Simulation

of

GSM/EGPRS Employing SDMA and Dynamic Channel Allocation

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A Thesis

in

The Department of Electrical and Computer Engineering

Presented in Partial Fulfillment of the Requirements for the Degree of

Master Of Applied Science at

Concordia University

Montreal, Quebec, Canada

March, 2003

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Abstract

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Lixia Sun

The development of GSM, the most successful second generation mobile communication standard, is currently focused on packet-switched services. The first solution to provide such services is known as General Packet Radio Service (GPRS), which has been deployed commercially. A further step comes with Enhanced General Packet Radio Service (EGPRS), which is able to provide third generation mobile data services with data rates up to 384 kbps for wide area coverage.

To improve the capacity and achieve more efficient utilization of the frequency spectrum of cellular systems, the Space Division Multiple Access (SDMA) using smart antenna has drawn wide attention in recent years. However, its application, especially on GSM/EGPRS network, still needs further investigation.

This work aims to evaluate the performance of the GSM/EGPRS cellular network employing SDMA, together with a complete Dynamic Channel Allocation (DCA) strategy, from various perspectives.

Simulation results show that SDMA can greatly increase the system capacity, and that the data rate, the distribution of users, the propagation environments, the channel assignment schemes, the antenna sidelobe levels and the buffersize of users all have great influence on the performance of the EGPRS network with SDMA.

Acknowledgments

First and foremost, my sincere appreciation goes to my academic advisor, Dr. Ahmed, K. Elhakeem, who has offered me invaluable guidance and advice throughout my work. Without him, my thesis would not have been accomplished.

I wish to express my gratitude to all professors who have ever taught me at Concordia University. Their enlightening lectures that allowed me to broaden my view and deepen my professional expertise are sincerely acknowledged.

I could never thank enough Mr. Steve Kremidas and his family, who helped me to settle down in this new country and support me whenever needed. Their kindness is deeply appreciated.

I would like to thank my fellow students who offered me their hands to solve all kinds of problems I encountered during the progress of my work. I will not forget those friends who were always on my side when I felt down.

My special thanks are also extended to my friends in China. I believe that their encouragement and trust, which have been accompanying me all these years, are driving me to success.

I thank my sisters, my brothers-in-law, my brother, and my lovely niece for their understanding and unfailing support.

Above all, I thank my dearest parents for their love and affection. It was my love for them that gave me strengths to start my program and enabled me to finish my thesis finally.

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Glossary of Acronyms

AMPS: Advanced mobile phone services.

8-PSK: 8-Phase Shift Keying

CDMA: Code division multiple access.

DCA: Dynamic Channel Allocation

ECSD: Enhanced Circuit-Switched Data

EDGE: Enhanced Data for Global Evolution

EGPRS: Enhanced GPRS

ETSI: European Telecommunications Standards Institute

FCA: Fixed Channel Allocation

FDMA: Frequency division multiple access

GERAN: GSM/EDGE Radio Access Network

GGSN: Gateway GPRS Support Node

GMSK: Gaussian Minimum Shift Keying

GPRS: General Packet Radio Services

GSM: Global System for Mobile communications

HSCSD: High Speed Circuit Switched Data

LLC: Logical Link Control

IMT-2000: Standard for 3G

ITU: International Telecommunications Union

LA: Link Adaptation

MCS: Modulation Coding Scheme

MSC: Mobile Switching Center

PCU: Packet Control Unit

PLMN: Public Land Mobile Network

QoS: Quality of Service

RLC/MAC: Radio Link Control/Multiple Access Control

SDMA: Space Division Multiple Access

SGSN: Serving GPRS Support Node

SINR: Signal-to-Interference Ratio

TDMA: Time Division Multiple Access

WCDMA: Wideband Code Division Multiple Access

Chapter 1

Introduction

1.1 Review

Mobile communications started from the pioneering work at AT&T Bell Laboratories, where the cellular concept was developed during the 1960's and 1970's.

The first-generation analog cellular systems became popular with a standard known as AMPS using Frequency Division Multiple Access (FDMA). Similar standards including TACS, NMT 450, and NMT 900 in Europe; ETACS in the United Kingdom; C-450 in Germany; and NTT, JTACS, and NTACS in Japan were developed around the world. The first generation cellular systems are generally incompatible with one another because of the different frequencies and communication protocols used [1][2].

The second generation systems use digital transmission, and employ Time Division Multiple Access (TDMA) or Code Division Multiple Access (CDMA) as a multiple access scheme. These systems include the GSM and DCS 1800 systems, the North American IS-54 system and IS-95 system, and the Japanese PDC system. The GSM, DCS1800, IS-54, and PDC systems use TDMA, whereas IS-95 uses CDMA. GSM is the most successful second generation standard. It not only became a Pan-European standard, but gained worldwide acceptance as the first universal digital cellular system with modern network features [1][2].

Standardization of third-generation (3G) mobile communication systems is now rapidly progressing in all regions of the world. The work is based on recommendations for International Mobile Telecommunications-2000 (IMT-2000), which have been developed by the International Telecommunication Union (ITU) since the late 1980s.

IMT-2000 systems will enhance the services provided by second generation systems with high data rates and multimedia capabilities. The 806–960 MHz, 1710–1885 MHz, and 2500–2690 MHz bands have been identified to provide advanced 3G services globally. The bit rate requirement is 384kbps with wide area coverage, and in the order of 2 Mbps in local areas [3]. Various candidate 3G systems, with CDMA2000, W-CDMA as representatives, are still under investigation,

The great success of GSM paves its way for its evolution toward 3G services. The Enhanced data rates for GSM Evolution (EDGE), which include Enhanced General Packet Radio Service(EGPRS) and Enhanced Circuit-switched Data (ECSD) is the evolutionary step after General Packet Radio Service (GPRS) that offers 3G data services in the existing GSM spectrum, and it has been accepted as a member of IMT-2000 family. The GSM/EDGE radio access network (GERAN) will be able to offer the same services as WCDMA by connecting to the same core network.

As we have already mentioned, cellular systems use different multiple access techniques, such as FDMA, TDMA and CDMA.

In FDMA system, the entire spectrum allocated to the service area is divided into channels of appropriate bandwidth, and users transmit on different frequencies.

In TDMA, used in digital system, such as IS-136 and GSM, the entire spectrum is divided into channels, as in FDMA systems. However, several users can use the same frequency channel. Each user is assigned a time-slot of a frame and transmits with its time-slot at a given time.

CDMA systems, such as IS-95, are based on spread spectrum transmission. All users within a cell share the same frequency band at the same time, and the users are separated by different spreading codes assigned.

In TDMA and FDMA systems, the entire set of channels available in the system is divided into N subsets of channels, the adjacent cells of the service area are grouped into clusters of a number N of cells, each subset is assigned to a cell in the cluster. This pattern is replicated over all clusters in the coverage area. The frequency reuse factor of a cellular system is given by 1/N, since each cell within a cluster is only assigned 1/N of the total available channels.

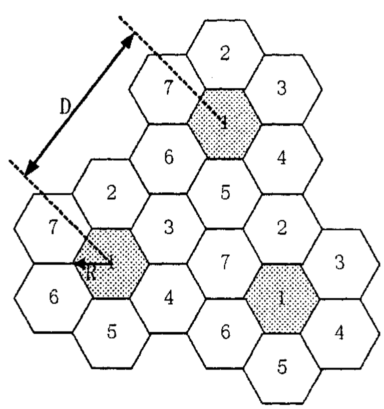


Figure 1.1 Illustration of Frequency Reuse with Cluster Size N=7

This implies that in a given coverage area, there are several cells, so called co-channel cells, that use the same set of frequencies. The interference resulting from the same channel used in co-channel cells is called co-channel interference. Co-channel users must be physically separated enough to reduce the co-channel interference in order to meet required performance. When the size of each cell is about the same, and the base stations transmit the same power, the desired signal to co-channel interference ratio (SINR) is a function of the radius of the cell (R) and the shortest distance between the centers of two cells sharing a given channel, so called reuse distance (D), as shown in Figure 1.1. It is clear that the smaller the cell cluster size N, the larger the number of channels available in the cell, thus the higher the system capacity. By system capacity we mean the amount of traffic a system can handle through efficient utilization of the frequency spectrum. However, reduction in cell cluster size increases co-channel interference, since the reuse distance D decreases.

Therefore, different channel allocation techniques are utilized in order to decrease cell cluster size, while controlling the co-channel interference. Among these techniques we have Fixed Channel Allocation (FCA) and Dynamic Channel Allocation (DCA) or a combination of the two.

In FCA techniques, a base station can only assign channels from its subset of channels to calls, guaranteeing the minimum tolerable SINR.

A DCA technique, with which the channel sets are not fixedly allocated to a cell, can be more adaptive to the variation of the traffic.

In recent years, the adaptive antenna array (smart antenna) has proven to be effective in reducing co-channel interference and, therefore, gained its popularity in cellular communication systems. An area that has drawn considerable attention from the research community is the exploitation of the spatial domain through antenna beamforming. Adding smart antennas at the base station enables Space Division Multiple Access (SDMA), a new multiple access scheme, which makes use of spatial separation of mobile users and can be used in combination with the above mentioned multiple access schemes. As a result, the system capacity can be greatly increased.

1.2 Motivation and Scope of the Thesis

As we have already known, EDGE is the next step in the evolution of GSM. EDGE can be introduced in two ways: (1) as a packet-switched enhancement for GPRS, known as enhanced GPRS or EGPRS, and (2) as a circuit-switched data enhancement called Enhanced Circuit-switched Data (ECSD). We will discuss them in more detail in Chapter 2.

The advantage of EDGE is its easier introduction based upon the current GSM infrastructure. Given that, whether EDGE can effectively compete with other third-generation technologies such as WCDMA and cdma2000 largely depends on its system capacity, and quality of services (Qos) it can offer. Since we are interested more in packet services, we choose to work on EGPRS.

SDMA has proven to be an effective technique in increasing the system capacity, but comprehensive research on EGPRS system performance based on SDMA still has not been done yet. By doing this work, we wish to provide the industry with some quantitative evaluation of EGPRS networks with the benefits of SDMA.

The main purposes of our work are to investigate:

1) The performance of a multi-cell EGPRS network using SDMA together with a complete DCA algorithm, with consideration of user mobility;

2) The influence of various parameters, such as mobile user data rate, buffer size, propagation environment, etc., as well as different channel assignment schemes-- static or adaptive, on the performance of the EGPRS network with SDMA;

3) The capacity improvement offered by EGPRS compared with conventional GSM in an SDMA system;

4) The influence of various parameters, such as spatial distribution of users, propagation environment, SINR requirement, antenna parameters, on the capacity of an SDMA system.

The simulations are programmed with C++ Builder, together with Microsoft Access database.

1.3 Thesis Outline

The rest of this thesis is organized as follows. In Chapter 2, we introduce GSM and its evolution. The services provided by GSM, its architecture, and its frame structure are briefly discussed. The new services—High Speed Circuit Switched Data (HSCSD), GPRS, EGPRS and ECSD evolved from GSM are presented, with GPRS and EGPRS in more detail. EGPRS is considered an add-on on GPRS.

A brief introduction of the adaptive antenna and its beamforming techniques is considered necessary, therefore, presented in Chapter 3. The benefits of adaptive antennas in mobile communications are discussed, followed by an introduction to beamforming techniques. In addition, the capacity improvement in GSM with application of adaptive antennas, proved by previous work, is briefly described.

Chapter 4 presents the SDMA system and channel allocation schemes. Since the operation of SDMA is based on spatial filtering, spatial properties of wireless communications channels are important factors in designing and analyzing SDMA systems. Macrocell and microcell spatial channel models, which are popularly employed, are described. Channel allocation constraints particularly for SDMA are discussed and several DCA schemes applied in SDMA systems are introduced.

Models employed in the simulations can greatly influence the simulation result. In Chapter 5, we explain models used in our simulations.

In Chapter 6, we concentrate on the description of our simulation, as well as on the analysis of the simulation results.

Finally, we summarize the contribution and conclusion of our thesis, and outline the future areas to be investigated in Chapter 7.

1.4 Contribution

The contribution of our work is the extensive performance evaluation of the EGPRS system with SDMA and DCA, which has not existed in the current literature. The calculation method of SINR of users in a 'wrapped around' cellular network we introduced would be helpful for researchers to make system design and analysis of the cellular networks. Our work can also greatly facilitate some future research in this area, such as the comparison of some different measurement-based dynamic channel allocation algorithms, channel allocation with priority, and capacity comparison of EDGE and CDMA.

Chapter 2

GSM Network and Its Evolution

2.1 GSM and Its Evolution

The GSM (Global System for Mobile Communications) is the most successful second generation digital cellular communication system. It was originally developed in order to create a common European mobile telephone standard and was first deployed in 1990, but finally became a worldwide mobile communication platform.

Today, GSM technology is in use by more than one in ten of the world's population and growth continues to soar with the number of subscribers worldwide expected to surpass one billion by the end of 2003.

The GSM was primarily designed to support voice traffic with very less emphasis on data services. As a result, the growth demand for data service has been hampered by low bit rate service, which is limited to only 9.6 kbit/s. The ubiquitous deployment of GSM around the world has led to some innovative ideas to provide high bit rate data services as quickly as possible by maintaining, if possible, the existing GSM infrastructure or part of it. Owing to this need, the European Telecommunications Standards Institute (ETSI) has focused its attention to increase the user data rate on the GSM radio interface. The following new data services are first deployed:

- General Packet Radio Service (GPRS)
- High Speed Circuit Switched Data (HSCSD)

And then, Enhanced Data for Global Evolution (EDGE) was introduced, which includes:

- Enhanced General Packet Radio Service (EGPRS)
- Enhanced Circuit Switched Data (ECSD)

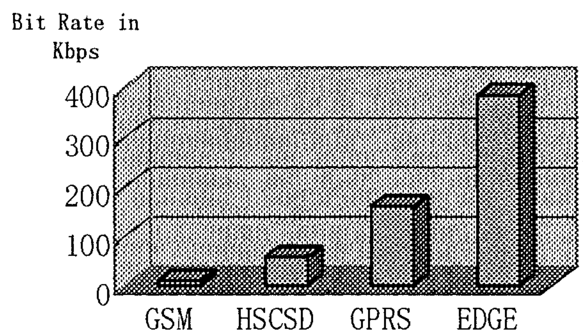


Figure 2.1 GSM User Bit Rate Evolution

Figure 2.1 demonstrates the data rate offered by GSM and its new services.

The standardization of GPRS and HSCSD has already been accomplished, whereas the standardization of EDGE is also almost complete. These data services will be elaborated in more detail.

2.2 A Brief Overview of GSM

GSM is a very complicated standard. A comprehensive overview is given in [4].

GSM uses two bands of 25 MHz for reverse link (from mobile to base station) and forward link transmission (from base station to mobile). It uses TDMA (actually a combination of FDMA and TDMA) scheme to provide base stations with simultaneous access to multiple users. The total number of available radio channels within 25 MHz is 125, spaced one from each other by a 200 kHz frequency band. In practical applications, a guard band of 100kHz is provided at the upper and lower end of the GSM spectrum, and only 124 radio channels are implemented. Each carrier frequency is then divided into 8 timeslots using a TDMA scheme. A TDMA frame is formed with 8 timeslots and lasts 4.615 ms.

A detailed GSM air interface specification is as shown in the Table2.1 [1]:

Table 2.1 GSM Air Interface Specifications [1]

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

A GSM physical channel corresponds to the recurrence of one timeslot every frame. On top of the physical channels, a series of logical channels are defined to perform a variety of functions. According to their functions, they can be divided into two categories: traffic channels and control channels.

- The traffic channels (TCH)

TCH is used to transport speech and data information.

Full-rate traffic channels (TCH/F) use a 26 TDMA frame multiframe structure. Among these 26 frames, 24 frames are reserved for traffic, 1 frame is used for control channel and the last frame is idle, which allows the mobile station to perform other functions, such as measuring the signal strength of neighboring cells.

As shown in the Figure 2.2, each user transmits a burst of data during the timeslot assigned to it. Normal burst is one of the five different data bursts defined in GSM for various traffic and control channels. It is used to carry speech or data information

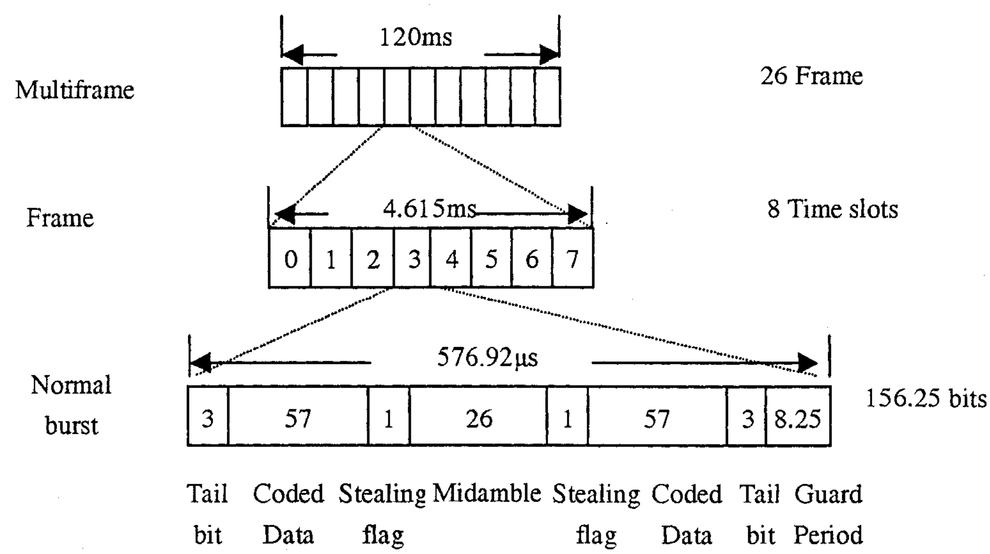


Figure 2.2 GSM Frame Structure

The traffic channels for the downlink and uplink are separated by 3 bursts so that mobiles do not need to transmit and receive at the same time. This simplifies considerably the electronics design of the system.

Half-rate traffic channels (TCH/H) double the capacity of the system. Though 26 frame multiframe structure is used, the internal structure is different.

- The control channels

The control channels are used for network management messages and some channel maintenance tasks. According to their functions, four different classes of control channels are defined: 1) broadcast channels used by the base station to provide the mobile station with the sufficient information it needs to synchronize with the network; 2) Common control channels helping to establish the calls from the mobile station or the network; 3) dedicated control channels used for message exchange between several mobiles or a mobile and the network.

GSM mainly provide telephone services, but also provide up to 9.6kbps data services, and supplementary ISDN services such as Short Messaging Services (SMS).

2.2.1 Architecture of GSM

The architecture of the GSM network, as shown in the Figure 2.3, includes two major subsystems: the Base Station Subsystem (BSS), and the Network and Switching Subsystem (NSS). A GSM subscriber requires a terminal called Mobile Station (MS) to connect to the network using the radio interface (Um).

1) The Mobile Station

The Mobile station includes the terminal and the Subscriber Identity Module (SIM).There are different types of terminals distinguished principally by their power and application. The SIM is a smart card that contains some parameters of the user such as its International Mobile Subscriber Identity (IMSI) in order to identify the subscriber to the system.

2) The Network and Switching Subsystem (NSS)

The NSS is responsible for call control, service control and subscriber mobility management functions.

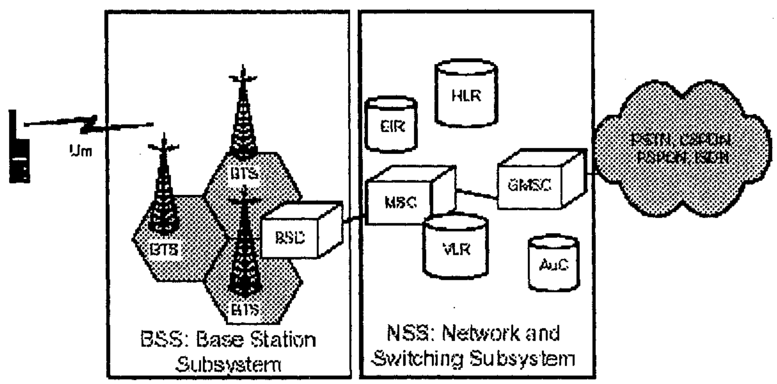


Figure 2.3 Architecture of GSM Network

MSC (Mobile Switching Center)

The MSC is mainly responsible for telephony switching functions of the network and performs authentication to verify the user's identity and to ensure the confidentiality of the calls.

HLR (Home Location Register)

The HLR is a database used to store and manage permanent data of subscribers such as service profiles, location information, and activity status.

The Authentication Center (AuC)

The AuC provides the necessary parameters to the MSC to perform the authentication. The AuC is generally integrated within the HLR.

The Equipment Identity Register (EIR)

The EIR is a database that contains information about the identity of the mobile equipment. It prevents calls from unauthorized MSs.

VLR (Visitor Location Register)

The MSC uses VLR, a database used to store temporary information about the subscribers, to serve visiting subscribers. When a subscriber roams to a new MSC area, a copy of all the necessary information is downloaded from the HLR into the VLR, so that calls of this subscriber can be processed without having to interrogate the HLR, which can be in another PLMN (Public Land Mobile Network). The temporary information in VLR is cleared when the mobile station roams out of this service area.

GMSC (Gateway Mobile Switching Center)

A GMSC is an MSC that serves as a gateway node to external networks, such as ISDN networks.

3) The Base Station Subsystem (BSS)

The BSS performs radio-related functions. It has BTSs (Base Transceiver Stations) and BSCs (Base Station Controllers).

BTS (Base Transceiver Station)

The BTS consists of radio equipment including transceivers and antennas and handles the radio interface to the MS.

BSC (Base Station Controller)

The BSC provides the control functions and physical links between the MSC and the BTS.A number of BSCs are served by one MSC, while several BTSs can be controlled by one BSC.

2.3 High Speed Circuit Switched Data (HSCSD)

The HSCSD was the first upgrade to be standardized by ETSI to bring high- speed data to GSM. The standardization started as early as 1994, therefore, HSCSD was the first high speed data extension to be ready for implementation in 1999.

The HSCSD provides data services by allocating multiple time slots throughout the call using circuit switching technique and making use of the existing GSM network. The main aim of HSCSD is to offer some real-time data services such as video, which require higher bandwidth but low transmission errors.

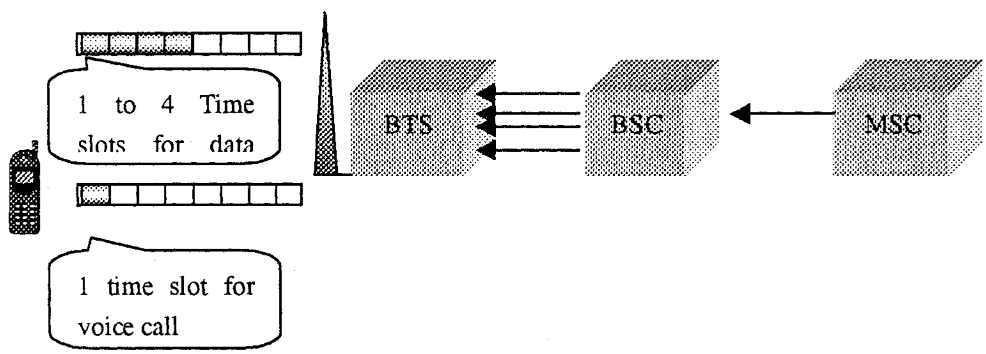


Figure 2.4 Illustration of HSCSD Operation

As mentioned previously, conventional GSM network supports maximum useful data 9.6kb/s user data rate with single time slot in both core network and BSS part. By changing puncturing scheme in the transmission path of data, the user data rate in core network is changed from 9.6kb/s to 14.4kb/s.

So concerning the core network, the actual improvement is 4.8kb/s in user data rate. In BSS part, the HSCSD implementation reserves multiple timeslots for one high-speed data call. The number of reserved time slots varies from one to maximum four. The maximum data rate can achieve in HSCSD is 57.6kbps.

The operation of HSCSD is illustrated in Figure 2.4.

HSCSD is a circuit-switched technology, therefore, it is a rather simple upgrade of conventional GSM in comparison with GPRS or EDGE.

2.4 General Packet Radio Service (GPRS)

GPRS is the first step in the data evolution of GSM platform. GPRS enabled networks to offer 'always-on', higher capacity, Internet-based content and packet-based data services. This enables services such as color Internet browsing, e-mail on the move, powerful visual communications and multimedia messages.

GPRS cooperates with GSM by upgrading GSM services provided from the mobile to data networks. Data packets are transmitted directly from the mobile, and technologies are employed to enable radio resources at the air interface to be used only when there is data to transmit or receive.

GPRS provides end-to-end packet services by allocating multiple time slots only when the data has to be sent providing efficient use of the radio resource. In this way, more than one user can share the same time slot(s). In the event when input traffic is very high, the users would experience higher delay due to lower throughput. In practice, GPRS is feasible for elastic traffic, which can tolerate delay, e.g., Web browsing, email and file transfer. In order to improve channel throughput by implementing a suitable link adaptation technique, four different coding schemes have been standardized. With all these features the data rate can be extended up to115 kb/s. Though GPRS and GSM use the same radio interface and physical layer, the former requires new packet oriented core network entities for operation. The packet switched core network in GPRS will allow mobile users to access packet data networks such as IP and X.25.An elaborate description of GPRS can be found in [5].

2.4.1 Architecture of GPRS

GPRS introduces the following two new major network elements: SGSN (Serving GPRS Support Node) and GGSN (Gateway GPRS Support Node). The architecture of GPRS network is as illustrated in Figure 2.5.

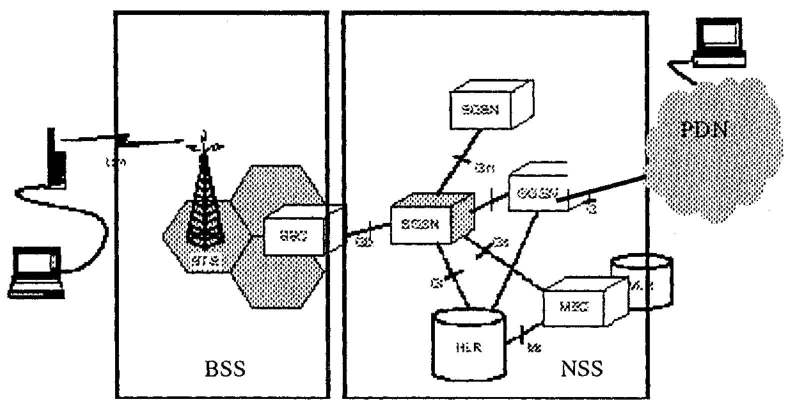


Figure 2.5 Architecture of GRPS Network

1) GPRS BSS

Because GPRS uses the same frequency bands, the same radio modulation and burst structure as GSM, GPRS has minor impact on the existing GSM BSS. This makes it easy to reuse existing component and links without major modifications.

A new component, called PCU (Packet Control Unit) was added to the BSS in the GPRS standard to support the handling of data packets. The PCU (not shown in Figure 2.5) is placed logically between the BSS and the GPRS NSS. Unlike the circuit-based connections, connections in GPRS have to be established between the BSS and the MS only when data needs to be transported over the air interface.

2) GPRS NSS

The GPRS NSS can be viewed as an overlay network over GSM providing packet-switched data transmission. GPRS introduces GPRS Support Node (GSN) to the GSM infrastructure (Figure 2.5). GSN can be either SGSN (Serving-GSN) or GGSN (Gateway-GSN). The SGSN sends data to and receives data from mobile stations, and maintains information about the location of mobile stations. GGSN is a gateway that allows mobile cell phone users to access the public data network (PDN) or specified private IP networks. The SGSN communicates between the MS and the GGSN.

The HLR database is updated to contain GPRS subscriber information. Adaptations to an existing MSC/VLR are not required but the GPRS standard suggests some enhancements to coordinate between the SGSN and the MSC/VLR if the optional interface between the two is to be supported.

Several interfaces have been introduced in GPRS to define entity-to-entity interactions as illustrated in Figure 2.5.

2.4.2 Air Interface

2.4.2.1 Radio Resource Management

The radio resources of a cell are shared by all GPRS and non-GPRS mobile stations located in a cell. A physical channels used for GPRS traffic is called packet data channel (PDCH). The PDCHs are taken from the common pool of all channels available in the cell. The mapping of physical channels to either packet switched (GPRS) or circuit switched (conventional GSM) services can be performed dynamically, depending on the current traffic load, the priority of the service, and the multislot class.

2.4.2.2 Logical Channels in GPRS

1) Logical Channels

Similar to the case of GSM, a series of logical channels are also defined to perform a variety of functions and divided into traffic channels and control channels [5]. Table 2.2 lists the packet data logical channels defined in GPRS [6].

Table 2.2 Logical Channels in GPRS

Group	Channel	Function	Direction
Packet data traffic channel	PDTCH	Data traffic	MS ↔ BSS
Packet broadcast control channel	PBCCH	Broadcast control	MS ← BSS
Packet common control channel (PCCCH)	PRACH PAGCH PPCH PNCH	Random access Access grant Paging Notification	MS → BSS MS ← BSS MS ← BSS MS ← BSS
Packet dedicated control channels	PACCH PTCCH	Associated control Timing advance control	MS ↔ BSS MS ↔ BSS

The packet data traffic channel (PDTCH) is employed for the transfer of user data. One mobile station can use several PDTCHs simultaneously.

The packet broadcast control channel (PBCCH) is used by the BSS to broadcast specific information about the organization of the GPRS radio network, such as system information about circuit switched services, to all GPRS mobile stations of a cell.

The packet common control channel (PCCCH), which consists of four sub-channels, is used to transport signaling information for network access management:

- The packet random access channel (PRACH) is used by a mobile station to request one or more PDTCH.
- The packet access grant channel (PAGCH) is used to allocate one or more PDTCH to a mobile station.
- The packet paging channel (PPCH) is used by the BSS to find out the location of a mobile station (paging) prior to downlink packet transmission.
- The packet notification channel (PNCH) is used to inform a mobile station of incoming multicast or group call messages.

The dedicated control channel consists of two sub-channels:

- The packet associated control channel (PACCH) conveys signaling information, such as power control, resource assignment, and reassignment information.
- The packet timing advance control channel (PTCCH) is used for adaptive frame synchronization.

2) Utilization of the Logical Channels

We use uplink packet transfer to illustrate how the logical channels described are utilized. A mobile station requests radio resources for uplink transfer by sending a "packet channel request" on the PRACH or RACH. A slotted-ALOHA random access techniques in used to transmit requests. The network answers on the PAGCH or AGCH respectively. It tells the mobile station which PDCHs it may use. A so-called uplink state flag (USF) is transmitted in the downlink to tell the mobile station whether or not the uplink channel is free. An MS monitors the USF and starts transmission depending on its assigned USF value [7]. A more detailed utilization of USF will be described later on when we talk about EGPRS.

2.4.2.3 Multiframe Structure

A multiframe structure for PDCHs consisting of 52 TDMA frames is shown in Figure 2.6. The 52-multiframe has a duration of 240 milliseconds. One multiframe of 52 radio frames is divided into 12 radio blocks of four radio frames each, two idle frames and two frames for control channel. A radio block is a series of four bursts that are not consecutive in the same radio frame, but belong to four consecutive radio frames in the same carrier in the same timeslot. Hence, one radio block can be shared among eight users simultaneously. In addition, since a multiframe consists of 12 radio block, in the time of one multiframe, 12×8=96 users can share the same medium[8].

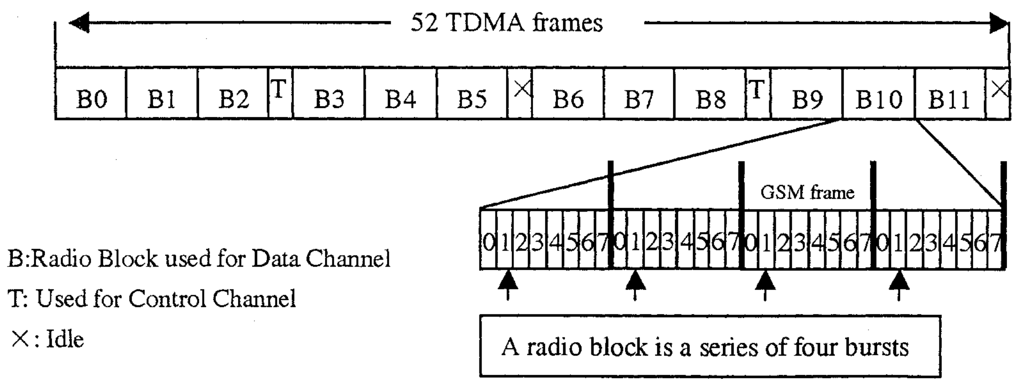


Figure 2.6 Frame Structure of GPRS

2.4.3 Transmission Protocols in GPRS

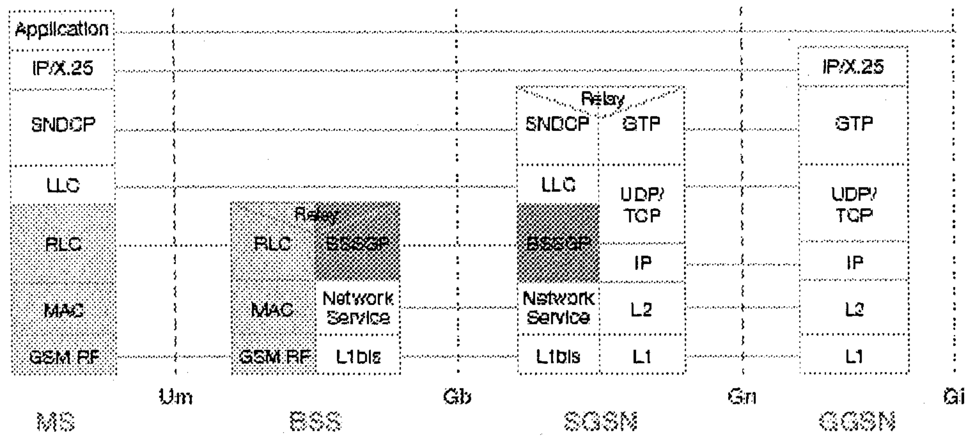


Figure 2.7 Transmission Protocols in GPRS[12]

(Shaded areas refer to protocols influenced by the introduction of EGPRS)

Like other data networks, the GPRS also has layered protocol architecture between different network entities. This layered protocol is responsible for data compression, resource sharing, flow control and error control. With the help of an efficient medium access control on the radio interface, multiple users can share the same channel(s). Similarly, the function of radio link protocol is to insure reliable data transfer over the radio interface. We will not discuss in detail the functions of each layers as this is not our main concern. The protocol stack is as shown by Figure 2.7.

2.4.4 Management in GRPS Network

2.4.4.1 Attachment and Detachment

In order to use GRPS network, a user must register with an SGSN. After the user's authorization is checked, the network copies the user's profile from the HLR to the SGSN, and assigns a packet temporary mobile subscriber identity (P-TMSI) to the user. This procedure is called GPRS attach. The disconnection from the GPRS network is called GPRS detach. It can be initiated by the mobile station or by the network (SGSN or HLR).

2.4.4.2 Session Management

To exchange data packets with external Packet Data Networks (PDN) after a successful GPRS attach, a mobile station must apply for one or more addresses used in the PDN. This address is called PDP address (Packet Data Protocol address). For each session, a so-called PDP context is created, which describes the characteristics of the session. This context is stored in the MS, the SGSN, and the GGSN. With an active PDP context, the mobile station 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.

2.4.4.3 Location Management

The network needs to keep track of the user's current location in order to route the incoming packet to him. Therefore, an MS needs to frequently send location update messages to its current SGSN. A state model shown in Figure 2.8 has been defined for location management in GPRS [5]. An MS can be in one of three states—READY, STANDBY, IDLE--depending on its current traffic amount; the location update frequency depends on the state of the MS.

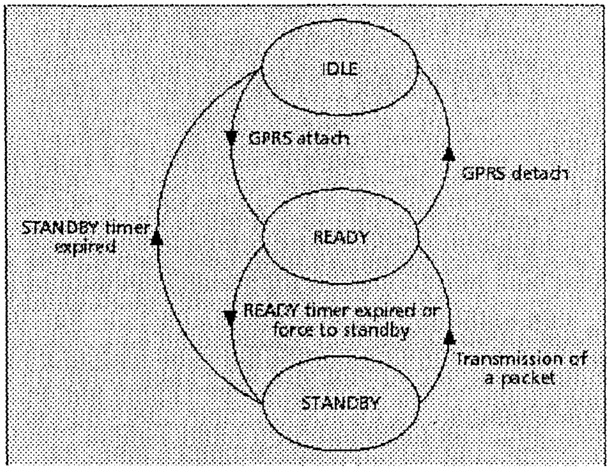


Figure 2.8 Location Management State Model

When in IDLE state, the MS is not reachable. After a GPRS attach, the MS turns into READY state. With a GPRS detach, it disconnects from the network and go back to IDLE state. If an MS does not send any packets for a longer period of time, and therefore the READY timer, which was started at GPRS attach, expires, this MS turns to the STANDBY state.

In IDLE state, no location updating is performed. In READY state, an MS informs its SGSN when it moves to a new cell. In STANDBY state, location information is updated only if the routing area (RA) is changed. This routing area is a subset of the GSM location area and consists of an operator-defined group of cells.

2.4.4.4 Interworking with IP Networks

A GPRS network, which supports both IPv4 and IPv6, can be interconnected with an IP-based packet data network, such as the Internet or intranets. From an external IP network's point of view, the GPRS network is just like an IP sub-network, and the GGSN serves as an IP router.

Each registered user who wants to exchange data packets with the IP network gets an IP address. The IP address is taken from the address space of the GPRS operator. The address resolution between IP address and GSM address is performed by the GGSN, using the appropriate PDP context.

A domain name server (DNS) managed by the GPRS operator or the external IP network operator can be used to map between external IP addresses and host names. A firewall is recommended to be installed between the private GPRS network and the external IP network to protect the PLMN from unauthorized access.

Therefore, a GPRS network can be seen as a wireless extension of the Internet all the way to a mobile station or mobile computer.

In summary, the GPRS technology provides the following benefits:

• Enables the use of a packet-based air interface over the existing circuit-switched GSM network. Because the radio bandwidth is used only when packets are sent or received, greater efficiency in the radio spectrum is achieved.

• Supports data rates of about 115 Kbps, which is much greater than the traditional 9.6 Kbps rate available in a circuit-switched connection.

• Supports virtual private network (VPN)/Internet service provider (ISP) corporate site access.

2.5 Enhanced Data for Global Evolution (EDGE)

In order to comply with the current and future data rate requirements, a large number of time slots would be needed to allocate to each data connection both in GPRS and

HSCSD. Besides that, some data rate requirements are so high that they cannot be offered even with multi-slot operation on the radio interface. Like in GSM, both HSCSD and GPRS utilize the same Gaussian Minimum Shift Keying (GMSK) modulation on the radio interface. As a result, achievable user data rate per time slot is not significantly increased.

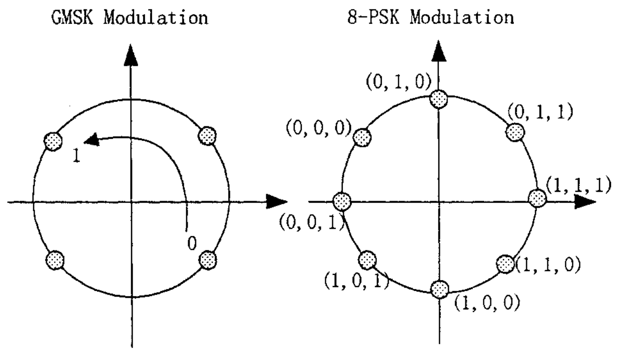


Figure 2.9 Benefit of Modulation

In order to further increase the radio interface data rate, EDGE was introduced. The aim of EDGE is to increase the gross bit rate on the GSM radio channels by modifying the existing physical layer[9]. Higher data rates are achieved by introducing Phase Shift Keying modulation with 8 states (8-PSK) as well as new channel coding schemes by exploiting the favorable radio interface conditions. Using 8-PSK, the symbol rate remains the same, but the total data is increased by a factor of three, since each symbol represents three bits instead of one, as shown in the Figure 2.9. This will lead to higher throughputs on the radio interface and consequently to lower transmission delays. The modification in physical layer significantly increases the spectral efficiency but will obviously require better propagation conditions or more complex receivers, since 8-PSK is more vulnerable than GMSK to inter-symbol interference. Data services expected to be supported by

EDGE include all previous services plus fast Internet access, video teleconferencing, high speed file transfer etc.

There are two different approaches by which the EDGE is expected to provide the data services to the mobile users. One method is based on packet switching technique, whereas the second approach relies on traditional circuit switching. Therefore, the two components of EDGE, as we already mentioned, are:

- EGPRS (Enhanced GPRS)
- ECSD (Enhanced Circuit Switched Data)

EDGE will provide two phases:

Phase 1: Single- and multislot packet-switched and circuit-switched services.

Phase 2: Real-time services.

2.5.1 Enhanced General Packet Radio Service

Enhanced General Packet Radio Service (EGPRS) is an enhanced version of GPRS. EGPRS is therefore an add-on to GPRS and cannot work alone. GPRS has a greater impact on the GSM system than EGPRS has. The higher rate in EGPRS has been achieved primarily by major modification of the GPRS specifications at the physical layer and RLC/MAC layer (protocols influenced by the introduction of EGPRS are as shown by shaded areas in Figure 2.7). As already mentioned, EGPRS used 8-PSK multi-level modulations. By using the new modulation and coding schemes and by making adjustments to the radio link protocols, EGPRS offers significantly higher throughput and capacity. An aggregate data rate per data call can be increased up to 384 kb/s by allocating 8 time slots. Theoretically, the data rate can reach 59.2 kbps per timeslot. Please be aware that the standard of EGPRS has not been finalized, the techniques in EGPRS we are talking about are subject to modification.

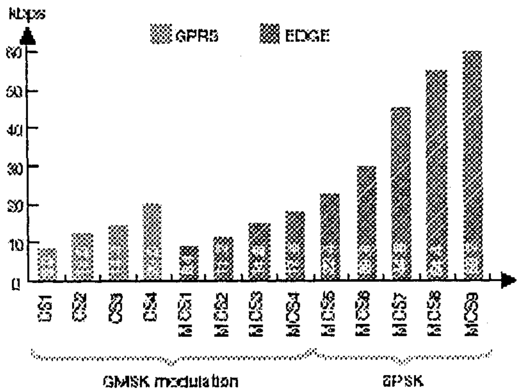
2.5.1.1 EGPRS Link-controlling Function

To prevent any throughput degradation due to the varying channel conditions, the EGPRS employs two types of link quality control techniques: link adaptation and incremental redundancy. Proposals for adaptive radio link protocol and performance simulation can be found in [10][11].

A link adaptation scheme regularly estimates the link quality and subsequently selects the most appropriate modulation and coding scheme for coming transmissions in order to maximize the user bit rate. In an incremental redundancy scheme, information is first sent with very little coding, yielding a high bit rate if decoding is immediately successful. If decoding fails, additional coded bits (redundancy) are sent, until decoding succeeds. The more coding that has to be sent, the less the resulting bit rate and the higher the delay.

EGPRS will support a combined link adaptation and incremental redundancy scheme. In this scheme, the initial code rate for the incremental redundancy scheme is based on measurements of the link quality.

To realize these adaptation techniques, nine different types of modulation and coding schemes have been defined in the standard. They are designated MCS1 to MCS9. The lower four EGPRS coding schemes use GMSK, whereas the upper five use 8PSK. The data rate vs. coding schemes is illustrated in Figure 2.10[12].



CS: Coding Scheme
MCS: Modulation Coding Scheme

Figure 2.10 User Data Rate Based on Different Coding Schemes

2.5.1.2 Multiplexing Capability

The EGRPS RLC/MAC layer is designed to efficiently support multiple data streams on the same packet data traffic channel(PDTCH), and to support a given data stream on multiple channels.

EGPRS uses an entity called a temporary block flow (TBF) for data transfer. A TBF is a virtual connection between an MS and the BSS, which is maintained for the duration of the data transfer and comprises a number of RLC blocks (each RLC block is coded and interleaved over 4 GSM frames, therefore, the logical resource assignment is one RLC block, and the transmission unit in the physical layer is four GSM time slots--one radio block). Each TBF is identified with 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. Radio blocks destined to different MSs are differentiated by their attached TFIs. After completion of the data transfer, the TFI 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 . Therefore, the BSS can communicate with a given MS on any of the channels assigned to it, and can also multiplex several TBFs destined to different MSs on the same physical channel.

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. Several mobiles may be assigned to the same uplink traffic channel but with different USFs. The USF is 3 bits long, which implies that up to eight different data transfers can be multiplexed on one channel.

By setting the USF flag of the corresponding downlink channel to an appropriate value, the base station uses a centralized in-band polling scheme to poll the desired MS. An MS listens to all the downlink traffic channels that are paired with the uplink channel assigned to it. If its USF appears in the downlink channel, the MS uses the corresponding uplink channel in the next logical frame. This is shown in Figure 2.11.

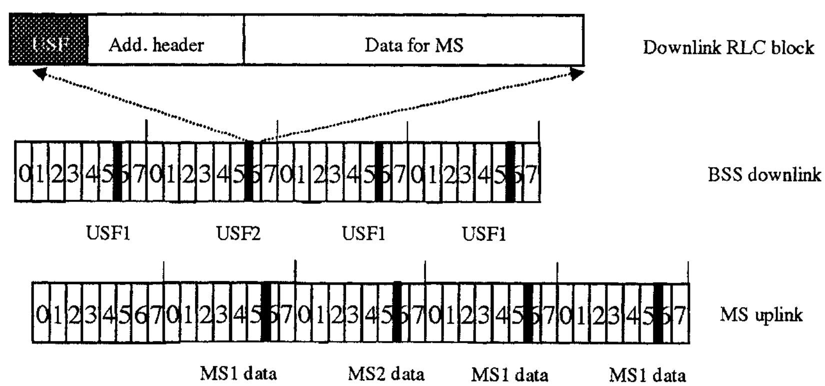


Figure 2.11 The principle of uplink multiplexing

2.5.1.3 Multiple Access Procedure

EGPRS allows two types of access procedure for data transfer, one-phase and two-phase access. In one-phase procedure, the uplink data transfer begins concurrently with the service negotiation and mobile verification, whereas in the two-phase procedure, the uplink data transfer begins only after the mobile verification and service negotiation are accomplished.

Because the MS is not identified in the initial access request and there is a lot of signaling exchange, the above-mentioned access procedures are unnecessarily slow especially for on-going session access. The following capabilities are desired:

a. Fast uplink access during an ongoing session.

b. Fast resource assignment for both uplink and downlink.

Therefore, the following common control channels are proposed[13].

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

The F-PACH is used exclusively for ongoing calls. The fast packet channel request 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.

b. Fast packet control channel (F-PCCH)

The F-PCCH serves two major functions: to transmit access grant and polling messages to specific mobiles. For this functionality, 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). The F-PAGCH is used to respond to access requests received on the

F-PACH. This response is typically an assignment message that specifies the channels, USFs, and other parameters for a set of MSs. The F-PPCH is used to poll different mobiles.

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 signaling overhead, which makes it feasible to meet the delay requirements. These channels could be 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.

An example of Overall Access Procedure:

▪ When a new call starts, the MS uses the normal PRACH to access the system; it establishes a TBF and obtains a TFI, USF(s), and PDTCH(s).

▪ At the end of each active period (e.g., no more data to send):

For the services not delay sensitive, such as 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-specific access procedure. The MS receives a 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.

2.5.2 Enhanced Circuit Switched Data (ECSD)

Enhanced Circuit Switched Data (ECSD) utilizes circuit-switching technique by allocating multiple full rate traffic channels like in the case of HSCSD. Due to multislot operation, ECSD is also suitable for non-bursty high-speed data applications such as video-conferencing. However, by virtue of physical layer modification, ESCD needs fewer resources on the radio interface compared with HSCSD in order to achieve similar throughputs. Several modulation and coding schemes have also been specified in ECSD to augment the user data rate. These new coding schemes would also allow the implementation of link adaptation technique to insure higher throughput even under bad channel conditions. ECSD is attractive in terms of both capacity and quality of service, as well as its easier implementation into the existing GSM network.

Chapter 3

Adaptive Antennas

3.1 Introduction

Wireless communications suffers from three major impairments, which limit their performance and capacity:

- Multipath fading: which results from the multipath that the signals take to reach the receiving antenna. These paths change with antenna location, direction, polarization and with time, resulting in fluctuation of the amplitude and phase of the received signals.

- Delay spread: the propagation delay varies among multiple paths. When the delay spread exceeds about 10 percent of the symbol duration, it can result in severe inter-symbol interference.

- Co-channel interference: the frequency reuse in cellular systems brings co-channel interference.

Smart antennas can effectively mitigate these problems and improve the wireless system performance in a broad range of ways.

✧ Reduction in Co-channel Interference

In the transmit mode, base station adaptive antennas can focus radiated energy toward the desired mobile, reducing the interference in other directions. As well, nulls can be steered towards co-channel mobiles, providing further reduction in co-channel interference. In the receive mode, a high antenna gain can be steered toward the desired signal, reducing the interference received from other co-channel mobiles.

✧ Improvement in Link Quality

Multipath in radio channels can result in fading or time dispersion. By forming narrow beams in certain directions and nulls in others, adaptive antennas can mitigate the impact of multipath, and reduce the delay spread. In the receive mode, when a beam is steered towards the desired signal, multipath components impinging upon the antenna from directions other than the direction of the desired signal are attenuated. In the transmit mode, narrow beam can be formed toward the desired direction, reducing multipath reflections that cause the delay spread.

Adaptive antenna can even exploit the diversity inherent in multipath, for example, the spatial filtering rake receiver used in CDMA system.

In addition, employing of smart antenna can increase the system capacity. The reduction in co-channel interference due to the use of adaptive antennas can be traded for more users in the cell. For example, in TDMA/FDMA systems, the reduction in co-channel interference may be sufficient to allow cluster size reduction, increasing the number of channels per cell. In CDMA systems, the use of adaptive antennas allows users to transmit less power, reducing the multiple access interference, which, in turn, increases the number of users in the cell.

As well, a base station adaptive antenna can be used to create additional channels in the cell, by spatial filtering. By steering narrow beams towards mobiles, in-cell mobiles can share the same channel, provided that they are sufficiently separated from each other.

This technique is so-called Space Division Multiple Access (SDMA).

A comprehensive introduction to smart antenna is presented in [14][15].

3.2 The Principle of Beamforming

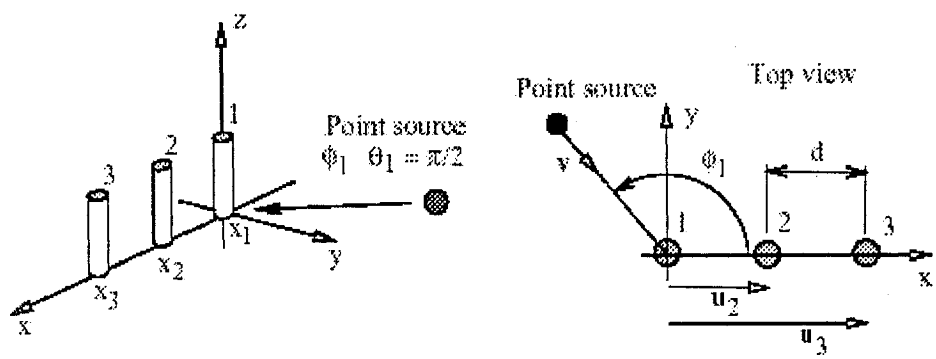


Figure 3.1 Linear Array

Consider a uniformly spaced linear array with three identical isotropic antenna elements as shown in Figure 3.1. The inter-element spacing d is usually chosen to be equal to half the wavelength of the received signal. A point source is located in the direction (ϕ, θ), and, for simplicity, we assume that $\theta = \pi/2$.

With the back ground noise ignored, and supposing that we have L resources, the complex envelop of the total received signal at the antenna element m is[15]:

$$x_m(t) = \sum_{l=1}^{L} m_l(t) e^{-j\Delta\psi_m(\theta_l, \phi_l)} \qquad (3.1)$$

where $m_l(t)$ is the complex envelop of signal transmitted by the point source l, and $\Delta\psi_m(\phi_l, \theta_l)$ is the phase shift which can be written in terms of the element location (x_m,y_m,z_m) and source direction (ϕ_l, θ_l) :

$$\Delta\psi_m(\phi_l,\theta_l) = \beta(x_m \cos\phi_l \sin\theta_l + y_m \sin\phi_l \sin\theta_l + z_m \cos\theta_l)$$

where $\beta = 2\pi f_0 / c$, f_0 is the signal frequency of source l.

Figure 3.2 shows the beamforming principle of an adaptive array, where signals received by multiple elements are weighted and combined to generate an output signal.

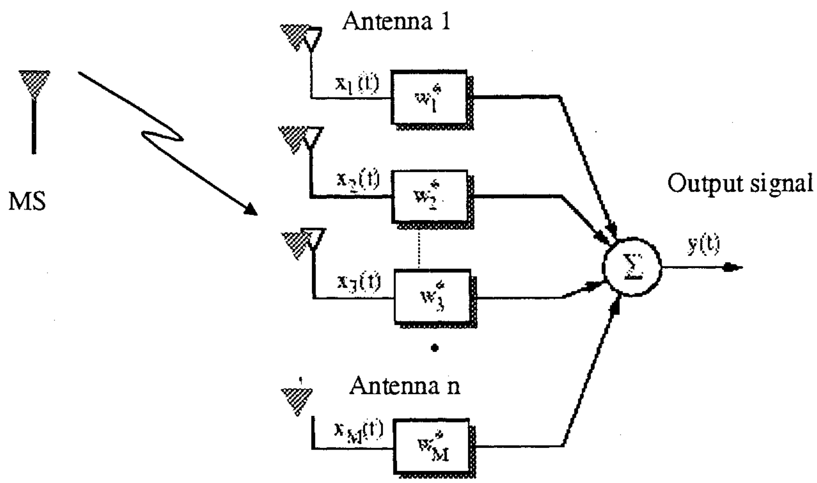


Figure 3.2 Principle of Antenna Array

It is convenient to use vector while working with array antennas. We define first the array weight vector as:

$$w=[\, w_1\ w_2\ w_3 \ldots w_M]^H \tag{3.2}$$

Where H represents the Hermitian transpose, a transposition combined with complex conjugation.

As shown in the Figure 3.2, the signals at output of antenna array, y(t), is related to the vector of signals at the input of the array, $x(t)=[x_1(t)\ x_2(t)\ldots x_M(t)]^T$ by:

$$y(t)=w^H x(t) \tag{3.3}$$

Consider now that there is only one point source, L=1, y(t) can also be written as:

$$y(t) = m_1(t)\sum_{m=1}^{M} w_m^* e^{-j\Delta\psi_m(\theta_1,\phi_1)} = m_{1(t)} f(\theta_1,\phi_1) \tag{3.4}$$

Where

$$f(\theta_1,\phi_1) = \sum_{m=1}^{M} w_m^* e^{-j\Delta\psi_m(\theta_l,\phi_l)} \tag{3.5}$$

is called array factor.

By adjusting w_m^*, we can form beams or nulls such that the signals received are maximized or minimized. The beam can be made to point in any direction by changing the weights. An example is as shown by Figure 3.3.

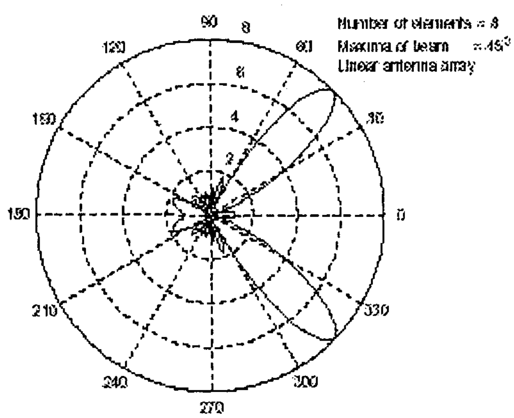


Figure 3.3 Array factor of a linear antenna array with maxima at 45^0 [16]

A beamforming network is usually used to produce M beams from M elements, Simple fixed beamforming can be implemented by using Butler Matrix[17]. The fixed beamforming matrix is bi-directional, that is, the same beam pattern used for transmit can also used for receive.

3.3 Switched Beam Systems

In smart antenna systems where beamforming is based on forming a set of fixed beams, steering towards predetermined directions, a switch is used to select a beam for transmission or reception based on a beam-selection algorithm. In the receive mode, the switched beam system selects the beam that provides the best reception of a particular signal. In the transmit mode, the beam selected is the one that better illuminates the region where the desired receiver is located [17].

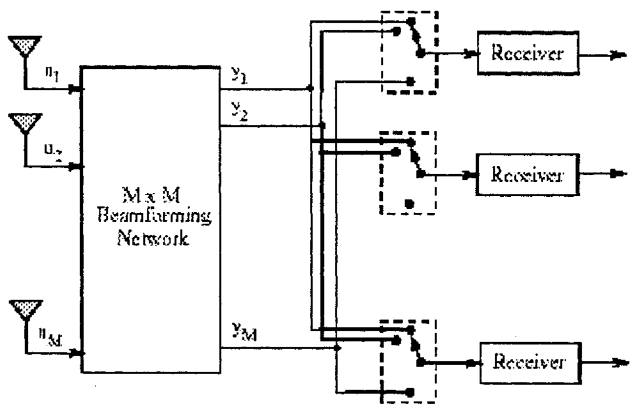


Figure 3.4 Switched Beam System

Figure 3.4 shows a switched beam system in the receive mode.

The mechanism for performing beam selection is highly dependent on whether we are considering an FDMA, TDMA, or CDMA system, though it is possible to consider switched beam approaches to each of these multiple access methods.

The advantage of switched beam systems is the low complexity compared to the complexity of fully adaptive antennas. Also, the integration of a switched beam system in an existing cellular system requires few modifications in the cellular systems.

There are some disadvantages, naturally, as payoff for simplicity. First, the system cannot provide protection from multi-path components which arrive with direction-of-arrival near that of the desired signal component. Systems based on fixed beamforming networks only are more sensitive to angular distribution of multipath components than systems based on adaptive beamforming networks. Second, switched beam systems cannot make use of path diversity by combining coherent multipath components. Finally, the user's signal strength fluctuates as user moves from the center of the beam to the edge of the coverage of a particular beam due to scalloping, which is the roll-off of the antenna pattern as a function of angle as the direction of arrival varies from the boresight of each beam produced by the beamforming network[17].

3.4 Adaptive Antenna Systems

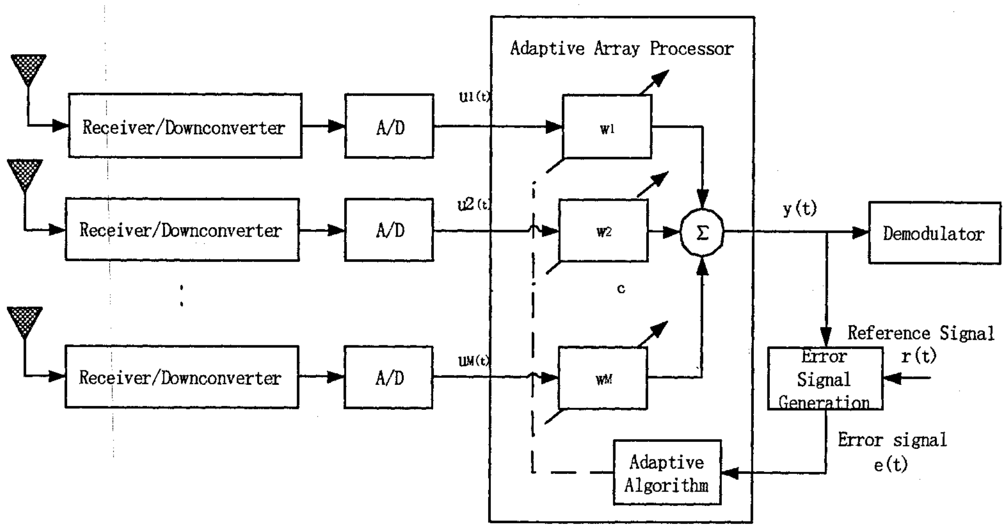


Figure 3.5 Adaptive Antenna System

A greater performance improvement than switched beam systems can be achieved by using adaptive antenna systems at cost of increased complexity in signal processing. A typical adaptive antenna systems is configured as shown in Figure 3.5.

In optimal beamforming techniques, a weight vector, which minimizes a cost function, is determined so as to maximize the quality of signal at the array output. Among those popular adaptive algorithms, Minimum Mean Square Error (MMSE) and Least Squares (LS) criteria have been widely used in communication systems. In both of these techniques, the error signal, $e_j(t)$, square of the difference between the array output, $y_j(t)=W^H{}_j\, u(t)$, and the reference signal r_j (t), a locally estimated desired signal for the jth user, is minimized by adjusting the weight vector W_j. Difference optimal beamforming techniques are talked in detail in [14][15]. We here take MMSE as an example.

- **Minimum Mean Square Error**

Let us assume that a reference signal r (t) can be generated at the receiver. The weights of the array can be chosen in order to minimize the mean square error between the array output y(t) and the reference signal r(t).

$$J(w) = E\{|r(t) - y(t)|^2\} = E\{|r(t) - w^H x(t)|^2\} = E\{|r(t)|^2\} - 2w^H c + w^H C w \tag{3.6}$$

where $c = E\{x(t) r(t)^*\}$ is the cross-correlation vector between the array signal vector and the reference signal, and $C = E\{x(t) x(t)^*\}$ is the correlation matrix,. The optimum weight vector w_{opt} can be computed by setting the derivative of $J(w)$ with respect to w equal to zero. Therefore, we have:

$$\begin{aligned} &\frac{\partial}{\partial w^*} J(w) = -2c + 2Cw = 0 \\ &w_{opt} = C^{-1} c \end{aligned} \tag{3.7}$$

3.5 Capacity Improvement with Adaptive Antenna

Extensive work that has been carried out, involving simulations and field trials, demonstrates that there is major performance improvement achievable for adaptive antenna base station architectures of rather limited complexity in GSM network[18][19].

In GSM, it is possible to increase capacity in at least two ways. One way is to use the SINR gain from adaptive antennas in order to implement tighter frequency reuse than what is used with conventional antennas in a network. For example, multi-beam antennas with four or eight beams in GSM could decrease the frequency reuse factor from 7 to 4, nearly doubling the capacity.

Another way to boost network capacity in GSM is to utilize fractional loading with very tight frequency reuse (typically 1/1 or 1/3). Fractional loading means only a fraction of the frequencies are used simultaneously to maintain the network quality. Using adaptive antennas, the fractional load can be increased, and more traffic can be carried for a certain reuse pattern. From [19], with adaptive antenna, the capacity increase is approximately 280 percent in a 1/1 frequency reuse fractional loading network.

Chapter 4

SDMA and Channel Allocation

4.1 An Introduction to SDMA

SDMA uses the spatial filtering capability of adaptive antennas to allow several users to share the same channel within the same cell, as shown in Figure 4.1.

Since an SDMA system aims at the exploitation of the spatial dimension to increase the number of channels simultaneously available, the capacity improvement depends on the ability of the system to allocate the same channel to several in-cell users. Three major factors determine this ability[20] :

➢ Parameters of adaptive antenna — the spatial filtering capability mainly depends on beamwidth and side lobe level;

➢ Propagation channel — multipath propagation channel with a large angular dispersion reduces the potential number of in-cell co-channel users;

➢ Spatial distribution of users — 'hotspot'(clustered users) makes it difficult to spatially separate users.

In addition, the channel allocation strategy employed also influences the capacity of an SDMA system. An appropriately designed dynamic channel allocation scheme will enable us to exploit the spatial filtering capability more efficiently. In the reminder of this chapter, we first discuss broadly about channel allocation schemes in cellular system, then we focus on particular issues of DCA in SDMA systems.

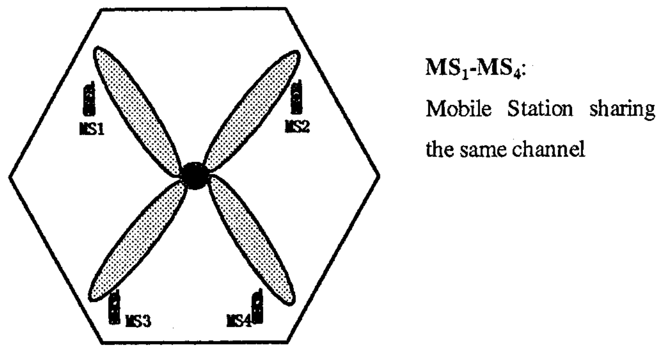


Figure 4.1 An Illustration of SDMA

4.2 Channel Allocation in Cellular Systems

A given radio spectrum can be divided into a set of non-interfering radio channels, and the channel sets can be reused—means that the channels can be used simultaneously while maintaining an acceptable received signal quality. This is the idea of channel allocation. The purpose of channel allocation technique is to minimize the number of required channels to serve all users by reusing channels more efficiently, while meeting some requirements in terms of link and service quality.

As we have known, co-channel interference caused by frequency reuse is the most constraining factor on the capacity of cellular systems, the main idea behind the channel allocation schemes is to make use of the radio propagation path loss characteristic to minimize the signal to interference ratio, thus increase the frequency reuse efficiency and system capacity.

Channel allocation techniques have been extensively investigated for application in cellular communication systems. An extensive survey of most of these techniques can be found in [21].

The allocation techniques can be classified into Fixed Channel Allocation (FCA), Dynamic Channel Allocation (DCA) and Hybrid Channel Allocation (HCA).

In FCA techniques, a fixed number of channels are assigned to the base station of each cell, following some reuse pattern according to desired link quality. A mobile can be allocated only a channel from the channel set assigned to its serving base station. FCA is very simple, it cannot adapt to traffic changes since the number of channels in each cell is fixed. FCA works well when users are uniformly distributed in the network and calls are placed evenly during all the time. But this is usually not the case in reality.

Some variations of FCA schemes have been proposed to cope with traffic changes, such as channel borrowing. In the channel borrowing technique, channels assigned to a base station can be borrowed by its neighboring base station, if necessary.

In DCA techniques, all channels are kept in a central pool and are assigned dynamically to cells as new calls arrive in the system. After a call is released, the channel is returned to the central pool. Because in general, when a call arrives, there are more than one channel eligible for use, the core idea behind all DCA techniques is to evaluate the cost of using each candidate channel, and select the one with the minimum cost provided that certain interference constraints are satisfied. The selection of the cost function differentiates DCA schemes. Since the number of assigned channels to a given cell is flexible, DCA techniques provide the system with more flexibility to cope with time varying traffic.

HCA techniques combine some features of both FCA and DCA techniques. In this class, part of the channel set is used for fixed allocation, while the rest are dynamically allocated. HCA techniques have been proven to give better results than both FCA and DCA over a wide range of traffic load.

Antenna array assisted DCA has proved to provided better performance than DCA with omni-direction antennas. A performance comparison of FCA and DCA with and without beamforming can be found in [22]. A lot of work has been done with the purpose of exploiting the potential of beamingforming to improve the performance of DCA, for example in [23], a new DCA algorithm is proposed to improve the performance by forming a beam pattern to cancel stronger co-channel interference with higher priority.

4.3 DCA and Smart Antenna in SDMA Systems

In SDMA system, the capacity improvement depends on the ability of the system to allocate the same channel to several in-cell users. Therefore, basic constraints must be abided by in-cell users so that they can share the same channel. We introduce two spatial channels models to lay a foundation for our future analysis.

4.3.1 Spatial Channel Model

The application of smart antenna system requires a better understanding of the spatial properties of the wireless communications channel because the spatial properties of the wireless communications channel can have a great impact on the performance of antenna array systems. An understanding of these properties is instructive to system design and evaluation, especially to an SDMA system, which makes use of spatial domain. The spatial domain properties tend to have much to do with the height of transmitting and receiving antennas relative to local environment.

A variety of spatial models existed in the literature. Following [24], they can mainly be classified into three groups:

- General statistically based models
- More site-specific models based on measurement data
- Entirely site-specific models

The first group of model which include Lee's model, Discrete Uniform Distribution Model, Geometrically Based Single Bounce Statistically Model, etc., are useful for general system analysis. The second group models, which include Extended Tap Delay Line Model and Measurement-Based Channel Model, are based on data measurement, and expected to have greater accuracy. The last group of models, such as Ray Tracing, requires a comprehensive description of the physical propagation environment as well as measurement to validate but has potential to be very accurate.

Among these models, the Geometrically Based Single Bounce Statistically Models are widely used in analysis and simulations in wireless systems [24].

4.3.1.1 Geometrically Based Singular Bounce Circular Model

Geometrically Based Singular Bounce Circular Model is used to depict the spatial properties of a macrocell environment.

Figure 4.2 shows the signal propagation in a macrocell environment. It is usually assumed that the scatterers surrounding the mobile station are about the same height as or are higher than the mobile. This implies that the received signal at the mobile station antenna arrives from all directions after bouncing from the surrounding scatterers, as illustrated in Figure 4.2.

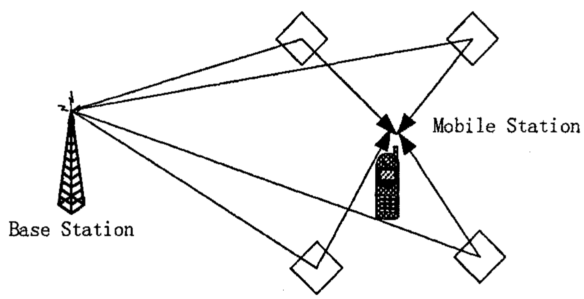


Figure 4.2 Propagations in Macrocell Environment

The Geometrically Based Singular Bounce Circular Model proposed by Petrus *et al.* [25][26], is based on the assumption that, in a macrocell environment, there will be no signal scattering from locations near the base station antenna. The radius of the scatterers characterizes the angle of arrival profile of the system. All the scatterers fall within the maximum radius specified, and multipath occurs due to scattering from objects within this radius.

Figure 4.3 shows the geometrical representation of this model. The scatterers lie inside a circle of radius r when the separation between the mobile and the base station is d.

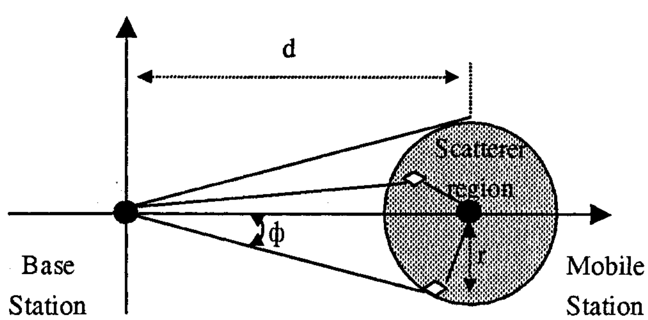


Figure 4.3 Illustration of Geometrically Based Single Bounce Circular Model

It is presented that the angular spread α is inversely proportional to the T-R separation, and radius of scatterers range from 30 to 200m. According to [27], the active scattering region around the mobile is about 100-200 λ (wavelengths of the signal transmitted).

The maximum angle of arrival at base station is:

$$\phi_{\max} = \sin^{-1}\left(\frac{r}{d}\right) \tag{4.1}$$

4.3.1.2 Geometrically Based Single Bounce Elliptical Model

Geometrically Based Single Bounce Elliptical Model is usually used in the microcell environment. In the microcell environment, the base station antenna is usually mounted at the same height as the surrounding objects. This implies that the scattering spread of the received signal at the base station is larger than in the macrocell case since the scattering also happens in the vicinity of the base station. The geometry of this model is shown in Figure 4.4, where the base station and the mobile are located at the foci of an ellipse scatterer region.

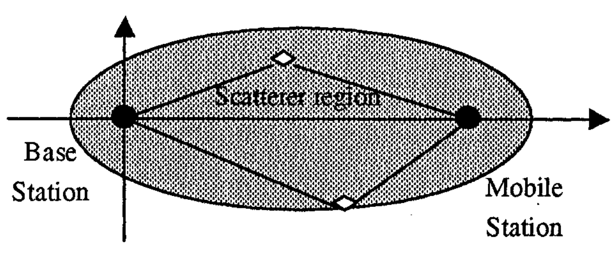


Figure 4.4 Illustration of Geometrically Based Single Bounce Elliptical

4.3.2 Constraints in SDMA

For the application of SDMA, two constraints have to be imposed [28]:

- Between any two users sharing the same traffic channel, a minimum angular distance must be guaranteed. This angular constraint can be derived from the beamwidth and the angular spread of multipath components of interfering co-channel users.

- Between any two users sharing the same traffic channel, for the difference between the received signal powers of any two mobile users, an upper limit exists to guarantee the required SINR in order that weak signals will not be interfered by strong co-channel signals. This is due to the limited null depth of the beam pattern. Assuming an exponential propagation law, the requirement of power level differences can be translated into an upper limit of the ratio between the distances.

4.3.2.1 Angular Constraint

An example of the angular constraint is depicted in Figure 4.5, where MS1 may suffer from interference caused by its co-channel user MS2.

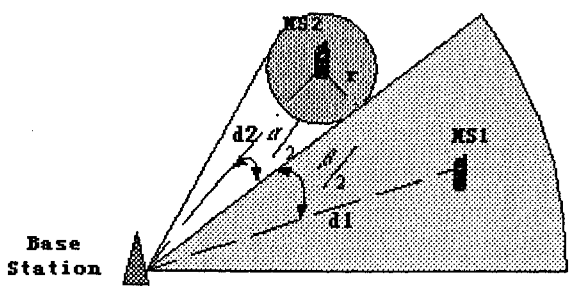


Figure 4.5 Illustration of Angular Constraint in SDMA

In case of a macrocell environment, where Geometrically Based Singular Bounce Circular Model can be applied, the signals received from MS2 are coming from a circular area with the radius r=200λ. If the T-R distance is $d2$, the angular spread α seen by a base station receiver is:

$$\alpha = 2\sin^{-1}\left(\frac{r}{d2}\right) \tag{4.2}$$

Therefore, as shown in Figure 4.5, suppose the beamwidth of the antenna array is β, to avoid the collision, the minimum angular distance between MS1 and MS2 is:

$$\Phi \min=\alpha/2+\beta/2 \tag{4.3}$$

Since α depends on the distance $d2$ of MS2, near base station interfering mobiles require larger minimum angular distances Φ min.

The effect of multipath spreading on MS1 is assumed to be negligible since usually $\alpha<\beta$. To be more precise, β in the above equation should be replaced with max(α ,β).

4.3.2.2 Distance Constraint

Because the sidelobe level (SLL) and the depth of nulls of the radiation pattern depend on the parameters of the array, there is a limit on the attenuation of the in-cell co-channel signals. If an in-cell co-channel mobile is too close to the base station, or if the desired user is too far from its serving base station, due to the limited side lobe level and the antenna gain, the SINR requirement of the desired user may not be satisfied. Hence there is a lower limit of distance to base station for co-channel interfering mobile, denoted by *dmin*; and there is an upper limit of distance to base station for the desired user, denoted by *dmax*. The ratio of *dmin* and *dmax* must meet a certain constraint to guarantee the SINR requirement.

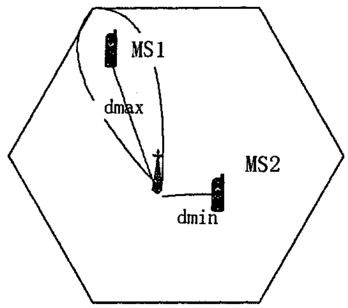


Figure 4.6 Illustration of Distance Constraint in SDMA

Consider two mobile users in one cell, as illustrated in Figure 4.6. Assume that MS1 is the desired user, using the exponential propagation law, we have the received signal power:

$$P_r = Po \cdot (\frac{d\max}{do})^{-n} \tag{4.4}$$

Where *Po* is the power received at a close-in reference point at a small distance *do* from the transmitting antenna, and n is the path loss exponent, which depends on the specific propagation environment; *dmax* is the T-R separation.

Suppose that no power control is applied, so each mobile transmits with the same power, and the sidelobe level is S_l, the interference signal power received from MS2 is:

$$I = S_l \cdot Po \cdot (\frac{d\min}{do})^{-n} \tag{4.5}$$

Suppose that the minimum acceptable signal-to-interference ratio is $SINR_0$. Therefore, we want:

$$\frac{S}{I} \geq SINR_0 \tag{4.6}$$

Substituting expressions (4.3) and (4.4) into (4.5), we get:

$$\frac{S}{I} = \frac{1}{S_I}\left(\frac{d\max}{d\min}\right)^{-n} \geq SINR_0 \tag{4.7}$$

The distance constraint is:

$$\frac{d\min}{d\max} \geq (SINR_0 \cdot S_I)^{\frac{1}{n}} \tag{4.8}$$

Because in our simulations, only downlink performance is evaluated, the distance constraint is not applied.

4.3.3 DCA in SDMA system

In order to efficiently exploit the potential capacity improvement provided by spatial filtering, an appropriate channel allocation strategy is desired in SDMA system.

A lot of research has been done in this area. Some of the algorithms can be found in references [20][29].

Four channel allocation algorithms are analyzed in [20]: Autonomous Reuse Partitioning Algorithm (ARP), Least Interference Algorithm (LIA), Concentrated Channel Load Algorithm (CCL), and Equal Channel Load Algorithm (ECL). The former two apply to both SDMA and non-SDMA systems, while CCL and ECL represent two basic approaches for assigning channels in SDMA systems.

- **Least Interfered Algorithm (LIA)**

In non SDMA mode, the Least Interference Algorithm (LIA) [30] assigns the least interfered channel, among the idle channels at the base station. All idle channels are measured, and the channel with the highest SINR above the required SINR threshold will be assigned to the mobile. If no channel is found, the call is blocked. The same approach is used for intercell handoff calls and channel reassignment, but SINR threshold may be different.

In its SDMA version, all channels are measured, including those already used in the cell. LIA tend to select idle channels before selecting a channel already in used in the cell. Since LIA attempts to minimize the interference on all channels, the overall co-channel interference is expected to be low; hence, channel assignment requests are expected to be less.

- **Autonomous Partitioning Reuse Algorithm (ARP)**

Autonomous Reuse Partitioning (ARP) [31], is based on measuring channels following an order that is common throughout all cells. The first channel that presents interference levels below a given threshold is assigned. When a call arrives at the system, it selects the first channel found that meets the condition SINR>SINR required. The same approach is used for handoff calls and channel reassignment. A consequence of this allocation strategy is that channels sensed first are reused more often in the system and, therefore, experience larger interference. Since the level of interference on those channels is high, only mobiles with strong desired signals can use those channels. Mobiles with strong signals usually are those close to their serving base stations. On the other hand, channels that are measured later are used fewer times in the system.

In its SDMA version, all channels are measured, including those already used in the cell. This scheme has proved to have a better overall channel reuse efficiency, but more channel reassignment may be needed.

- **Concentrated Channel Load Algorithm (CCL)**

CCL tries to allocate the channel with the largest number of in-cell co-channel users at that moment at the serving base station while satisfying the SINR requirement. Since the algorithm tries to reuse channels within cells as much as possible, it may lead to high level of co-channel interference, and, consequently, a large number of channel reassignment requests.

- **Equal Channel Load Algorithm (ECL)**

ECL allocates the channel least used within a cell, among all channels satisfying the performance requirement. This algorithm tries to use all idle channels before reusing a channel within the cell, thus a low level of interference is expected.

The channel allocation strategy employed in our simulations belongs to Least Interfered Algorithm. The detailed description of the algorithms will be given in Chapter 6.

Chapter 5

Simulation Models

This chapter presents the models used in our simulations. Models are important because the accuracy of the simulation depends much on the accuracy of the mathematic models. These models include: the cellular network model which include a cell model and network mode; a propagation model which explains the exponential propagation law and log-normal shadowing; an antenna model and spatial channel model, an interference calculation and a user mobility model. The purpose of these models is to provide us a relatively realistic simulation environment while keeping the complexity to a reasonable level.

The application of these models depends on our specific algorithm or requirement. Because we have three parts of simulations, within which there may be different algorithms that require different models, as we will mention explicitly in chapter 7.

5.1 Cellular Network Model

5.1.1 Cell Model

The actual coverage of a cell is called the footprint and is determined from field measurement or propagation prediction models. Even though in reality the footprint is

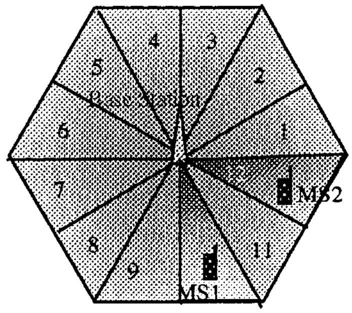


Figure 5.1 Cell Model

amorphous, a regular cell shape is needed for system design and performance analysis. When considering geometric shapes that cover the entire region without overlap and with equal area, there are three preferable choices: a square, an equilateral triangle, and a hexagon [1]. We choose hexagonal center-exited cell model, i.e. the base station is located at the center of the cell.

We divide the cell into 12 virtual sectors, each corresponding to an area covered by a beam with beamwidth 30^0. This can be regarded as an ideal case of switched beam antenna system, with 12 beams each steering at its fixed direction but no overlap in-between. A set of channels (unless specifically mentioned, a channel is referred to as a timeslot in our simulated network hereafter) can be used in every sector as long as SINR requirement is satisfied. As shown in Figure 5.1, MS1 in sector 10 can share the same channel with MS2 in sector 12.

5.1.2 Network Model

The size of the network in terms of number of cells is an important issue when simulating a cellular network. Considering especially the co-channel interference, the simulated network must be large enough to emulate the behavior of a real cellular system. The average level of interference is desired to be uniform over all cells. When a finite bounded area is used to simulate the coverage area of a cellular network, it is obvious that the average interference measured at center cells is higher than at cells close to the boundaries of the coverage area, i.e. the central cells experiencing more unfavorable conditions than the edge since there are no mobiles roaming outside of the edge cells. This may affect the simulation results.

A solution for mitigating the effects of different levels of average interference is to simulate a coverage area with a larger number of cells, but collect the performance measurements only at central cells. This solution is not efficient, since only central cells are effectively used among all the cells simulated. In addition, a large number of cells requires long simulation time, which is not desired.

In order to avoid 'edge effects', in our simulation, 'wrap around' technique is used, thus allowing the simulation area to be replicated around itself, as shown by the Figure 5.2. Mobile stations and their signals are 'wrapped around' from one side of the network to the other: each cell is seemed to be in the center of the network, and it can 'see' all other cells. In this case, a mobile station that has left the network at its edge reentered the network on the opposite side, while imposing co-channel interference on all users who are located at either edge of the network. Therefore, calls are not lost because mobiles move out of the coverage area. The interference level is uniform among all cells within the network. A similar idea of 'toroidal universe of cell" can be found in [20].

As the case shown in the Figure 5.2, the edge cells on the one side are connected with those on the opposite side; the users who move out of cell 10 reenter cell 14, or move from cell 12 to cell 8.

A total of 19 cells are used in our simulation, and the idea of 'wrap around' is also illustrated in the Table 5.1.

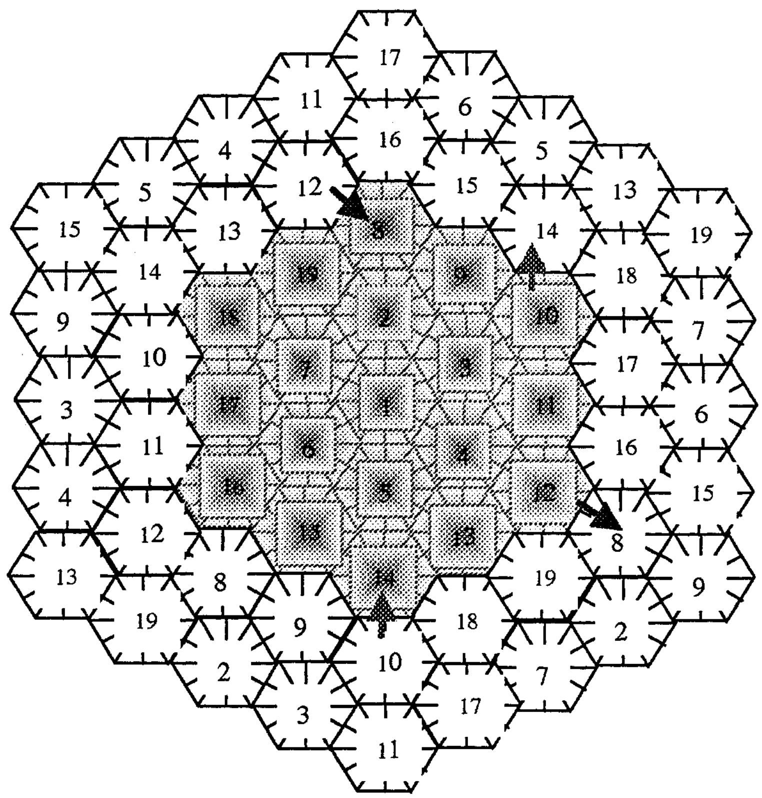


Figure 5.2 The Cellular Network Model

In the table, column 1 shows the 'base' network. When a user comes into cell 1, he sees the network as it is--the center cell is cell 1, whose surrounding cells are from cell 2 to cell 19, with cell 2 to cell 7 the first tier cells, and cell 8 to cell 19 the second tier cells.

When a user comes to cell2, he sees the network as shown in column 2. Cell 2 becomes the center cell, and the first tier cell 8,9,3,1,7 and 19 is to cell 2 what cell 2 to cell 7 is to cell 1, and the second tier cell 16, 15, 14, 10, 11, 4, 5, 6, 17, 18, 13, and 12 is to cell 2 what cell 8 to cell 19 is to cell 1, and so on, so forth.

No matter which cell is the center cell, the absolute No. of the first tier and the second tier cells can be mapped into relative No. of cell 2 to cell 19 respectively, as shown in the table5.1. The center cell will suffer from the interference from its surrounding first tier and second tier cells.

Table 5.1 Cell Mapping Table

Center Cell	Cell1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19
First Tier Cells	Cell2	8	9	3	1	7	19	*16*	*15*	*14*	10	11	4	5	6	17	18	*13*	*12*
	Cell3	9	10	11	4	1	2	*15*	*14*	*18*	*17*	*16*	12	13	5	6	7	19	8
	Cell4	3	11	12	13	5	1	9	10	*17*	*16*	*8*	*19*	*18*	14	15	6	7	2
	Cell5	1	4	13	14	15	6	2	3	11	12	*19*	*18*	*10*	*9*	*8*	16	17	7
	Cell6	7	1	5	15	16	17	19	2	3	4	13	14	*9*	*8*	*12*	*11*	*10*	18
	Cell7	19	2	1	6	17	18	*12*	8	9	3	4	5	15	16	*11*	*10*	*14*	*13*
Second Tier Cells	Cell8	16	*15*	9	2	19	*12*	*17*	*6*	*5*	*14*	10	3	1	7	18	*13*	*4*	*11*
	Cell9	15	*14*	10	3	2	8	*6*	*5*	*13*	*18*	*17*	11	4	1	7	19	*12*	*16*
	Cell10	14	*18*	*17*	11	3	9	*5*	*13*	*19*	*7*	*6*	*16*	12	4	1	2	8	*15*
	Cell11	10	*17*	*16*	12	4	3	*14*	*18*	*7*	*6*	*15*	*8*	*19*	13	5	1	2	9
	Cell12	11	*16*	*8*	*19*	13	4	10	*17*	*6*	*15*	*9*	*2*	*7*	*18*	14	5	1	3
	Cell13	4	12	*19*	*18*	14	5	3	11	*16*	*8*	*2*	*7*	*17*	*10*	*9*	15	6	1
	Cell14	5	13	*18*	*10*	*9*	15	1	4	12	*19*	*7*	*17*	*11*	*3*	*2*	*8*	16	6
	Cell15	6	5	14	*9*	*8*	16	7	1	4	13	*18*	*10*	*3*	*2*	*19*	*12*	*11*	17
	Cell16	17	6	15	*8*	*12*	*11*	18	7	1	5	14	*9*	*2*	*19*	*13*	*4*	*3*	*10*
	Cell17	18	7	6	16	*11*	*10*	*13*	19	2	1	5	15	*8*	*12*	*4*	*3*	*9*	*14*
	Cell18	13	19	7	17	*10*	*14*	*4*	*12*	8	2	1	6	16	*11*	*3*	*9*	*15*	*5*
	Cell19	12	8	2	7	18	*13*	*11*	*16*	*15*	9	3	1	6	17	*10*	*14*	*5*	*4*

Note: the italic number of cells means due to 'wrapped around'.

5.2 Propagation

5.2.1 Exponential Propagation Law

Propagation measurements in a mobile radio channel show that the average received signal strength at any point decays as a power law of the distance of T-R separation -- separation between a transmitter and a receiver. The average received power $\overline{Pr}$ at a distance of *d* from the transmitting antenna is approximate by:

$$\overline{\mathrm{Pr}} = P_0 (\frac{d}{d_0})^{-n} \tag{5.1}$$

or

$$\overline{\mathrm{Pr}}(dBm) = P_0(dBm) - 10n\log(\frac{d}{d_0}) \tag{5.2}$$

Where *Po* is the power received at a close-in reference 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, as listed in Table 5.2 [1].

Table5.2 Typical Path Loss Exponents for Different Environments[1]

Environment	Pass Loss Exponent (n)
Free space	2
Urban area cellular radio	2.7 to 3.5
Shadowed urban cellular radio	3 to 5
In building line-of-sight	1.6 to 1.8
Obstructed in building	4 to 6
Obstructed in factories	2 to 3

5.2.2 Log-normal Shadowing

The propagation model in equation 5.1 is valid for line of sight transmission. It doesn't take into consideration that for the same T-R separation, the surrounding environment may quite different, causing the measured signal different from the average value predicted by the equation. Measurement has shown that, at any particular T-R separation *d*, the receiving signal level is random and distributed log-normally (normal in dB) about the mean distance-dependent value:

$$Pr = \overline{Pr} + \chi_\sigma \tag{5.3}$$

Where $\overline{Pr}$ is the mean received power, which gets from the equation 5.1, and χ_σ is a zero mean Gaussian distributed random variable (in dB) with standard deviation σ (in dB).

Usually σ is chosen from 6dB to 10dB. We use 8dB in the simulation.

5.3 Co-channel Interference

In simulation, only down-link performance is evaluated. The ratio of signal to interference SINR is calculated and used for channel allocation.

Let *n* be the number of co-channel interfering resources, the signal to interference ratio for a mobile receiver is:

$$\frac{S}{I} = \frac{S}{\sum_{i=1}^{n} I_i} \tag{5.4}$$

Where the nominator S is the desired signal power and the denominator is the sum of interfering power from all co-channel base stations. Throughout our simulations, we consider both the first tier and second tier of co-channel cells.

We assume that the interference signals add incoherently, so that the powers can be summed.

In our case, the SINR is calculated as

$$\frac{S}{I}=\frac{S}{\sum I_{\text{intercell}}+\sum I_{\text{intracell}}} \tag{5.5}$$

Where $I_{intracell}$ means interference coming from in-cell co-channel users, that is, coming from the sidelobe of the co-channel beams within the same cell. $I_{intercell}$ means interference coming from other cells including both main beam interference and sidelobe interference. By main beam interference, we mean that the ongoing call is within the main lobe of the co-channel beam; while the sidelobe interference means that the ongoing call is outside of the angle covered by the main lobe of the co-channel beam. We will explain this in the coming section. Background noise in the network is neglected.

5.3.1 Main Beam and Sidelobe Interference

The co-channel interference received at the mobile, caused by a given co-channel base station, is attenuated when the mobile is not within the main lobe of the antenna of that co-channel base station transmission.

It is obvious that the extent to which the co-channel interference is reduced depends on the beamwidth and the side lobe level of the base station antennas. For the antenna array, the beamwidth and sidelobe level will have much to do with the number of elements in the array.

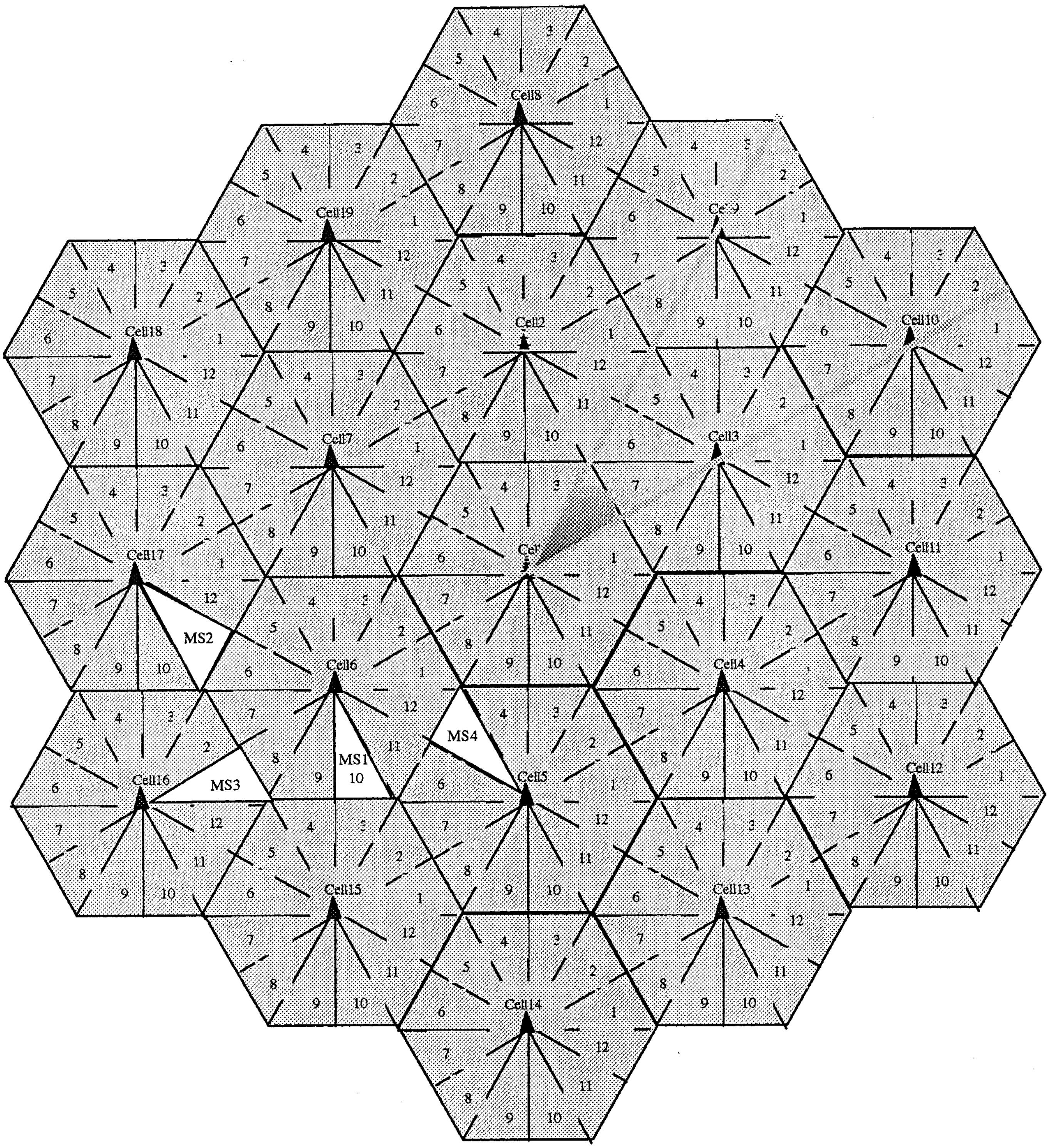


Figure 5.3 Mainbeam and Sidelobe Co-channel Interference

Table 5.3 Interfering Sector Table

Center Cell Sector No.	Interfering Sector No. of First Tier Cells						Interfering Sector No. of Second Tier Cells											
	Cell 2	Cell 3	Cell 4	Cell 5	Cell 6	Cell 7	Cell 8	Cell 9	Cell 10	Cell 11	Cell 12	Cell 13	Cell 14	Cell 15	Cell 16	Cell 17	Cell 18	Cell 19
1	10	8	5	3	1	12	10	9	8	6	5	4	3	2	1	1	12	11
2	10	7	5	3	2	12	10	9	7	6	5	4	3	2	2	1	12	11
3	10	7	5	3	2	12	10	8	7	6	5	4	3	3	2	1	12	11
4	9	7	5	4	2	12	9	8	7	6	5	4	4	3	2	1	12	11
5	9	7	5	4	2	12	9	8	7	6	5	5	4	3	2	1	12	10
6	9	7	6	4	2	11	9	8	7	6	6	5	4	3	2	1	11	10
7	9	7	6	4	2	11	9	8	7	7	6	5	4	3	2	12	11	10
8	9	8	6	4	1	11	9	8	8	7	6	5	4	3	1	12	11	10
9	9	8	6	4	1	11	9	9	8	7	6	5	4	2	1	12	11	10
10	10	8	6	3	1	11	10	9	8	7	6	5	3	2	1	12	11	10
11	10	8	6	3	1	11	10	9	8	7	6	4	3	2	1	12	11	11
12	10	8	5	3	1	12	10	9	8	7	5	4	3	2	1	12	12	11

Note: The Cell No. is the relative No. toward the center cell.

For example, in our cellular network model, as shown in the Figure 5.3: beam (sector) 2 of cell1 will have mainbeam interference on the co-channel users in sector 2,3,4,5,6,7 of cell 3 and cell 10, and sector 9,10,11,12,1,2 of cell 9. Co-channel users in the rest cells of the network will suffer from sidelobe interference.

Take another example in Figure 5.3. Suppose MS1—MS4 are co-channel users. MS1 will suffer mainbeam interference from the base stations of MS2 and MS3, and sidelobe interference from the base station of MS4. Meanwhile, what MS2 or MS3 suffers is all sidelobe interference from other three base stations. MS4 gets mainbeam interference only from the base station of MS3, but sidelobe interference from the rest of the two base stations.

A detailed interfering relations between sectors in different cells is as shown in Table 5.3. The first column of the table consists the sector no. of the center cell, which can be one of any 19 cells. Sectors that are in the same row with a sector in first column are this sector's corresponding main beam interfering sectors of the first and second tier cells relative to the center cell (relative no. of cells is as described in chapter 5.1.2). For example, sector 9 of cell 2 will have main lobe interference on sector 4,5,6,7,8,9 of cell 1; sector 10 of cell 2 will have main lobe interference on sector 1,2,3,10,11,12 of cell 1.

5.4 Antenna Model and Spatial Channel Model

In simulation, a brickwall antenna radiation pattern, which is the worst case, is assumed, as shown in the Figure 5.4:

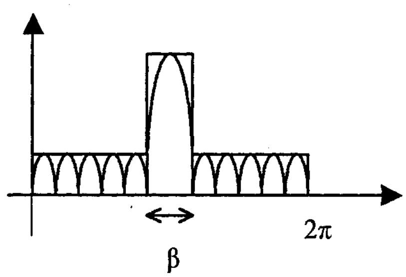


Figure 5.4 Brickwall Antenna Pattern

Therefore, the directions out of main beam angle will all suffer from sidelobe interference.

In reality, two types of antenna array-- the linear array and the circular array--are widely used. Linear array has two major sidelobes, whereas the circular array has one major sidelobe.

As presented in section 5.1.1, beamwidth β=30 is chosen. There are 12 beams (sectors) in each cell. Different sidelobe levels are used in simulations. The maximum antenna gain within the main lobe is set to 0 dB, which is a valid simplification since SINR, not absolute power levels, form the basis of the system performance specification. We use different SLL in our simulations. For example, by SLL=20dB, we mean that the side lobe power is 1/100 of the main lobe power. All base stations and mobile stations are assumed to transmit with identical powers, respectively.

The Geometrically Based Singular Bounce Circular Model, described in section 4.2.1, is used in the simulation. We have $\beta/2 = 15^0$. Following Lee [27], the radius r of the scatterers is 200λ, where λ is the wavelength of the transmitted signal. Given 900MHz spectrum in GSM, we have λ= 0.3m, r=200λ=60m.

The dispersion angle α is:

$$\alpha/2 = \sin^{-1} 60/d \qquad (5.6)$$

Where d is the distance between the mobile user and the base station.

This is the dispersion angle we used in our simulations, when multipath phenomenon is taken into consideration.

5.5 User mobility

One of the important models that have to be studied and implemented is the user mobility. Since SDMA utilizes the spatial dimension, user mobility can have big influence on the performance.

In particular, two parameters are of interest: (i) location of the user when the call is started; (ii) velocity with which the user is moving and his direction.

Some previous works have attempted to characterize different kinds of user movement. A comprehensive mathematical treatment of user movement within the cell, including channel holding time and average number of handoffs, can be found in [32]. The impact of some mobility parameters such as the speed of the mobile and the cell size on the performance of cellular networks in the presence of smart antenna can be found in [33].

For the present study, we consider 4 kinds of users:

(1) High speed users: This class of users has high velocity and move at a constant angular direction or a constant distance direction. These users emulate highway users with speed 72km/h.

(2) Low speed users: This class of users moves at relatively low velocity at a constant angular direction or a constant distance direction. These users emulate city drivers with speed 36km/h.

(3) Pedestrian Users: This class of users represents walking mobile users with speed 3.6km/h.

(4) Stationary users: This class of users has zero velocity. They are immobile during the call.

Mobile users in a cell are a mix of the above-mentioned classes. We assume that each class of users is generated randomly in our simulation, though the mix of users may depend on the location, time, day of the week etc.

Doppler effect[1] , which may result from user mobility, is neglected.

Chapter 6

Simulation Description and Result Discussion

6.1 Overall Description

Using the model described in section 5.1, a cellular network of 19 hexagonal cells with cell radius 3km is simulated. A channel pool of 6 carriers, each having 8 timeslots, thus giving us maximum 48 channels, is used in the whole system. This channel pool is supported by every cell. A complete dynamic channel allocation scheme is used. SDMA is applied. Each cell includes 12 virtual sectors. Mobile users can use all 48 channels in every beam within a cell, as long as the SINR requirement is met.

Though there are only 48 channels in the whole network, these 48 channels can be used in every cell—and actually in every sector, therefore, a channel pool of $48\times12=576$ channels is obtained in each cell. And the maximum channel resource pool in the whole network can be considered to be $576\times19=10944$ channels! The channel resources pool is as shown by table 6.1. In table 6.1, the first column (id) consists of all channels numbered 1-10944 in the pool; the second column (cid) contains channels with channel id 1-48, which are repeatable in each sector. Users using the channel with the same cid are co-channel users.

For channel allocation, we employ the least interference first algorithm--the channel with the best SINR among eligible channels is assigned to a new call. This channel allocation algorithm should be based on actual SINR measurement, which is reasonably represented by calculation in our simulation. Call admission control is implemented to guarantee the quality of service requested by new users.

The following assumptions are made in the simulations:

1) There is perfect spatial filtering between adjacent sectors, i.e., no beam collision between adjacent sectors.

2) The antenna gain is uniform within one sector, and scalloping effect is neglected.

3) All base stations transmit with the same power. Power control is not considered.

4) The exact locations of all mobile users are known to the network.

5) The instantaneous channel information is available to all cells in the network. This requires exchange of large amount of information between base stations (BS).

6) The network is synchronous from cell to cell, thus channels on different time slots of the same frequency are orthogonal.

7) There is perfect filtering between adjacent channels, so adjacent channel interference can be neglected.

8) Users cannot use multi-carrier channels, i.e., timeslots belonging to different carriers cannot be allocated to one user so as to ease frequency synthesis problem at MS.

9) EDGE network with perfect link quality is assumed. The maximum data rate per timeslot can carry is 48kbps, providing 384kbps with the 8 slots on each available carrier. 52 multiframe structure is used. Each multiframe lasts 240 ms.

Table 6.1 Channel Resource Table

id	cid	cellno	sectorno	frequency	timeslot
1	1	1	1	1	1
2	2	1	1	1	2
3	3	1	1	1	3
4	4	1	1	1	4
5	5	1	1	1	5
6	6	1	1	1	6
7	7	1	1	1	7
8	8	1	1	1	8
9	9	1	1	2	1
10	10	1	1	2	2
11	11	1	1	2	3
12	12	1	1	2	4
13	13	1	1	2	5
14	14	1	1	2	6
15	15	1	1	2	7
16	16	1	1	2	8
17	17	1	1	3	1
18	18	1	1	3	2
19	19	1	1	3	3
20	20	1	1	3	4
21	21	1	1	3	5
22	22	1	1	3	6
23	23	1	1	3	7
24	24	1	1	3	8
25	25	1	1	4	1
26	26	1	1	4	2
⋮	⋮	⋮	⋮	⋮	⋮
47	47	1	1	6	7
48	48	1	1	6	8
49	1	1	2	1	1
50	2	1	2	1	2
⋮	⋮	⋮	⋮	⋮	⋮
10942	46	19	12	6	6
10943	47	19	12	6	7
10944	48	19	12	6	8

10) There are only packet users with equal priority in the network. Yet we are trying to represent different packet data users with different rate, and scope of rate variation.

6.2 Performance Measurement

6.2.1 Quality of Service (QoS)

QoS implies differentiation in services available to users according to, for instance, the requirements of a particular type of traffic. The traffic type can be real time voice, streaming video, video conferencing, file transfer, real time video, etc. The mobile user and the network will agree on a particular QoS profile during the initial service negotiation stage and the network will attempt to deliver this QoS.

Therefore, the quality of service supported by the network depends on the type of service. Conventionally, call blocking rate and call dropping rate are used to define the quality of service provided by GSM network. But for packet services, the Quality of Service profile is different, normally includes the delay class, the priority class, the reliability class, the peak and average throughput class.

For a 3G system, services are broadly classified into four classes: conversational, streaming, interactive and background [34]. Each has different a Quality of Service requirement.

6.2.2 Performance Measurement in Our Simulations

in our simulations. Since our simulations just dealt with admission control of radio access network, only the performance of this part, instead of end-to-end data transmission, is evaluated. With these measurements, though may not be standard methods, we hope to give an idea of Quality of Service provided by the network under different situations.

6.2.2.1 Call Blocking Probability

Upon arrival of a fresh call, the respective base station assigns a channel to it according to a specific strategy. Different channel allocation algorithm in our simulation will be described later on in this chapter. If no channel can be assigned, then the call has to be blocked, as a result of insufficient channels or as a consequence of low SINR level. If the requesting user is blocked without access, this is called blocked call cleared; if the call request may be accepted but has to wait for a certain time until a channel becomes available, this is called blocked call delayed.

In our measurement, a cleared call-- a call that is not admitted to the system, is regarded as blocked. Call Blocking Probability *Pb* is defined as:

$$Pb = \frac{Number\ of\ blocked\ calls}{Total\ Number\ of\ Arrival\ Calls} \times 100\% \qquad (6.1)$$

6.2.2.2 Call Dropping Probability

An ongoing call may need to release its old channel(s) and request channel(s) reassignment if the mobile user moves to another cell or sector, or collide with other co-channel users. Due to mobility, the SINR of a user may deteriorate to below threshold. If an ongoing call cannot find new channel after it has released its channel for 2.4 second (10 multiframes), or suffers from low SINR for 4.8s (20 multiframes), the call is dropped.

The Call Dropping Probability *Pd* is defined as :

$$Pb = \frac{Number\ \ of\ \ Dropped\ \ Calls}{Total\ \ Number\ \ of\ \ Admitted\ \ Calls} \times 100\% \tag{6.2}$$

6.2.2.3 Average Normalized Delay

Packet delay in a packed-switched network normally consists of three components: transmission delay, retransmission delay as well as queuing delay, the time during which the packet has no access to a channel.

In our simulation, only queuing delay is calculated. Average Normalized Delay *Dn* denotes that, for each Kbits data transmitted, how much delay (in second) is incurred. It is defined as:

$$Dn = \frac{\sum_{j=1}^{N} \left[\frac{\sum_{i=1}^{Mj} D(j,i)}{\sum_{i=1}^{Mj} P_s(j,i) \cdot L_p} \right]}{N} \left[\frac{s}{Kb} \right] \tag{6.3}$$

$D(j,i)$ (in second) equals to $P_{bu}(j,i) \times T$, where $P_{bu}(j,i)$ is the number of packets in buffer at iteration *i*, and T (in second) is the duration of the iteration. Therefore, $D(j,i)$ is the delay of the call *j* at iteration *i*. Iteration is a time unit used in our programming; $P_s(j,i)$ is the packets got served for user *j* at iteration *i*, and L_p (in kb) is the size of the packet; M_j is the total number of iteration during which user *j* is 'online' , and *N* is the total number of online users.

6.2.2.4 Spectral Efficiency and Slot Capacity

The Spectral Efficiency is normally used to measure the system capacity for packet data traffic while the system fulfils the satisfied user requirements,

We use this in the simulation to evaluate the system performance under different carried load. The Spectral Efficiency *Se* is defined as:

$$Se = \frac{Sum\ of\ Transmitted\ Bits}{(SimulationTime) \times (Number\ of\ Cells) \times (Spectrum)} \times 100\% \left[\frac{Bps}{Cell \cdot Hz}\right] \quad (6.4)$$

Where Spectrum is the spectrum allocated to the whole system, which is equal to 200kHz/carrier × 6carrier=1200kHz in our simulated network.

An alternative capacity measure denoted Timeslot Capacity can also be used. Timeslot Capacity *Tc* is defined as the maximum carried traffic per timeslot, measured in kbps. The relation between the Spectral Efficiency and the Timeslot Capacity is defined as:

$$Tc = Spectral\ Efficiency\ \times Timeslot\ Spectrum\ \ [\frac{kbps}{Cell \cdot Slot}] \quad (6.5)$$

Since the timeslot spectrum is 200kHz per carrier / 8 time slots per carrier = 25kHz. Thus, if the Spectral Efficiency is 0.5 bps/Hz/Cell, the Timeslot Capacity is 0.5bps/Hz/Cell × 25kHz = 12.5kbps.

6.2.2.5 Average Buffer Overflow Rate

For packet data users, each call has its own buffer. The packets that have been generated by users but cannot get access to the network will be left in the buffer. If the packets in buffer are larger than the buffer size, the buffer will overflow, and extra packets generated above the buffer size will be lost.

The Average Buffer Overflow Rate *Op* is defined as:

$$Op = \frac{\sum_{j=1}^{N}\left[\frac{\sum_{i=1}^{Mj} O(j,i)}{M_j}\right]}{N} \times 100\% \tag{6.6}$$

Where *O(j,i)=1* whenever user *j* has buffer overflow at iteration *i*, *N* means total number of online users; M_j stands for total number of iteration during which user j is calling.

6.2.2.6 Average Throughput per Slot

The Average Throughput here denotes the average data rate served by one time slot. The packets generated by each user are successfully served if there are resources available and the resources employed meet the user's SINR requirement. The value of average throughput is normally less than 48 kbps, which is the maximum data rate carried by one channel. The Average Throughput *Rav* is defined as:

$$Rav = \frac{\sum_{j=1}^{N}\left[\frac{\sum_{i=1}^{Mj}[Ps(j,i)]}{S_j} \cdot \frac{Lp}{M_j \cdot T}\right]}{N}\left[\frac{Kbps}{Slot}\right] \tag{6.7}$$

Where *Ps(j,i)* is the number of packets served at iteration *i* for user *j*; *N* is the total number of online users; L_p is the length of each packet (in kb); M_j is the total number of iteration of user *j*; *T* is the time duration of each iteration (in second), and S_j is the average number of slots assigned to this user.

6.2.2.7 Handoff

If the channel assignment has been successful, the call remains on that channel until call completion. Handoff is performed when one of the following events occurs: the signal quality SINR falls below a threshold; the spatial separation between two mobile users is no longer sufficient; mobile users move to another cell or another sector.

The call is switched to a newly allocated channel upon a successful handoff. If the new channel belongs to the same base station, this is referred to as an intra-cell handoff. This is the case when a user move to another sector within the same cell, collide with another in-cell co-channel user, or when the channel quality degrades due to the signal strength degrades when the user has moved too far from the base station. Inter-cell handoff is performed when users move to another cell.

Total handoff Ht including both intra-cell and inter-cell handoff is defined as:

$$Ht = \sum_{j=1}^{N}\sum_{i=1}^{M} H(j,i) \tag{6.8}$$

Where $H(j,i)$ stands for the number of handoff of user j at iteration i.

The more the handoffs, the more intensive computation is needed by the network.

6.2.2.8 Average Packet Loss Rate

Due to buffer overflow, some data packets are lost before being served by the network.

Average Packet Loss Rate P_l is used to denote what percentage of packets is lost due to buffer overflow. It is defined as:

$$P_l = \frac{\sum_{j=1}^{N} \{ [\sum_{i=1}^{Mj} Pg(j,i) - \sum_{i=1}^{Mj} Ps(j,i)] \Big/ \sum_{i=1}^{Mj} Pg(j,i) \}}{N} \times 100\% \tag{6.9}$$

Where Ps is number of packets that have been served successfully, whereas Pg is number if data packets generated by calls.

6.2.2.9 Normalized Carried Load

When measuring the traffic intensity carried by our simulated network, we use the system with 1/1 frequency reuse factor as reference. If considering our modeled network as 1/1 frequency reuse system, we have a set of 48 channels repeatable in each cell in the whole system of 19 cells. The total channel pool denoted by Cpool is $48 \times 19 = 912$ channels.

The Normalized Carried Load ρ is defined as:

$$\rho = \frac{\sum_{j=1}^{N} C(j)}{Cpool} \tag{6.10}$$

Where *C(j)* stands for the channels carrying call *j*, thus the nominator is total number of channels employed by a number of N calls in the network; the denominator *Cpool* is the total channel pool of the whole network with 1/1 frequency reuse.

If ρ=1, it means that all channels in the channel pool of this 1/1 reuse system is full occupied. If ρ=2, it means that channels employed have doubled 1/1 frequency reuse system.

6.2.2.10 Offered Erlangs

In the simulation, in each iteration, we have a total number of *n* subscribers in the network, each starting a call with probability *p*. H denotes the mean call duration, which is exponentially distributed. If *np* is in the order of 1, the process of call arrival is approximately a Poisson process. Calls enter the network with rate: $\lambda = np$.

The Offered Erlangs A in the network is defined as:

$$A = \lambda \cdot H \tag{6.11}$$

6.3 Simulation Part 1—Performance of the Network

The downlink performance of the network under different normalized carried load is evaluated.

Only streaming data users with variable rate and requesting multi-slots are simulated in this part. We consider three classes of users:

User Class1: the average time slots needed by each user is 6—average user data rate is 288kbps since each slot carries 48kbps. The user rate changes uniformly from requesting 4 time slots to requesting 8 time slots.

User Class2: the average data rate is 144kbps and requires 3 time slots. The user rate changes uniformly from requesting 1 time slot to requesting 5 time slots.

User Class3: the average data rate is 144kbps and requires 3 time slots. The user rate changes uniformly from requesting 2 time slots to requesting 4 time slots.

The data rate of an ongoing call varies every 4 multiframe (960 millisecond).

User mobility is considered. Mixed users of different velocity move according to the model described in chapter 5.5. User location and his SINR are updated every 10 multiframe. Inter-cell and intra-cell handoff is considered.

In doing this simulation, we are aimed at investigating the downlink performance of EGPRS network under different buffersize, channel assignment schemes, propagation environments, sidelobe level of antennas, etc.

6.3.1 Simulation Description

The simulation goes as follows:

- Channel allocation: calls arrive at a sector of a cell--either evenly or unevenly distributed in the network, then request for channel allocation. Channels are allocated to users according to the least interference first algorithm.
- User packets generation and service: if users are accepted, they start to generate packets continuously; those packets will be served by the channels assigned in each frame. Every 4 multiframe, the user rate varies uniformly within its variation scope.
- User location and SINR updating, and handoff management: mobile users move from cell to cell, from sector to sector. Their location and SINR are updated every 10 multiframe. Intra-cell and inter-cell handoffs need to be managed.
- Channel release: handoffed or dropped calls release their channels to the channel pool.

Once calls are accepted to the network, they will stay until the end of the simulation unless they are dropped. The simulation length depends on the total iteration number, an input parameter.

Some details of the simulation are given in the following section.

6.3.1.1 Simulation without Multipath Consideration

- *Channel Allocation*

Algorithm Description:

1) A new call arrives to the network and initiate channel assignment request.

2) The BS searches in the channel resources pool of 10944 channels, as shown by channel resource table 6.1.

3) If at some frequencies there are free timeslots that are not yet used in the network,

and the number of free timeslots is larger than or equal to the number of channels requested, sort the carrier that has enough free timeslots available by number of timeslots. Assign the user with the carrier which has smallest number of free timeslots but enough for the new call. The new call is accepted into the network. The channel assignment is finished. Otherwise, go to step 4).

4) Check the channel pool of 48 channels in the current sector. Take all carriers with number of free timeslots larger than or equal to number of channels requested by the new call as candidate carriers. Calculate the user's SINR on each free timeslot of candidate carriers (detailed algorithm is described after in **a.**). If its SINR is larger than SINR requested by the new call, the timeslot is usable. Otherwise, it is unusable. Now candidate carriers become all carriers with number of free **usable** timeslots larger than or equal to number of channels requested by the new call. If number of candidate carriers is zero, the new call is rejected. The channel assignment is finished. Otherwise go to step 5).

5) Calculate SINR of all co-channel users of those usable channels on candidate carriers of the new call(detailed algorithm is described later on in **b.**).

6) Timeslots whose co-channel users' SINR has dropped to less than SINR requirement are unusable. New candidate carriers are selected again according to those free usable timeslots left. If no candidate carrier is left, the call is blocked. Otherwise, go to next step.

7) Sort candidate carriers for new call by SINR and number of timeslots. Choose the Carrier with the best SINR, and with the smallest number of usable timeslots if SINR is the same. The new call is accepted into network. The channel assignment is finished.

Figure 6.1 is the flow diagram of the channel allocation process.

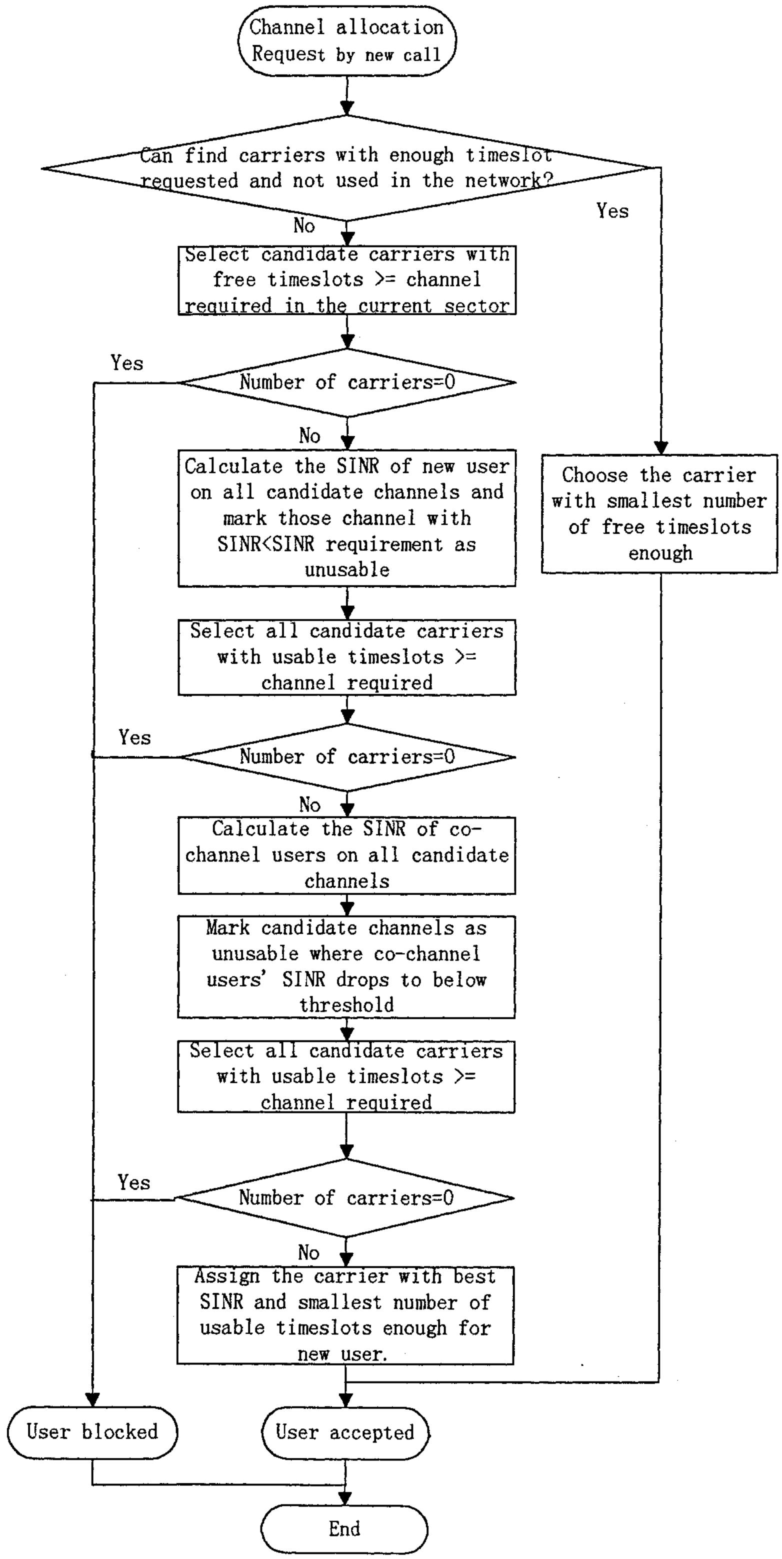


Figure 6.1 Flow Diagram of Channel Allocation

a. SINR on Candidate Channels in Current Sector

After candidate carriers are chosen, SINR of the new call on all free timeslots of candidate carriers needs to be calculated one by one.

Algorithm description:

1) Choose the first one of all eligible channels of a candidate carrier;

2) Select all co-channel users of this channel.

3) Find the cell position of the first co-channel user with regard to current cell where the new call is located, that is, find the relative cell no. of this co-channel user. The distance between the serving base station of this active interfering cell and the new user is calculated accordingly.

4) Find whether or not the sector where the co-channel user resides is the main interfering sector toward the current sector where the new call stays (By checking the interfering sector Table5.3 using the relative cell no.) (please refer to Figure 5.3). Calculate the interference accordingly, using the value of distance got in step 3).

5) Calculate the interference caused by next co-channel user as step 3), 4), and so on, until interference caused by all co-channel users are investigated.

6) Get the interference the new user would suffer by summing interference coming from all co-channel users calculated from step 3) to step 5). And calculate SINR of the new user.

7) Choose the next candidate channel, repeat step 2) to step 6). And then the next, until SINR on all candidate channels are calculated.

Calculation of a new call's SINR on one candidate channel is as shown by the flow diagram in Figure 6.2.

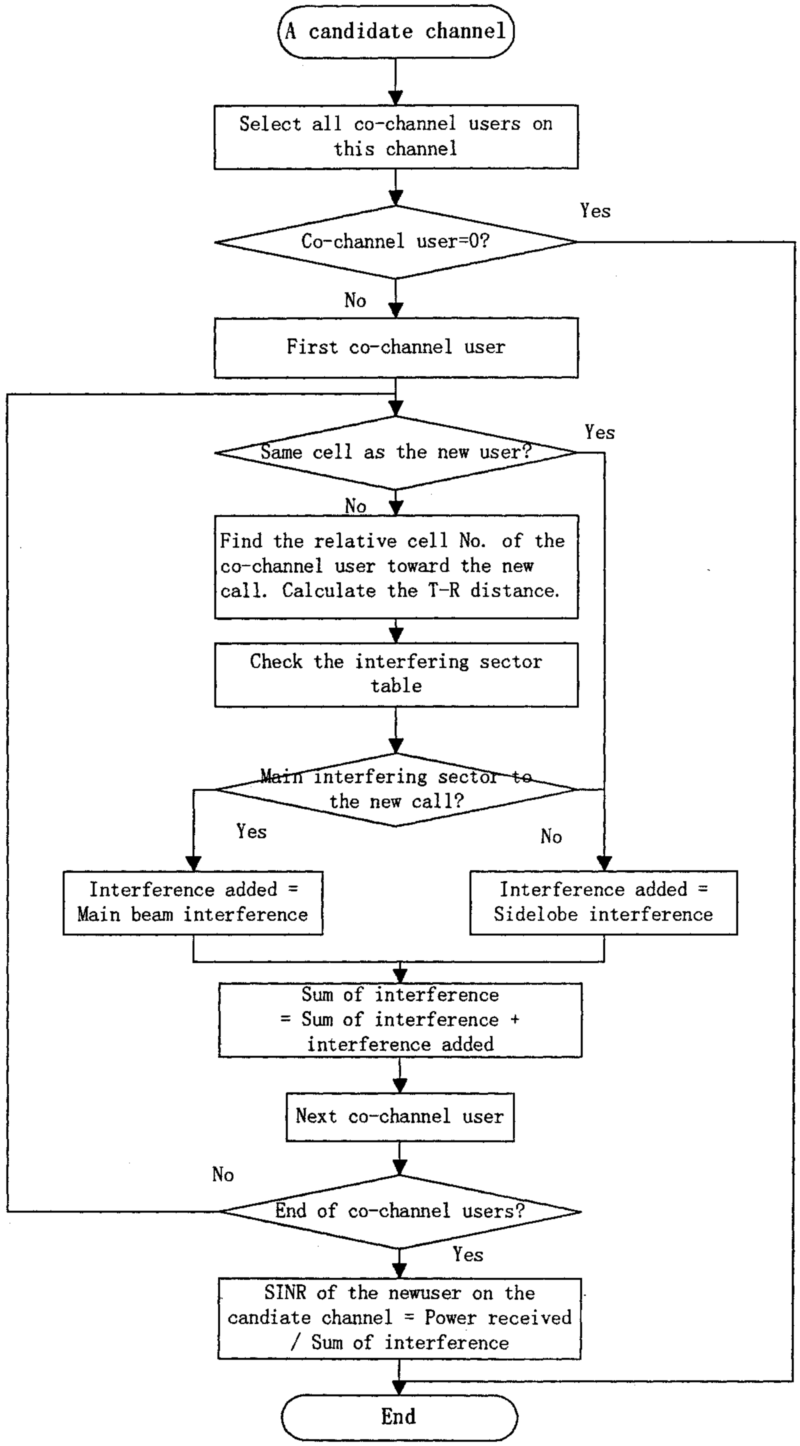


Figure 6.2 Flow Diagram of a New Call SINR Calculation

b. New SINR of Co-channel Users

The SINR of all co-channel users of all candidate channels need to be calculated, with the purpose of finding for new calls the best SINR channels that are compatible with the on-going calls.

Algorithm description:

1) Select all candidate channels.

2) Choose the first candidate channel.

3) Select all co-channel users of this channel.

4) Find the cell position of the new call relative to the cell of the first co-channel user, that is, find the relative cell No. of the new call. The distance between the serving base station of the new user and the co-channel user is calculated.

5) Find whether or not the sector where the new call resides is the main interfering sector toward the sector of the co-channel user. Calculate the interference accordingly, using the value of distance got from step 4), and this new interference is added to the already existed interference the co-channel users already suffered previously.

6) Calculate the new SINR of this co-channel user.

7) Go to next co-channel user, repeat step 4) to 6), and so on, until all users are investigated for the new calls.

8) Go to next candidate channel, repeat step 3) to step 7), and so forth, until the end of all candidate channels.

The flow diagram of new SINR calculation of co-channel users of one candidate channel is as shown in Figure 6.3.

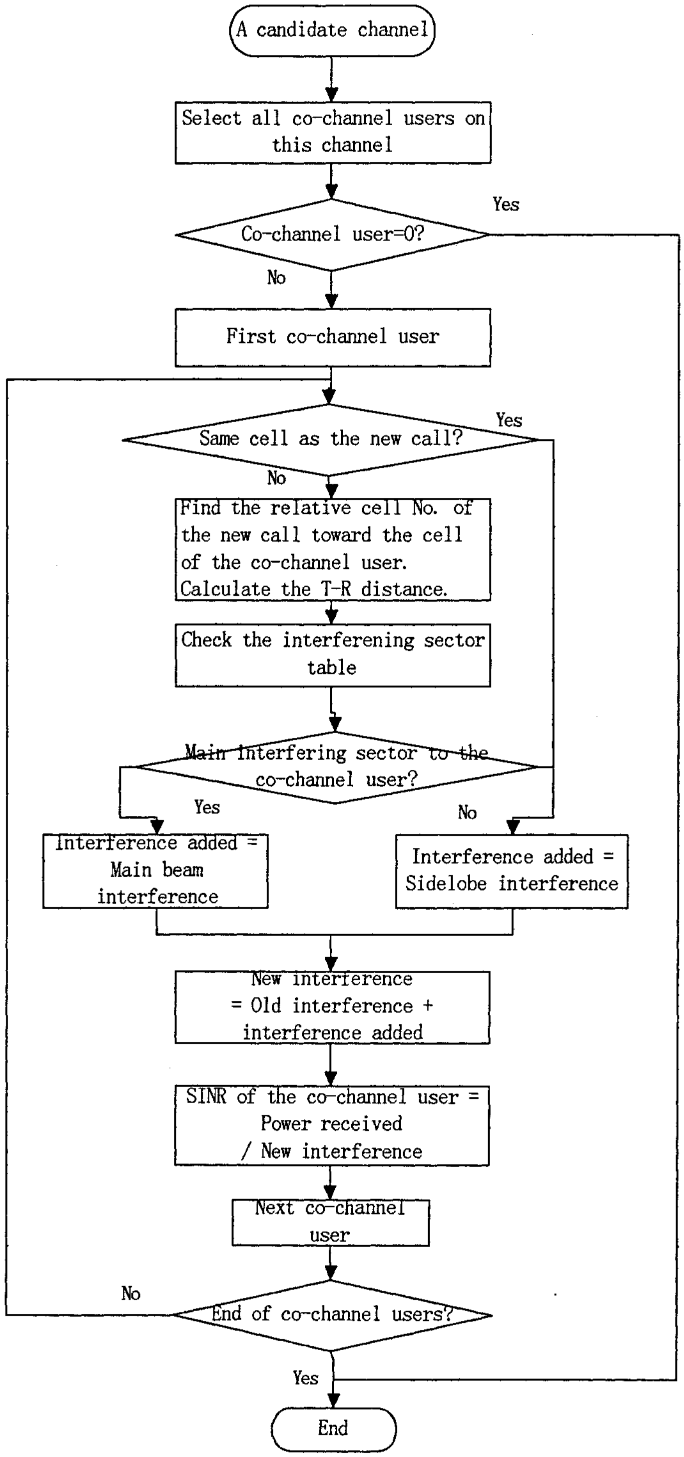


Figure 6.3 Flow Diagram of Co-channel User SINR Calculation

- ***Users Location and SINR Updating***

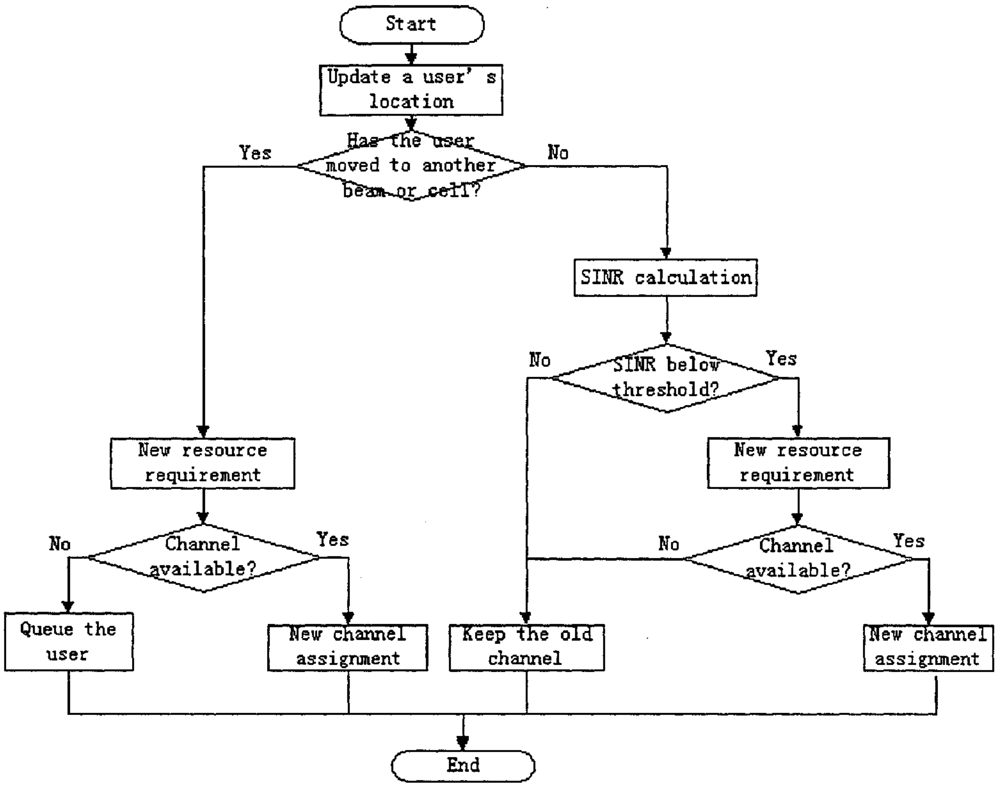


Figure 6.4 Mobile Users Location Updating

Mobile users will move with different speeds and in different directions. Location updating is done every 10 multiframes. After location updating, some users may move to another cell, and request new channel from that cell. If there are channels available, inter-cell handover is accomplished, otherwise, the roaming calls are queued. If after 10 multiframes, a roaming call still can not find channel needed, this call will be dropped. The same algorithm applies to users who have moved to another sector within the same cell. Users either switch to other channels, in which case, the intra-cell handover occurs, or will be queued when no channels are available.

Some calls may suffer from low SINR after location updating, therefore, they will also request new channels. If new channels are found, they transfer to these channels, otherwise they will keep using what they have. If a call who suffers from low SINR cannot find new channel(s) after 20 multiframes, this call will be dropped.

SINR of All users' is updated according to their new location.

The flow diagram of location and SINR updating is as shown by Figure 6.4.

- ***Channel release***

If an on-going call has moved to another cell or sector or it has been dropped, the channels used are released to the channel pool.

When a new call comes to the network, the interference is added to all co-channel users. Similarly, when channels are released, the interference the call had on co-channel users need to be deducted. SINR of all co-channel users is recalculated.

- ***Packets Generation and Service***

Once a call is accepted, it starts to generate packets continuously. The mode the packets generated depends on the user class. In each frame, the packets generated will be served if there are channels available. Those packets that cannot get served will be queued in the buffer, resulting in queuing delay. A buffer is overflowed when the packets in it exceed its size. Extra packets are lost.

6.3.1.2 Simulation with Multipath Consideration

In the channel assignment used above, since line of sight is assumed, mobile users can move to the edge of a sector then start to handoff without interfering with adjacent sectors.

Taking multipath into consideration, we only apply the angular constraint, as discussed in Chapter 4.3.1, in channel allocation, since we deal with only downlink.

Two channel allocation schemes are simulated.

1) Timid Allocation

To avoid beam collision and reduce possible interference, a mobile call will not be assigned the channels that are already used by the adjacent sectors.

2) Aggressive Allocation

In this case, when a call requests a channel, it can be assigned all eligible channels (those which meet the SINR requirement) with prerequisite that the angular constraint has to be satisfied. If a call is so close to the base station that the spread angle of its scatterers is larger than the beam width, the same channels that have already been used in both of the adjacent sectors cannot be used. If it is so close to one adjacent sector that its scatterers collide only with this adjacent sector, the co-channels already used by this adjacent sector cannot be assigned.

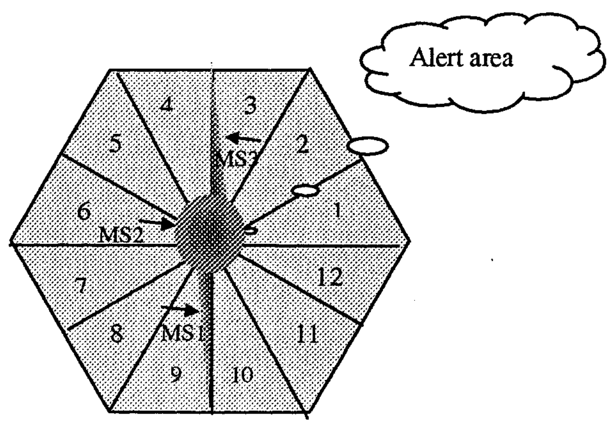


Figure 6.5 Illustration of "Alert" Areas

The channel allocation algorithm is similar to what has been described in 1 of chapter 6.3.1.1, except that step 4) is modified as: take all frequencies with free timeslots larger than or equal to channels requested by the new call. Check if the new call's scatterers collide with the adjacent sector(s), the co-channels of those channels that are already used in the adjacent sector(s) cannot be used. The rest parts are the same.

Therefore, in location updating, after a roaming user's location has been updated, he needs to check if he has move to an "alert" area, which means that the scatterers collide with adjacent sector(s) or angular constraint is no longer satisfied. Within these areas, he cannot use the co-channels of the collided sector. If he does, he needs to handoff to another channel. The "alert" areas are as shown by shaded regions in Figure 6.5. For example, if MS1 in sector 9 has moved to the shaded area, he cannot use channels that are already used by users in sector 10. Similarly, MS3 in shaded area of sector 3 cannot use the same channels as users in sector 4. Please notice that if MS2 in sector 6 has moved to the shaded area in the center of the cell, he cannot use channels that are occupied by users in sector 5 and sector 7.

The 'alert area' is calculated according to angular constraints, which, in our case, depend on the distance of the mobile to its serving base station.

6.3.1.3 Static vs. Adaptive Channel Assignment

Using the complete dynamic channel allocation algorithm described, a new call or a handoff call is served by the network if the resource is available. The problem is that the user rate is variable. So different channel assignment schemes will influence a lot the final performance.

In the simulations, we have tried two types of channel assignment schemes.

1) Static Channel Assignment

With this scheme, the channel allocated to a user is fixed during the call, although the user rate varies. Since the user data rate is assumed to vary from minimum rate (minrate) to maximum rate(maxrate) uniformly, two static assignment schemes are simulated:

Static assignment 1: The assigned channel =minrate+ (maxrate-minrate)$\times$ 1/2

Static assignment 2: The assigned channel =minrate+ (maxrate-minrate)$\times$ 3/4

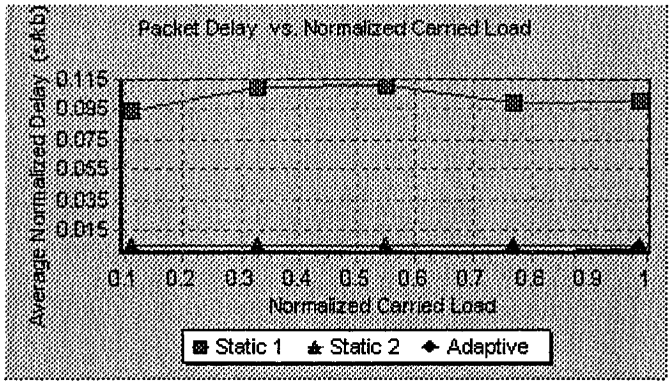
2) Adaptive Channel Assignment

With adaptive channel assignment, the assigned channel changes according to the user rate adaptively. If there are not enough channels for an on-going call in the network, the call is assigned with what is available for it. The extra packets generated by the call will be put into buffer. In the following iteration, the network will assign him more slots than needed, so that the packets in buffer will be sent out. So there will be no dropped calls with adaptive channel assignment.

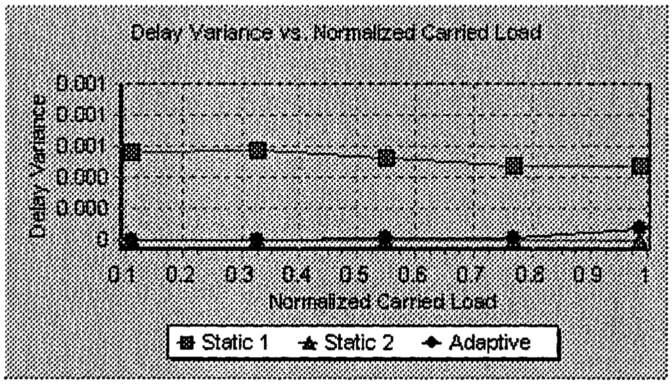
6.3.2 Simulation Results and Discussions

6.3.2.1 Performance with Different Channel Assignment Schemes

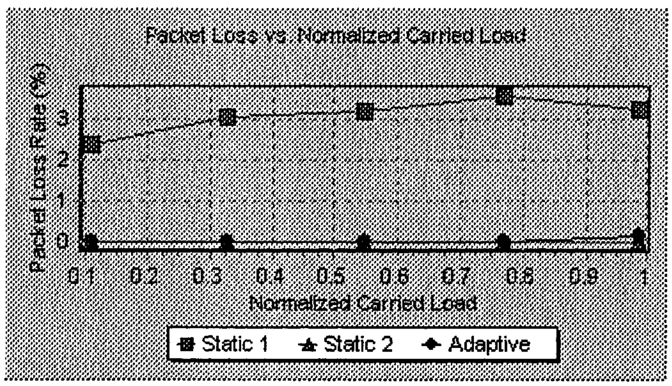
We simulated two kinds of channel assignment schemes: static channel assignment 1 and 2, as well as one adaptive channel assignment scheme, as already described . The simulation results are shown by Figure 6.6 a) to g).


Packet Delay vs. Normalized Carried Load
Average Normalized Delay (s/kb)
0.115
0.095
0.075
0.055
0.035
0.015
0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9 1
Normalized Carried Load
Static 1
Static 2
Adaptive

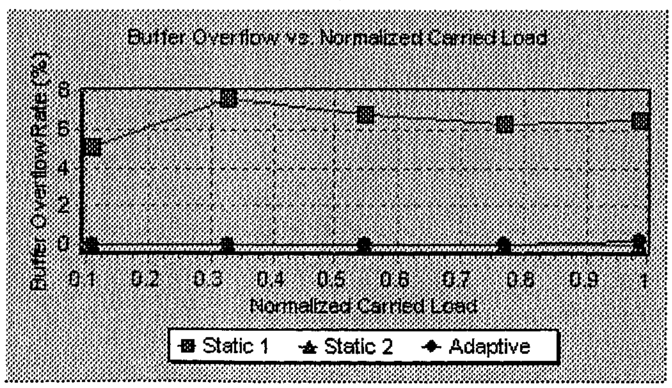
a)


Delay Variance vs. Normalized Carried Load
Delay Variance
0.001
0.001
0.001
0.000
0.000
0
0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9 1
Normalized Carried Load
Static 1
Static 2
Adaptive

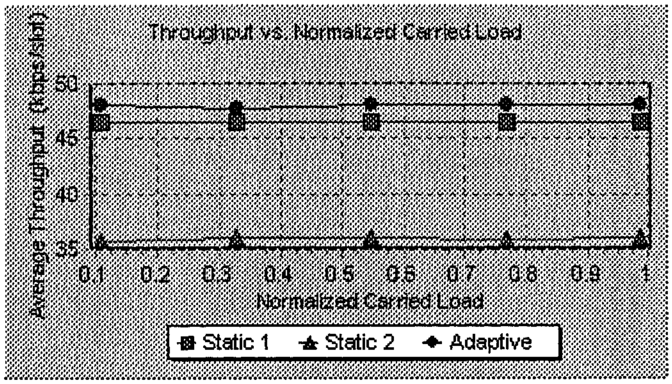
b)


Packet Loss vs. Normalized Carried Load
Packet Loss Rate (%)
3
2
1
0
0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9 1
Normalized Carried Load
Static 1
Static 2
Adaptive

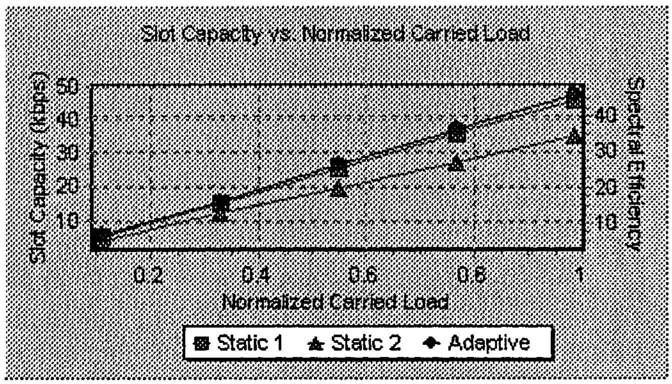
c)


Buffer Overflow vs. Normalized Carried Load
Buffer Overflow Rate (%)
8
6
4
2
0
0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9 1
Normalized Carried Load
Static 1
Static 2
Adaptive

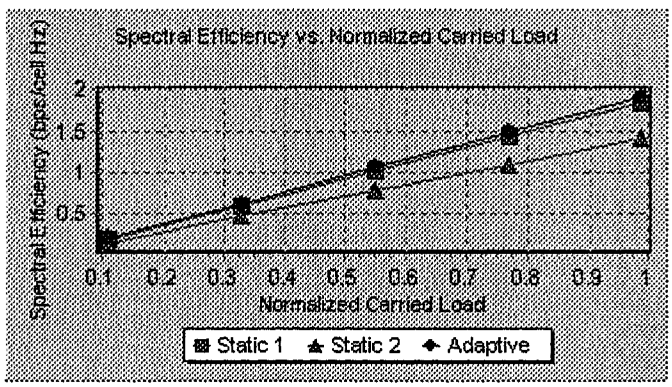
d)



e)



f)



g)

Figure 6.6 Simulation Results with Different Channel Assignment Schemes
(SLL=40dB; n=3; Buffersize=345.6kb; User Class=2; SINR Required=12)

We can see from the figures that, in comparison with static channel assignment schemes, the adaptive one achieves much better overall performance, with very little packet loss and delay, high average throughput and spectral efficiency.

Between two static schemes, the static scheme 1 have larger packet loss rate and longer delay, but much higher spectral efficiency. Because the user data rate is uniformly changed around the average rate, the average throughput can be achieved is very high with this simple allocation scheme. But if the user rate doesn't vary uniformly, the result will be different.

Though scheme 2 has no packet loss and very small delay due to generous resources allocation, a lot of resources are wasted, thus resulting in low average throughput and spectral efficiency.

Note that for the same scheme, the performance varies with different carried load. Some seemingly unreasonable fluctuations may be explained by the random nature of users' mobility and data rates. This phenomenon may also happen in the following simulations.

6.3.2.2 Static Assignment with Different Parameters

In this chapter, we only use static channel assignment scheme 1 and check how other parameters affect the performance of the network. Remember that the assigned channels equal to the average data rate of users.

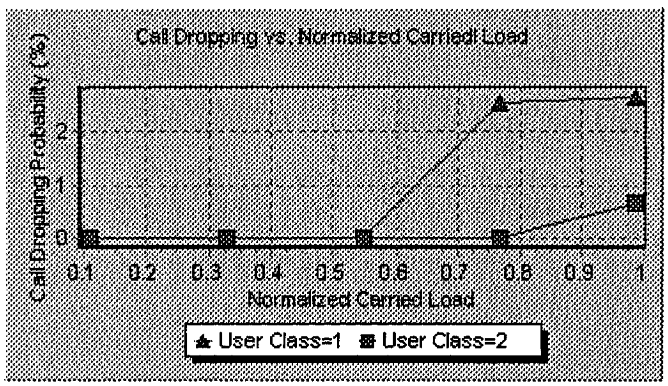
1) Different User Class

- User Class 1 and User Class 2

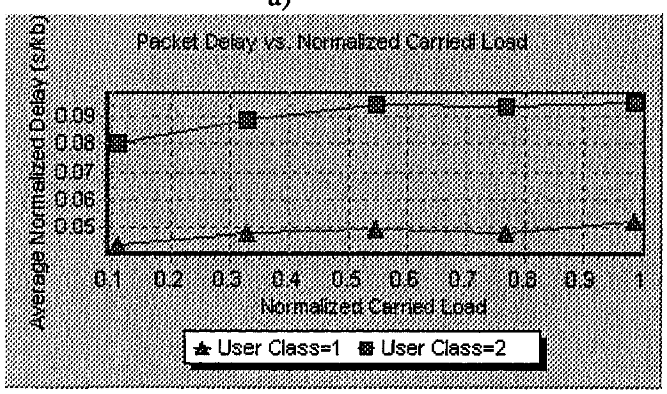
User Class 1 and 2 have different average rate, requesting average 6 or 3 timeslots respectively, but have the same scope of variation, that is, user rate varies with ± 2 timeslots.

The result of Figure 6.7 a) shows that user class 1, which requires more slots than user class 2 tends to have higher dropping rate. This is due to that high rate users are not only interference limited, but also block limited, which means due to insufficient resources. The trunking efficiency of the network is actually decreased by multislot request.

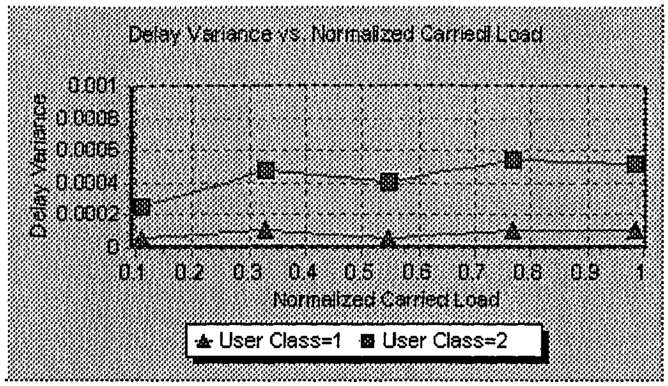
Figure b) and c) tell us that class 1 has lower delay and delay variance. Because the data rate of class1 users has the same scope of variation as that of class 2, but the average rate is much faster, the same number of packets in the buffer will cause less delay. For the similar reason, class 1 has lower packet loss rate and higher throughput.



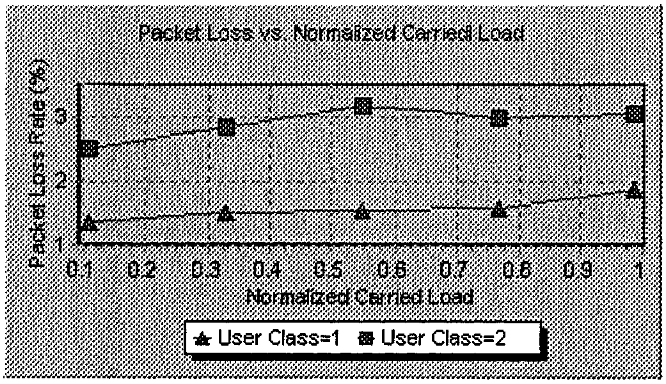
a)



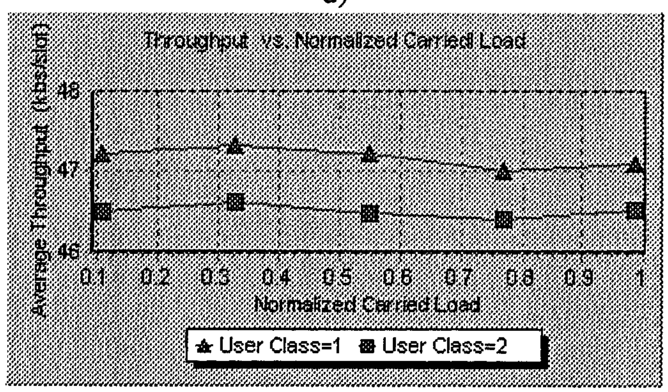
b)



c)



d)



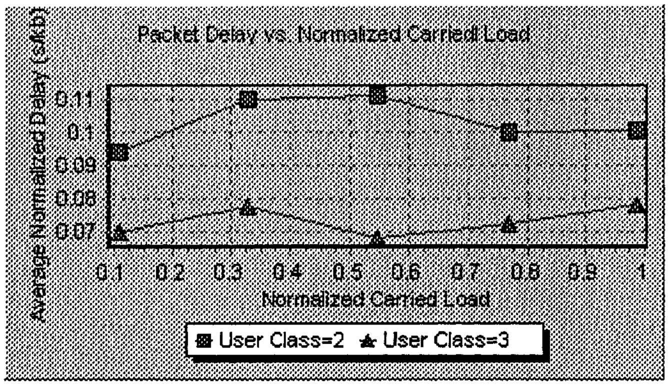
e)

Figure 6.7 Simulation Results with Different User Class (1)
(SLL=20dB; n=3; Buffersize=345.6kb; SINR Required=12dB)

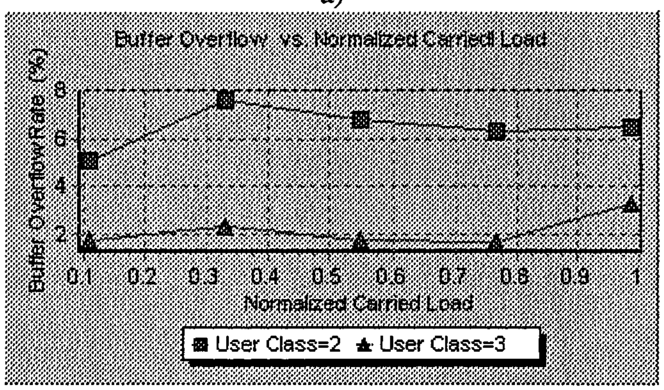
- User Class 2 and User Class 3

These two classes of users have the same average rate (each needs average 3 slots), but different scope of variation. Class 2 user rate varies with ± 2 slots, whereas class 3 user rate varies with ± 1 slot.

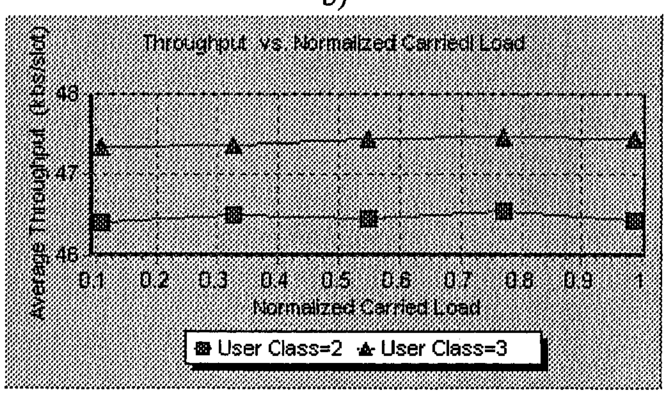
The conclusion is evident from Figure 6.7 a) to d). Users with less rate variation will have less delay, less packet loss, less buffer overflow and higher throughput in case that the assigned channels are fixed during the call.



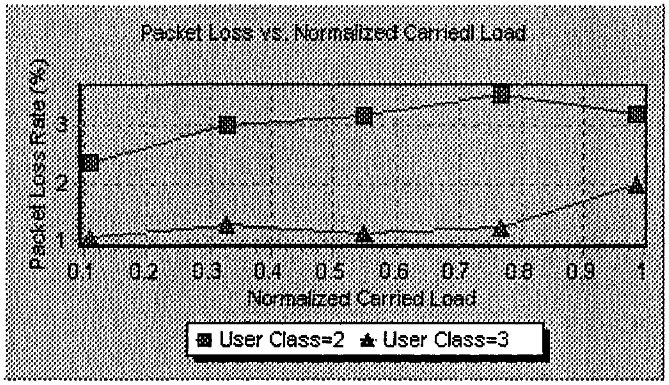
a)



b)



c)



d)

Figure 6.8 Simulation Results with Different User Class (2)
(SLL=40dB; n=3; Buffersize=345.6kb; SINR Required=12dB)

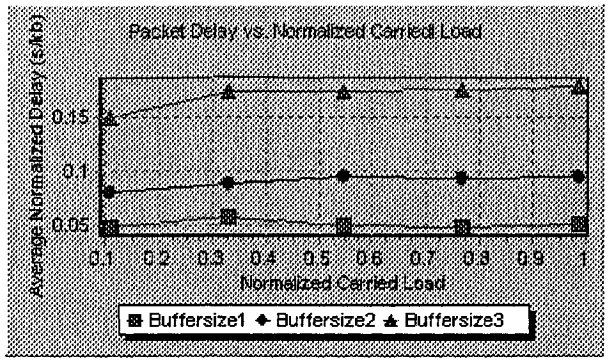
2) The Same User Class

In this section, we want to find out, for the same class of users, how other parameters influence the result. Only user class 2 is simulated.

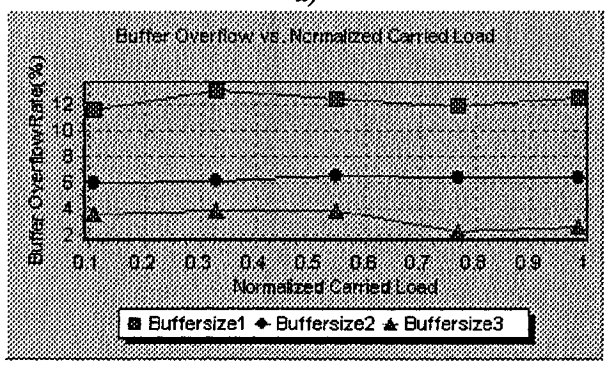
➢ Buffer size

The buffersize1,2,3 we use here are equivalent to packets that can be transmitted over 15, 30, 50 multiframes. Because with 48kbps data rate per slot, a multiframe (240 ms) per slot can transmit $48 \times 0.24 = 11.50$kb. For class 2 users with average rate 3 slots, it will take respectively 1.2, 2.4, 4 seconds to transmit all packets in buffer when the buffer is full.

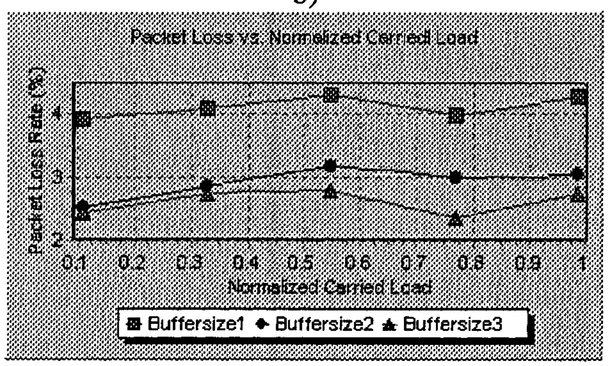
Figure 6.9 a) to d) shows that different buffer size results in different performance. For the same class of users, with small buffersize, we have less delay, but more buffer overflow, hence packet loss, resulting in low throughput.



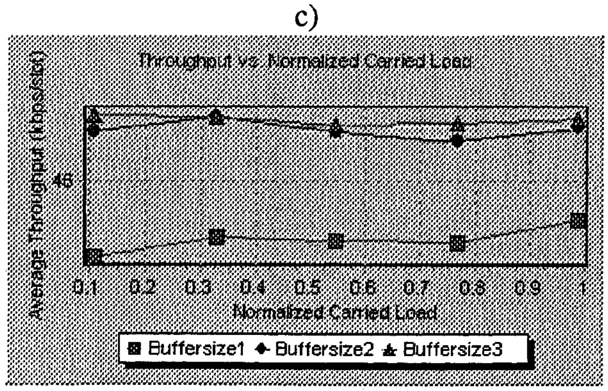
a)



b)



c)



d)

Figure 6.9 Simulation Results with Different Buffersize
(SLL=20dB; n=3; SINR Required=12dB)

➢ Sidelobe Level (SLL) and SINR Requirement

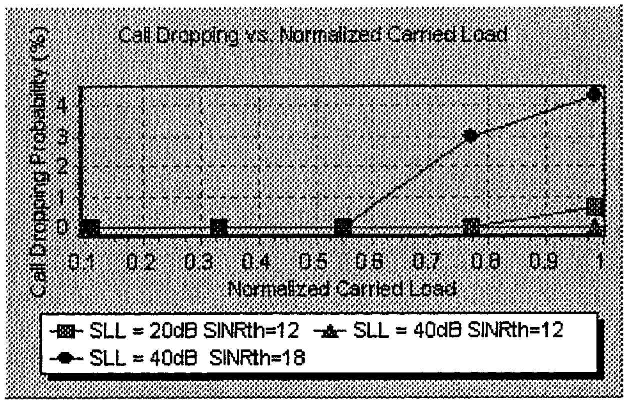


Figure 6.10 Simulation Results with Different SLL and SINR Requirements
(n=3; Buffersize=345.6kb)

Sidelobe level of antennas affects much the blocking and dropping probability of the system. Low SLL is required especially in SDMA system.

With other parameters fixed, SINR requirement by users makes difference in performance, as shown by Figure 6.10.

➢ Distribution of Users

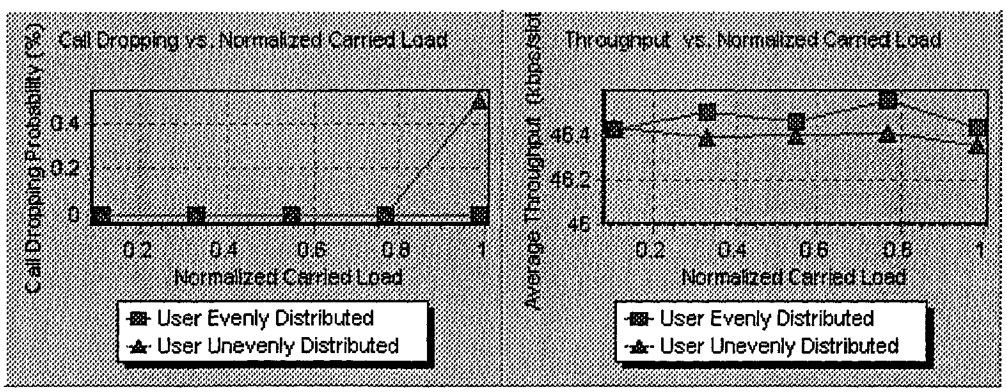
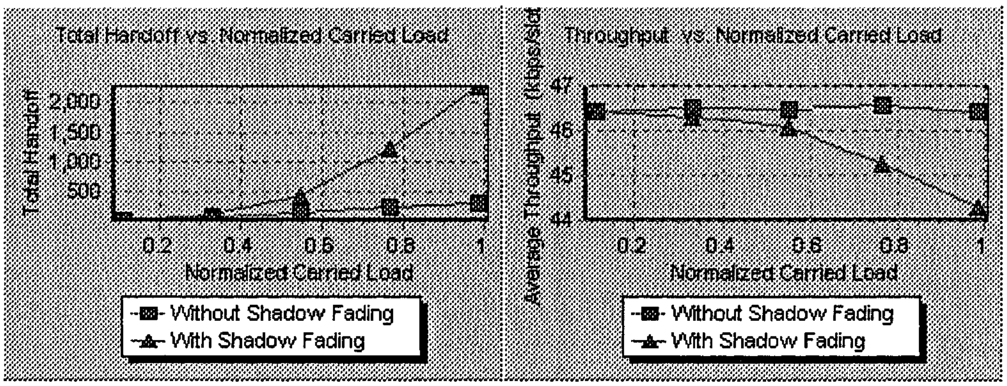


Figure 6.11 Simulation Results with Different Distribution of Users
(SLL=40dB; n=3; Buffersize=345.6kb; SINR Required=12dB)

The distribution of users is a key factor to be considered when choosing the channel allocation algorithm and employing SDMA. The output depends on how much mobile users are clustered. In our case, we simulated the uneven distribution of users with 70% probability to generate calls at cell 2 to cell 7, whereas 30% at cell 8 to cell 19. This leads to higher call dropping probability and slightly lower throughput comparing with the case of even distribution.

➢ Shadow fading



a) b)

Figure 6.12 Simulation Results with or without Shadow Fading
(SLL=40dB; n=3; Buffersize=345.6kb; SINR Required=12dB)

Shadow fading tends to have negative influence on the network, as can be seen from Figure 6.12. A much larger number of intra-cell handoff, more than 2000 within 500 iterations, happens since the SINR of the on-going call fluctuates to below threshold causing by shadow fading effect. And throughput rate carried by each slot goes down.

➢ With Multipath Considered

From the simulation result (Figure 6.13), the influence of multipath is not so obvious, though the consideration of multipath brings us much more trouble in channel allocation.

This may be due in part to the uncertainty of the calls, which are randomly generated, as we have explained earlier.

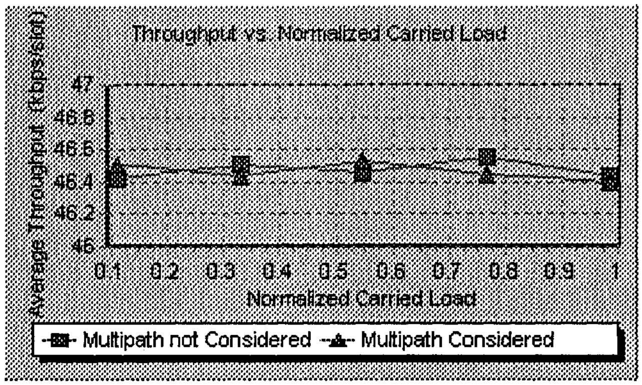


Figure 6.13 Simulation Results with or without Multipath Considered
(SLL=40dB; n=3; Buffersize=345.6kb; SINR Required=12dB)

6.4 Simulation Part 2—EGPRS vs. Conventional GSM

In this part, we want to evaluate the capacity improvement of EGPRS comparing with conventional GSM under SDMA.

Only users requesting less than one slot are considered so that multi-users can share one slot. EGPRS can accommodate maximum 12 users in one slot (1/12 rate) in one multiframe (suppose that USF bits can be extended for 12 users), whereas conventional full-rate GSM can accommodate only one user in one slot. We simulate both 1/6 rate users, which means that 6 users can share one slot, as well as 1/12 rate users.

Each time unit, a number of n subscribers in the network initiate calls with probability p. The different values of p give us different offered traffic intensities. The call duration is exponentially distributed with mean H=20 time units. One time unit equals to one multiframe, to shorten the simulation time.

User mobility is not considered, and the user rate is assumed fixed during the call due to relatively short dwell time in the network.

Call Blocking Probability at different offered traffic intensity (offered Erlangs) is calculated and compared.

6.4.1 Simulation Description

The simulation goes as follows: 1. Channel allocation: users are uniformly distributed in the network. Calls arrive at the network according to an approximate Poisson process and request channel allocation. The channels are allocated to users according to the least interference first algorithm. Calls stay with their channels till the end of their call duration. 2. Channel release: the call duration is exponentially distributed. Once users finish their call, they will leave the network and channels employed are released to the channel pool.

Since multiuser can share one slot, the process of channel allocation and release is different from that described in chapter 6.3.1.1..

Some details of the simulation are given in the following section.

- *Channel Allocation*

Algorithm Description:

The simplified flow diagram is shown by Figure 6.14.

1) When a call arrives, we search in the channel pool to find whether there is eligible channel with enough space to accommodate this call for him, if not, the new call will be

blocked.

2) If there exists candidate channels, we first look for channels (timeslots) which are already used by other users in the current sector but still not used else where in the network, and then channels not used in the whole network. As on these channels, there is no interference from others users. Among these channels, we will choose the one with the smallest space left but still enough to accommodate the new call. In this way, we save more channel and impose less interference on the network..

3) If the above-mentioned channels do not exist, we go to find the channels that are used by on-going calls in current sector as well as in other sectors, but have enough room for the new call. The new user's SINR need to be calculated, but other co-channel users' SINR remain unchanged, as it is reasonable to assume the new call added to a used channel will not add extra interference to other co-channel users. The slot with the best SINR, and with smallest room left but enough to accommodate the new user will be chosen.

3) The last choice is to take a channel which is free in this sector, but already used in other sectors. The SINR of the new call on all these candidate free channels will be calculated (similar to described in 6.3.1.1) in order to find the one offering the best SINR. Once the candidate channel with the best SINR is selected, the interference added to other co-channel users are calculated (see Figure 6.15). If all other co-channel users' recalculated SINRs are above threshold, this candidate channel is assigned to the new call. Otherwise, we go to the channel with second best SINR, and continue the co-channel users SINR recalculation as above. If till the end of candidate channel list, no channel can be selected, the new call will be blocked.

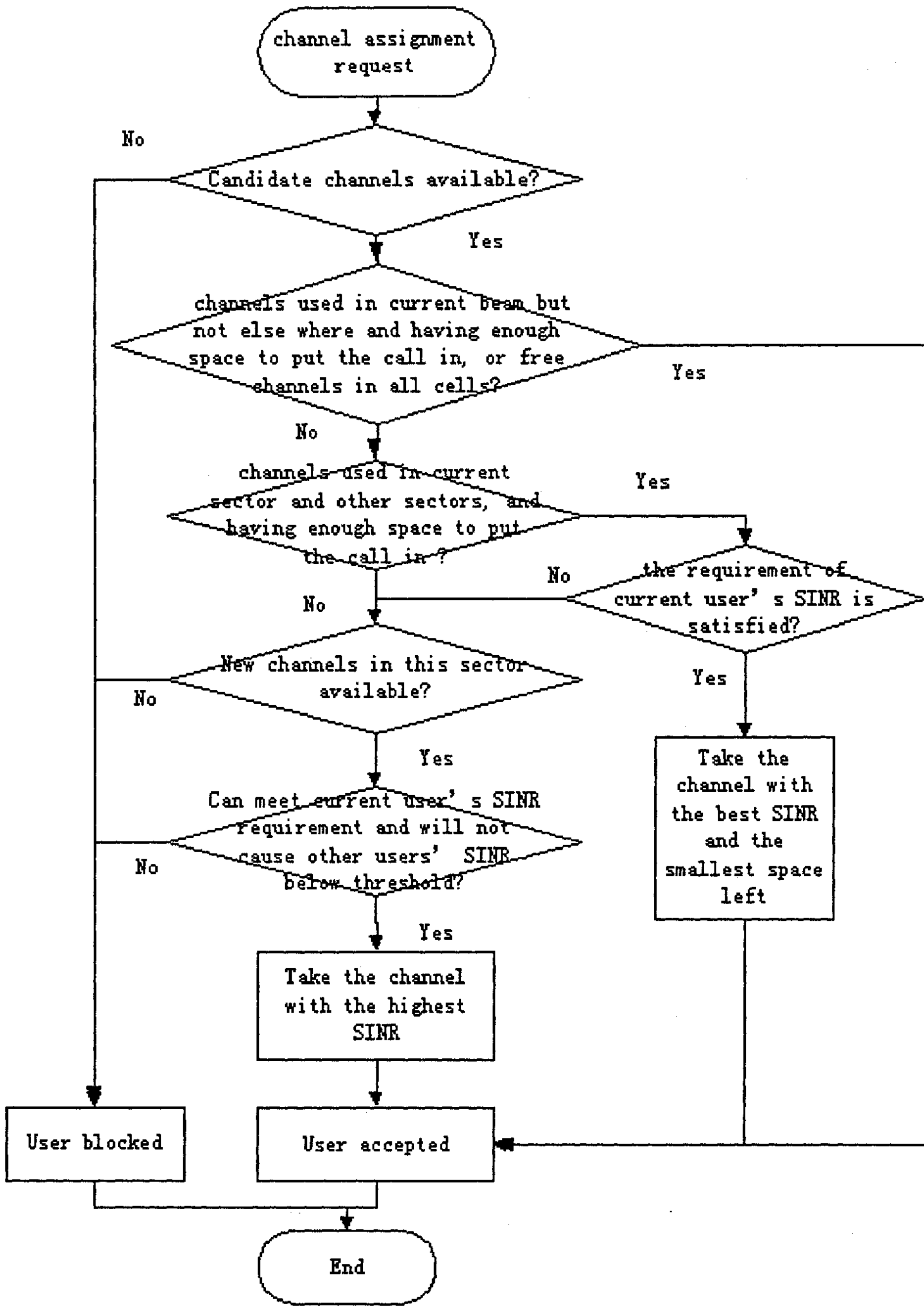


Figure 6.14 Channel Allocation for Small Rate Users

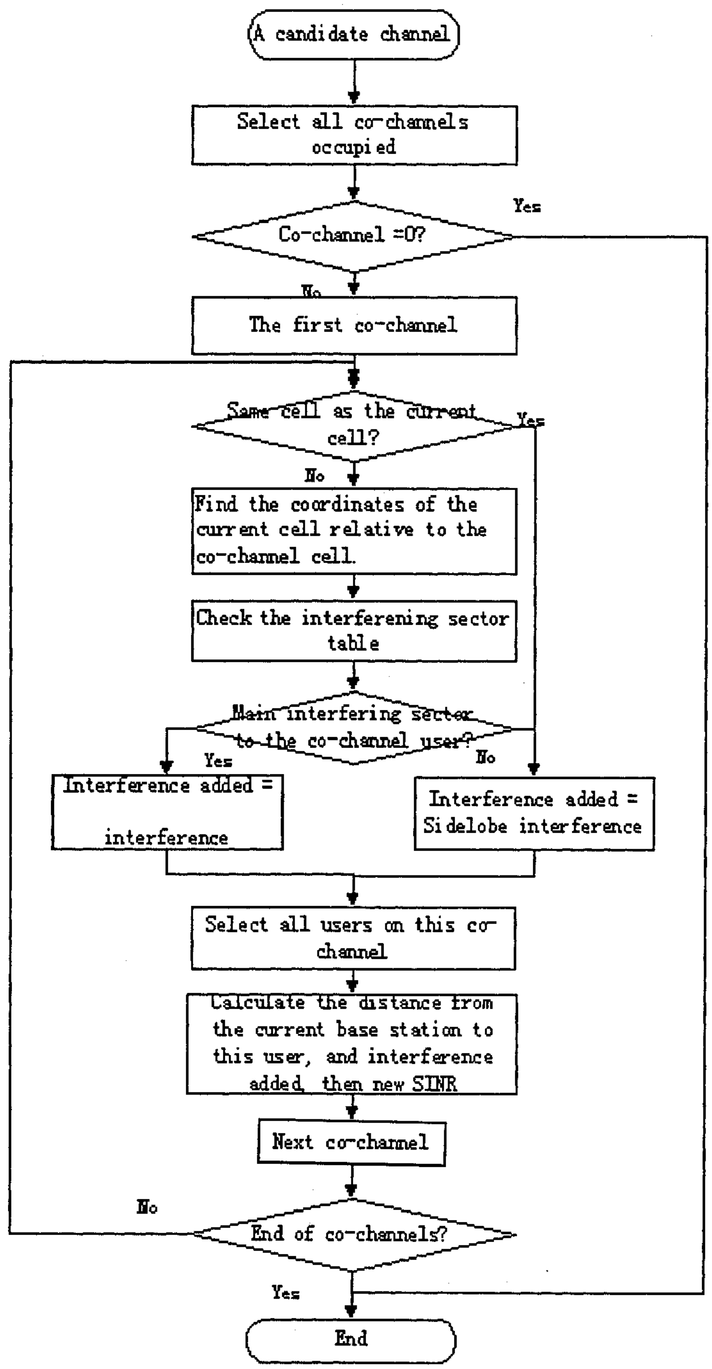


Figure 6.15 Flow Diagram of SINR Calculation of Co-channel Users

- ***Channel Release***

After a user finishes its call, the channel used will be returned to the channel pool. In case that multiuser share one slot, the finished call cannot release the whole channel but only a part of channel. If a whole channel becomes totally free after being released, its co-channel interference on other users needs to be released, i.e. other co-channel users' SINRs need to be recalculated. Otherwise, SINR recalculation is not needed. The process of channel release is illustrated by Figure 6.16.

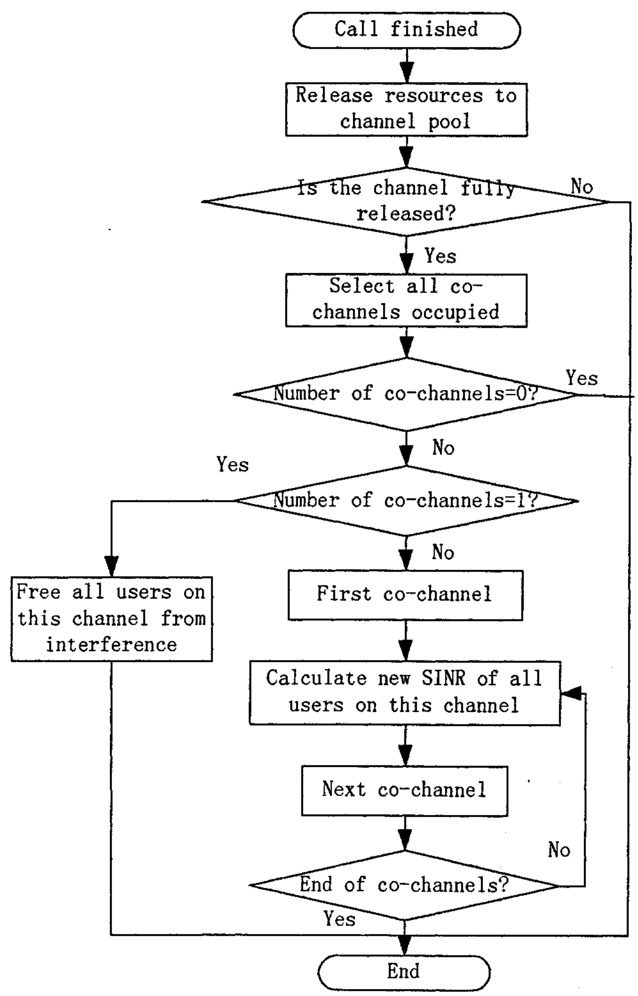


Figure 6.16 Flow Diagram of Channel Release for Small Rate Users

6.4.2 Simulation Results and Discussions

In our simulation, with 1/6 rate users in the network, the capacity of EGPRS at 2% blocking rate is about 270 offered Erlangs, more than 5 times that of GSM. Theoretically, this figure should be less than 6 times, because of the interference of users imposed on the network.

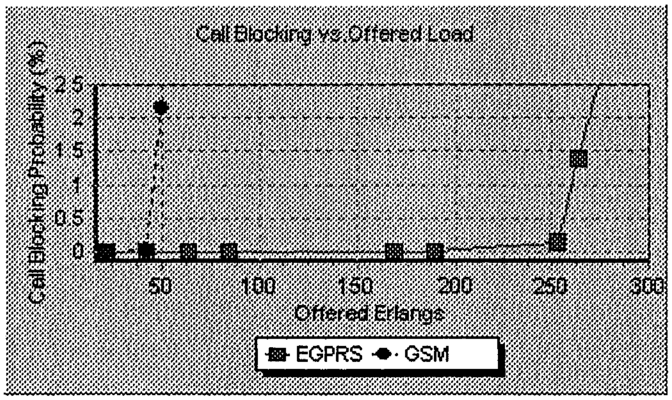


Figure 6.17 Simulation Results with EGPRS and GSM
(SLL=30dB; n=3; SINR Required=12dB)

Small rate users lead to significantly higher numbers of users accommodated by the network at certain Erlangs, as shown in Figure 6.18. The different SINR requirements also make big difference in the final result.

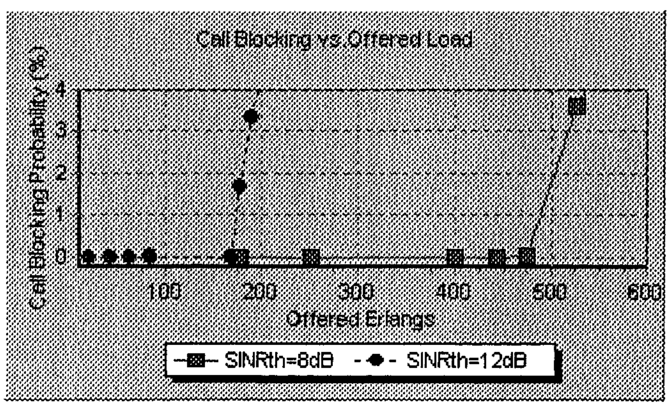


Figure 6.18 Simulation Results with EGPRS
(SLL=20dB; n=3; Userclass=1/12 Rate)

6.5 Simulation Part 3—Capacity of the System

We are eager to find out how much traffic our simulated network- EGPRS with SDMA and DCA, is able to carry simultaneously, as well as the impact of various conditions will have on the result. The blocking probability to reach certain carried load is used to illustrate the capacity of the system.

6.5.1 Simulation Description

We keep adding users to the network. Channel allocation methods described previously in this chapter are used respectively to assign channel(s) to users requiring multi-slots or users requiring less than one slot.

The more users are added, the SINR of each call in the network is getting lower. When the number of calls reaches something like a 'saturation' point at which the average SINR of all calls becomes so marginal, the network could hardly accept any more calls.

We only compare the capacity with call blocking probability less than 2.5%.

6.5.2 Simulation Result

1. Without Multipath

➢ Influence of User Distribution

As already mentioned, 'user evenly distributed' means that users are uniformly distributed in all cells, while unevenly distributed means that users are more likely (70% probability) to appear in the inner cells--from cell 2 to cell 7. The result tells us, as expected, that the distribution of users has impact on the system capacity. The clustering of users leads to high blocking probability, as can be seen from Figure 6.19.

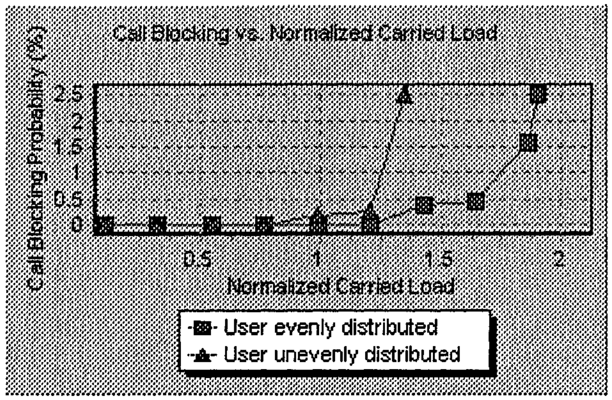


Figure 6.19 Simulation Result of Different User Distribution
(n=3; SLL=40dB; SINR Required=12dB; User Class=2)

> The influence of Propagation Environment and SINR Requirement

From the Figure 6.20, we can easily get the conclusion that the propagation environment (with different n) and SINR value required by the users are quite influential factors on the system capacity.

It is encouraging to find that, in the most lenient case (with n=4 and SINR=8dB), the carried load is about 4 at 2% blocking probability. This means that the number of channels we can simultaneously employ is three times the channel pool of the 1/1 reuse system. Obviously, the number of channels we can simultaneously employ, or the number of users the network can simultaneously accommodate also depends on the user rate or user class.

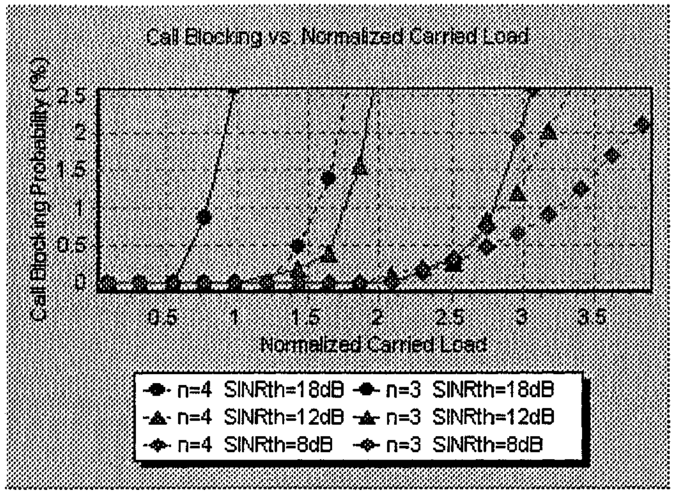


Figure 6.20 Simulation Result of Different Pass Loss and SINR Requirement
(SLL=40dB; User Class=2)

➢ The influence of SLL

The influence of SLL is obvious, as shown in Figure 6.21. The SLL design is especially important in SDMA system. If SLL is large, the network cannot accommodate many in-cell calls even they are spatially separated enough, because the sidelobe interference will be large.

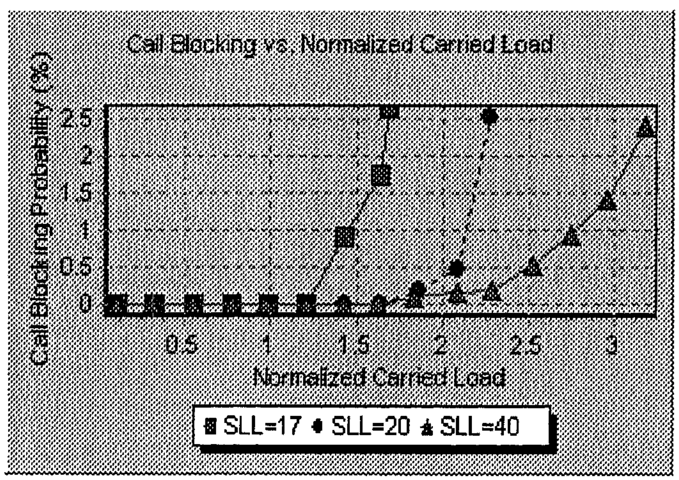


Figure 6.21 Simulation Result of Different SLL
(n=4; SIR=12dB; User Class=2)

> The Influence of User Rate

In this simulation, we have class 1 users requesting average 6 slots per call; class 2 users requesting 3 average slots per call, and 1/12 rate users-- 12 users can share one slot. It is not strange that, at a specific blocking rate, 1/12 rate users have lower carried load than others. Because, for the same carried load, the network carries much more 1/12 rate users. For example, at carried load 0.9, we have 150 class 1 users online, 300 class 2 users online, and more than 9000 users of 1/12 rate online.

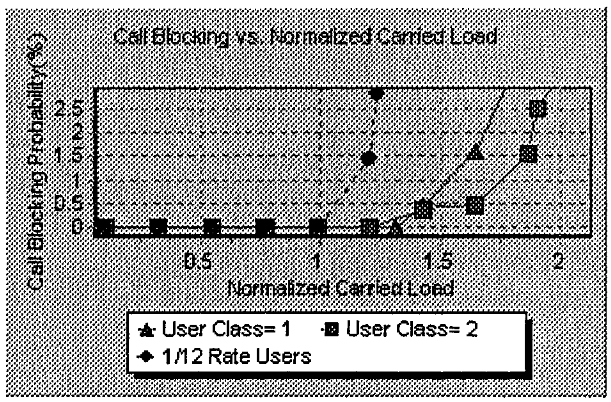


Figure 6.22 Simulation Result of Different User Rates
(n=3; SLL=40dB; SINR Required=12dB)

But when the user rate is too high, as in the case of user class1 requiring 6 slots, the capacity of the system is not only limited by the co-channel interference, but also by insufficient resources. That is why user class 1 has higher blocking probability than user class 2 at the same carried load.

> The Influence of Multipath

As illustrated by Figure 6.23, timid channel allocation leads to obviously low capacity in the system because the users cannot use channels occupied in the adjacent sectors. This decreases the number of available channels.

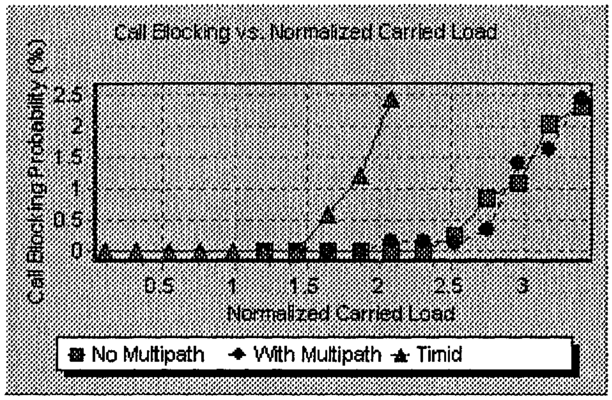


Figure 6.23 Simulation Result of Different Channel Allocation Consideration
(n=4; SLL=40dB; SINR Required=12dB User Class=2)

Channel allocation with consideration of multipath achieves similar result with that without multipath. Because only downlink performance is evaluated, multipath doesn't not obviously decrease the system capacity, as we expected.

Chapter 7

Conclusion and Future Work

7.1 Conclusion

The evaluation of the performance is conducted using simulation models with purpose of representing a real world cellular system. The users are modeled as a mixture of high speed, pedestrian, stationary users requiring streaming variable data services.

Through simulation studies, we show that the performance of the EGPRS network employing SDMA has much to do with different channel assignment schemes, antenna parameters, and the size of buffer offered for users. Adaptive channel assignment achieves much better overall performance comparing with static assignment schemes when user rate varies, at the cost of more intensive computation. When using static assignment schemes, the size of buffer needs to be carefully considered, as there is a trade-off between the user packet loss rate and the packet delay. One thing needs to point out here is that, the simulation results we provided, such as delay, buffer overflow, packet loss, are conservative, because the variation scope of the user rate we simulated is very large.

Reasonably, in SDMA system, the sidelobe level (SLL) of antennas is a very important parameter. If the SLL is high, the sidelobe interference is high, we cannot benefit as much from SDMA (as shown by Figure 6.20). Of course, SINR requirement of users, propagation environment, and the user class are also very important parameters in analyzing system performance.

We also compare the efficiency of EGPRS in case of more than one users sharing one slot with conventional GSM, which accommodate one user per slot. The difference of capacity between them depends much on the user rate. In case of 1/6 rate user, the number of users carried by EGPRS is above 5 times the figure carried by GSM.

The capacity of SDMA system depends on various parameters. For example, high rate users requiring more slots may be block-limited as opposed to interference-limited when users require less slots per call, leading to high blocking rate; clearly, the system can accommodate less users when they are unevenly distributed, when the SINR requirement by calls is high, when propagation environment is unfavorable, or when SLL is high. Taking multipath into consideration, different channel allocation algorithms, either more aggressive or more timid, can lead to different results. Aggressive algorithm can accommodate more users, but requires more computation, whereas timid way is to the opposite. As a whole, the capacity of EGPRS network with SDMA is very impressive, as shown by the simulation result in Figure 6.19, in which Channel utilization can be four times that of 1/1 frequency reuse system.

In conclusion, the simulation results drawn from our simulated network are very cheering. We believe that, an EGPRS network, together with properly designed SDMA and DCA strategy, can be a strong competitor to offer 3G services in the near future. We will use mathematical analysis and experiments using testbed to further verify our simulation results.

7.2 Future Work

➢ Throughout our work, the brick wall antenna model is used. The next step in our work would be to use a more realistic radiation pattern to analyze the performance of SDMA with some representative beamforming techniques.

➢ There are estimation errors when using calculation rather than measurement to get value of SINR. The errors can be reduced if more accurate environmental parameters and a more elaborate propagation channel model are used.

➢ In our simulations, we have dealt with only radio access part of a cellular network. Our simulation result is based on the assumption that, when SINR requirement is met, the data rate carried by each channel is 48kbps. An EGPRS simulation bed taking different layers of data transmission into consideration will allow us to have more comprehensive and trusty performance analysis.

➢ We only analyzed downlink performance in our work. A comprehensive evaluation of downlink and uplink performance with consideration of large-scale path loss and small-scale fading and multipath needs to be done.

➢ A more specific data traffic model, user mobility model and propagation channel model in an EGPRS network need to be studied and applied in the simulations.

➢ The analysis of a combined application of the adaptive antenna, power control, dynamic channel allocation, packet scheduling, frequency hopping in AN EGPRS system is a very attractive but tough job.

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