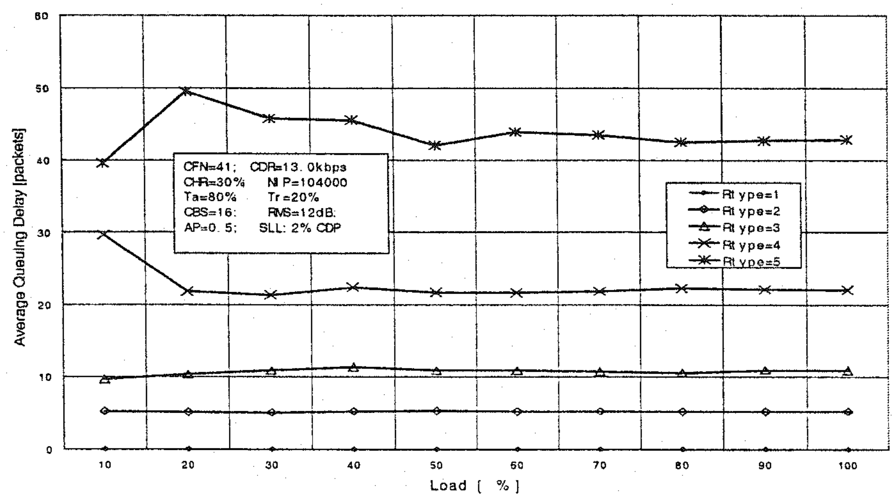


Figure 4.44 Average queuing delay in SRA vs. Load with Rate type (Rtype)

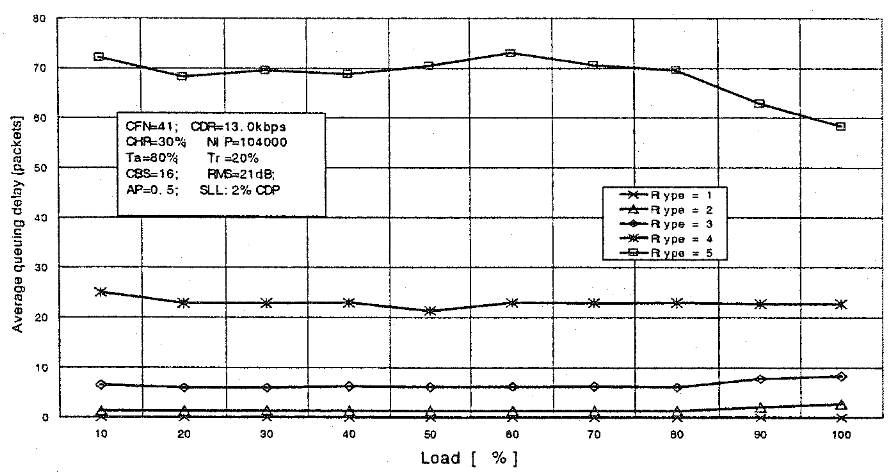


Figure 4.45 Average queuing delay in ARAOC vs. Load with Rate type (Rtype)

Figure4.44 to Figure4.47 show the comparison of the variance of the average queuing delay (VQD) & average queuing delay (AQD) vs. Load in both SRA and

ARAOC. The AQD & VQD for both the algorithms are unchanged by the sector load; but they increase as the traffic rates escalate. When the traffic class is equal to 5, the AQD & VQD of SRA is less than those of ARAOC. In the ARAOC, the AQD of traffic class 2 for ongoing voice call is about 2 packets (equal to 2*4.615ms = 9.23 ms), and the AQD of traffic class 4 for ongoing video call is about 22 packets or 101.52 ms. The values are acceptable as a real-time call.

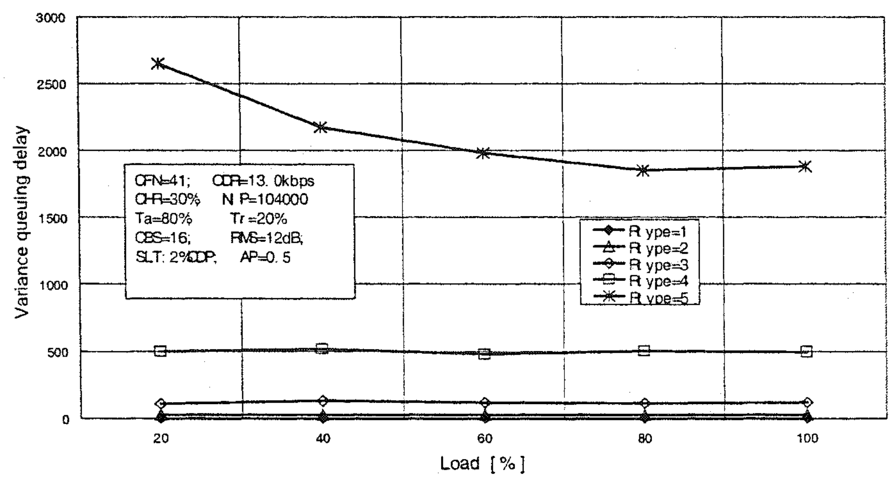


Figure 4.46 Variance queuing delay in SRA vs. Load with Rate type (Rtype)

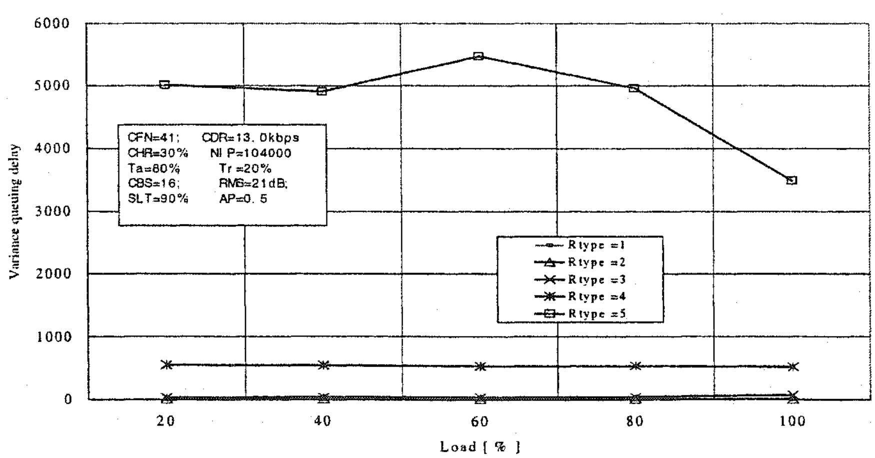


Figure 4.47 Variance queuing delay in ARAOC vs. Load with Rate type (Rtype)

However, the AQD & VQD of ARAOC in traffic classes of 2 & 3 are better than those of SRA. It satisfies the requirements of QoS for real-time services.

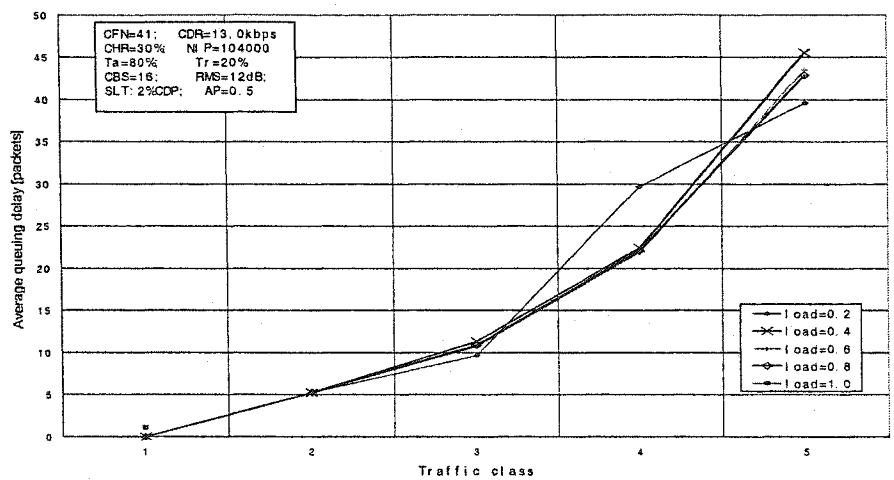


Figure 4.48 Average queuing delay in SRA vs. Traffic class with Load

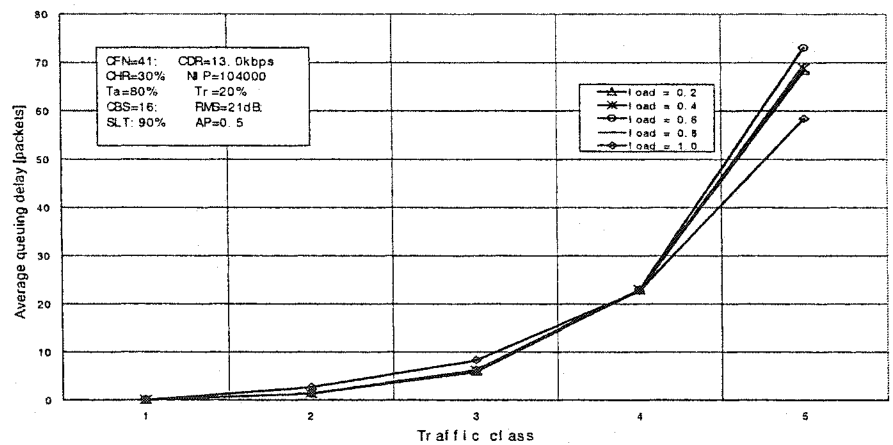


Figure 4.49 Average queuing delay in ARAOC vs. Traffic class with Load

Figure4.48 and Figure4.49 show the comparison of average queuing delay vs. Traffic classes in both SRA and ARAOC. Clearly, when the traffic class increases from 4 to 5, the AQD of SRA increases more slowly than that of ARAOC. However, the AQD of ARAOC in traffic classes of 1, 2 and 3 also increases much slowly than those of SRA. The AQDs in ARAOC for traffic classes of 2 and 4 are corresponding to 2 and 22 packets.

From the observation of Figure4.26 to Figure4.49, we analyze the influence of various traffic classes on the performance.

The CDP of ARAOC in traffic class 2 to 5 are far less than that of SRA, and less than 2% requirement at 95% load. Similar effects with CDP also apply to the CBP: the ARAOC can greatly lower both CDP and CBP for the traffic classes of 2-5. Thus it can improve the system capacity.

The NUE of SRA at 94% load is 50%, 65%, 68%, 76% and 80% for traffic class of 1, 4, 5, 3 and 2 respectively. Hence, the traffic class of 1, 4 and 5 will deteriorate system performance. Compared with the NUE of SRA, the NUE of ARAOC has a higher value. At a load of 94%, it can reach up to 47%, 80% and 84% for a traffic class of 1, 5, as well as 2, 3 and 4. Therefore, ARAOC can greatly improve the system performance.

The ABO of SRA has bigger values of 12% and 15% corresponding to the traffic classes of 2 - 4 and 5. The larger ABO results in more data error rate for ongoing calls, and a larger retransmission rate. In contrast, when the sector load increases up to 80%, the ABO of ARAOC is below 1.5% for class 1 to 4, as well as 4% for class 5. Otherwise, it is below 4%, 5%, and 7.5% at 94% load for traffic class 2 & 4, 3, 5, respectively. Therefore, ARAOC is much better than SRA in satisfying the requirements for packet loss and error rate.

When the traffic class increases from 4 to 5, the AQD of SRA increases more slowly than that of ARAOC. However, the AQD of ARAOC in traffic class of 1, 2 and 3 also increases more slowly than that of SRA.

Chapter 5
Summary and Future Work

5.1 Conclusion

The first objective of this thesis is to investigate the relationship between the network capacity and the quality of service in order to increase the utilization of radio resources. The second objective is to develop algorithms for system to manage the radio resource efficiently, and to control call admission, overload and congestion. The final objective is to investigate the influence of various parameters, such as various traffic rates; several channel buffer size, and different channel assignment parameters.

In order to accomplish the research objectives of this thesis, we have proposed dynamic co-channel interference algorithms to estimate Signal-to-Interference Ratio (SIR) for each channel based on cell cluster size of 1 and 1/3 and actualize SIR based on measurement at the mobile terminal. Furthermore, a dynamic channel allocation and frequency hopping schemes for every four multi-frames were developed to reduce co-channels interferences.

We have also proposed an adaptive resource assignment along with overload control (ARAOC) algorithm, and developed a new congestion control scheme. Finally, we have shown the results simulated by both SRA and ARAOC algorithms and investigated the performance of the system and the QoS of users.

In the proposed dynamic co-channel interference algorithms for the downlink, the cell cluster size and co-channel interferences were considered. When cell cluster size is 1/3, the fixed channel allocation algorithm is not available, but the dynamic channel allocation and frequency-hopping algorithm is available because it almost can avoid the worst case. When cell cluster size is 1, both the fixed and dynamic algorithms are available. Moreover, the dynamic algorithm can support higher modulation scheme and data rate due to better SIR.

In the proposed ARAOC algorithm, a new congestion control scheme has been developed to control user data rates, packet loss rate as well as real-time call delay. The consideration of algorithms can not only improve the capacity, but also decrease the call dropping rate, packet loss rate and the quality deterioration of the on-going calls possibly caused by admitting new calls.

In order to investigate system performance and the QoS of users, we apply both SRA and ARAOC algorithms to implement simulation results in GSM/GPRS systems.

By the simulation results of probability of call dropping and blocking in ARAOC algorithm, we can see that the call dropping probability achieves under 2% at full load, and call blocking probability is reduced to 4.5% for traffic classes 2-5. Both CDP and CBP in ARAOC are much better than those in SRA algorithm. Thus, ARAOC can greatly increase system capacity.

Simulation results of network utilization efficiency (NUE) show that NUE achieves above 80% on traffic classes 2-5 for user data transmission, in addition to 6.4% for signalling transmission in ARAOC algorithm. NUE in ARAOC is higher in value than that in SRA. Therefore, ARAOC can improve the system performance.

From the average buffer overflow (ABO) observation, we can find that ABO of ARAOC is 10 times lower than that of SRA when the load is less than 80%. Especially, ABO of ARAOC is under 4% at full load (94% of load) for real time traffic classes 2 and

4. The simulation results of average queuing delay (AQD) show that AQD of ARAOC in traffic classes 1,4 and 5 is not better than that in SRA, but AQD of ARAOC in class 2 & 3 is much better than that of SRA. Specially, the AQD of ARAOC in class 2 (real time voice) is about only 2 packets (2*4.615ms=9.23ms). Therefore, ARAOC can greatly improve the QoS of users and systems for real-time services and it can guarantee the required QoS for all traffic services.

By the above comparisons and discussions, we can conclude that the ARAOC produces higher system capacity, as well as better performances of users and the system. Especially, when the major traffic in the system is bursty data traffic, network performance should be much better. We use new schemes to control congestion. Our study indicates that adaptive assignment algorithm for various traffic services may significantly improve the performance of the network.

Most recent mobile systems (such as IS-136 and GSM) provide voice service with tolerable call dropping probabilities around 2%, speech rate at 13 kb/s, data rate up to 9.6 kb/s. To compare these integrated data services with GSM, we assume that their characteristics and environments are similar to those of GSM systems, such as channel data rate of 13.0kb/s, modulation of 0.3 GMSK and call dropping probability of 2% etc. Then we investigate the system capacity and performances introduced by various services in both SRA and ARAOC algorithms. Therefore, we can infer that the system, if uses different modulations (GMSK and 8-PSK) and channel data rates can support higher user rates and system capacity.

5.2 Future work

In the proposed dynamic co-channel interference algorithms along with frequency hopping and channel allocation, the modeling of adjacent co-channel interferences is based on a homogenous network with regular hexagonal cells. In reality, the cell geometry and the cell size in the practical network might not be the same for all cells. It would be interesting and useful to model the effect of channel allocation with different cell geometry and cell size.

Even though the co-channel interference for each channel has been estimated in both the worst and average cases for downlink, the measurement of its actual SIR has not been modeled. For a complicated metropolis radio network, it is necessary to model various environment including multi-path loss, fading etc.

In the proposed ARAOC algorithm, variability of transmission rate for each user has been considered, but both incremental redundancy (IR) and link adaptation (LA) between coding and modulation schemes have not yet been considered. ARAOC algorithm results in low system capacity and high dropping rate for a traffic class with low data rate. Hence, it is necessary to do link adaptation in order to improve the system performance.

In our simulation, only downlink performance is investigated. Since the environments in downlink may not be exactly the same as those in uplink, it is necessary to simulate both uplink and downlink in the future.

References

[1] S. Dixit, et al., "Resource Management and Quality of Service in Third-Generation Wireless Networks" IEEE Communications Magazine, p.125, Feb. 2001.

[2] O. Lataoui, et al., "A QoS Management Architecture for Packet Switched 3rd Generation Mobile Systems" Bell Labories, Lucent Technologies.

[3] ETSI, "Requirements for the UMTS Terrestrial Radio Access System," UMTS 04-01 UTRA, June 1997.

[4] M. Zeng, et al., "Harmonization of Global Third Generation Mobile Systems", Proceeding 20th Biennial Symposium Communications, p. 139-143, May 2000.

[5] J. Cai and D. Goodman, "General Packet Radio service in GSM," IEEE Commun. Mag., Vol. 35, no. 10, oct. 1997, p. 122-31.

[6] G. Brasche and B. Walke, "Concepts, Services, and protocols of the new GSM Phase 2+ General Packet radio service," IEEE Commun. Mag., vol. 35, no. 8, Aug. 1997, p.94--104.

[7] C. Bettstetter, H. Vogel, and J. Eberspacher, "GSM Phase 2+, General Packet radio Service, GPRS: Architecture, Protocols, and Air Interface," IEEE Commun. Surveys, 3rd qtr. 1999.

[8] "3rd Generation Partnership Project; Technical Specification Group Services and Systems Aspects; QoS Concept," 3G TR23.107 v3.1.1.

[9] 3GPP TS 23.107, "QoS Concept and architecture", version 3.0.0.

[10] J. Kim and A. Jamalipour, "Traffic Management and QoS Provisioning in Future Wireless IP Networks", IEEE Pers. Commun., p. 46, Oct. 2001.

[11] J. G. Sempere, "An overview of the GSM system".

[12] T. S. Rappaport, "Wireless Communications-Principles and Practice", New Jersey, NJ, Prentice Hall, 1996.

[13] GSM 05.02, Multiplexing and multiple access on the radio path.

[14] E. Dahlman et al., "WCDMA- The radio Interface for Future Mobile Multimedia Communications", IEEE Transactions on Vehicular Technology, vol. 47 no.4, November 1998.

[15] C. Bettstetter, et al, "GSM Phase 2+ General Packet Radio Service GPRS: Architecture, Protocols, and Air Interface", IEEE Commun. Surveys, vol.2 no.3, third quarter 1999.

[16] J. Cai, et al., "General Packet Radio Service in GSM," IEEE Communications Magazine, p.125, Oct. 1997.

[17] D. Molkdar and W. Feathstone, "System Level Performance Evalution of EGPRS in GSM Macro-cellular Environments", 2000 IEEE, p. 2654, VCT2000.

[18] X. Qin, et al., "RLC/MAC Design Alternatives for Supporting Integrated Service over EGPRS", IEEE Personal Communications, p.26-27, April 2000.

[19] ETSI GSM 05.03 v. 7.1.0, "Digital Cellular Telecommunications System (Phase 2+); Channel Coding," 1998.

[20] M. Eriksson, A. Furuskar, et al., " The GSM/EDGE Radio Access Network-GERAN; System Overview and Performance Evoluation." In Vehicular Technology Conference (VTC), Tokyo, May 2000.

[21] N.Sollenberger et al. "The Evolution of 136 TDMA for Third-Generation Wireless Services." IEEE Personal Communications, Vol.6, 1999.

[22] S. Eriksson et al., " Comparison of Link Quality control Strategies for Packet Data Services in EDGE", in Proceedings of IEEE VTC'99.

[23] A. Furuskar et al., "EDGE: Enhanced Data Rates for GSM and TDMA/136 Evolution", IEEE Personal Communication, June 1999, Vol.6, No.3.

[24] GSM 03.02, Network architecture.

[25] GSM 03.60, GPRS, service description, stage 2.

[26] GSM 09.60, GPRS, GPRS Tunneling Protocol (GTP) across the Gn and Gp interface.

[27] GSM 04.65, GPRS, MS-SGSN, Subnetwork Dependent Covergence Protocol (SNDCP).

[28] GSM 04.64, GPRS, MS-SGSN, Logical Link Control (LLC) layer.

[29] GSM 04.60, GPRS, MS-BSS interface, Radio Link Control/Medium Access Control (RLC/MAC) protocol.

[30] GSM 02.60, GPRS, Service description, Stage1.

[31] G. Brasche and B. Walke, Concepts, Services, and Protocols of the New GSM Phase 2+ General Packet Radio Service, IEEE Communications, vol.35, no. 8, p. 94-104, August 1997.

[32] J. Eberspacher and H. J. Vogel, GSM: Switching, Services and Protocols, John Wiley & Sons, 1998.

[33] GSM 03.64, GPRS, Overall description of the GPRS radio interface, Stage 2.

[34] GSM 05.02, Multiplexing and multiple access on the radio path.

[35] GSM 05.03, Channel coding.

[36] R.Johannesson and K. Sh. Ziganggirov, Fundamentals of Convolutional Codes, IEEE press, 1999.

[37] GSM 09.61, GPRS, interworking between the PLMN supporting GPRS and Packet Data Networks (PDN).

[38] M. Rahnema, "Overview of the GSM System and Protocol Architecture," IEEE Communications, vol. 31, no. 4, p. 92-100, April 1993.

[39] S. Faccin, Liangchi Hsu, R. Koodli, Khiem Le, and R. Purnadi, GPRS and IS-136 Integration for Flexible Network and Services Evolution, *IEEE Personal communication*s, vol. 6, no. 3, p. 48-54, June 1999.

[40] G. Heine, GSM Networks: Protocols, Terminology, and Implementation, Artech House, 1999.

[41] H. Arslan, S. C. Gupta, G. E. Bottomley, and S. Chennakeshu, "Successive cancellation of adjacent channel signals in FDMA/TDMA digital mobile radio systems," in *Proc. IEEE Veh. Technol. Conf.*, Ottawa, ON, Canada, May 1998, p. 1720–1724.

[42] B. C. Ng, M. Cedervall, and A. Paulraj, "A structured channel estimator for maximum likelihood sequence detection," *IEEE Commun. Letter*, vol. 1, p. 52–55, Mar. 1997.

[43] T. S. Pappaport, "Wireless communications principles & practice", p.38.

[44] T. S. Pappaport, "Wireless communications principles & practice", p.104.