Network Capacity and Quality of Service Management in F/TDMA Cellular Systems

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ABSTRACT

As a consequence of rapidly increasing mobile communications, efficient utilization of the scarce radio resources becomes one of the most important issues in the system evolution. Increase of the system capacity has been investigated in two ways. The first way is to replace the fixed channel allocation (FCA), with the more efficient dynamic channel allocation (DCA). The second way is to utilize those traffic channels not being used by voice services to provide a packet data service, like general packet radio service (GPRS) and cellular digital packet data (CDPD). In this thesis, the author have proposed two DCA schemes and developed an analysis method to investigate the GPRS impact on the GSM voice services. In addition, the GPRS downlink performance is investigated and some guidelines or principles for GPRS network planning have been presented.

In the proposed DCA algorithms, the effect of the channel allocation on existing calls is considered by the evaluation of the call outage rate or a cost function. In the first proposed algorithm, in order to evaluate the call outage caused by those candidate channels, a method of estimating the average signal to interference ratio (SIR) variation of on-going calls due to the assignment of a coming call has been developed. This algorithm improves the capacity or QoS performance compared with the first available and maximum SIR schemes. In the second proposed algorithm, a cost function has been introduced to estimate the cost of the assignment of a candidate channel. The performance evaluation shows that by using the cost-function for channel pre-selection the problem of high intracell handover rate for the first available based scheme can be decreased to an adequate level and the time of the call set-up can be shortened.

An analysis method to calculate the outage probability of the GSM-GPRS network for both the non-frequency hopping and frequency hopping systems has been presented to investigate the GPRS impact on GSM voice services. It is found that: GPRS affects more on the QoS of voice services of the network with small reuse factor; GPRS will reduce the cell service area, but the reduction percentage of the cell service area for the system with small reuse factor is higher than that for the system with large reuse factor; those channels unused by voice services might not all be used for carrying GPRS traffic; the number of unused voice channels which can be allocated to GPRS depends on the difference between the outage level of the existing GSM network and the maximum acceptable level.

From final part of this work, it is found that: GPRS capacity performance in downlink is quite different from that in uplink because of the difference in the transmission protocols; multiple-slot allocation does not show a gain of the mean throughput neither a decrease on the mean delay compared to single slot allocation. This result is different from the result of the uplink performance. In multi-rate services, the multi-slot services significantly increase the delay of the single-slot service, consequently, a control of the multi-slot services is needed.

PREFACE

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Shaoji Ni
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CONTENTS

ABBREVIATIONS

3G	The Third Generation
AMPS	Advance Mobile Phone System
ARP	Autonomous Reuse Partitioning
BLER	Block Error Rate
BS	Base Station
CDPD	Cellular Digital Packet Data
CDMA	Code Division Multiple Access
CS	Coding Scheme
CT-2	Enhanced Cordless Telephone
DCA	Dynamic Channel Allocation
DECT	Digital European Cordless Telephone
DTX	Discontinuous Transmission
FA	First Available
FCA	Fixed Channel Allocation
FDD	Frequency Division Duplexing
FDMA	Frequency Division Multiple Access
FH	Frequency Hopping
FM	Frequency Modulation
FSK	Frequency Shift Keying
GPRS	General Packet Radio Service
GSM	Global System for Mobile Communications
IS-95	Interim Standard 95
ITU	International Telecommunications Union
JDC	Japanese Digital Cellular
MCO	Minimum Call Outage
MP	Maximum Packing
MS	Mobile Station
MSIR	Maximum Signal to Interference Ratio
NMT	Nordic Mobile Telephone
PCCCH	Packet Common Control Channel
PDCH	Packet Data Channel

QoS	Quality of Service
SDCCH	Standalone Dedicate Common Channel
SIR	Signal to Interference Ratio
SORP	Self Organizing Reuse Partitioning
TDD	Time Division Duplexing
TDMA	Time Division Multiple Access
UMTS	Universal Mobile Telecommunications System
USDC	U.S. Digital Cellular

LIST OF PUBLICATIONS

P1 Shaoji Ni and Sven-Gustav Häggman, "Dynamic Channel Allocation Based on SIR Estimation", in proceedings of *the 2nd International Workshop on Multi-Dimensional Mobile Communication* (MDMC'96), Seoul, Korea, July 96, pp. 177-181.

P2 Shaoji Ni, "Cost-Function Based Distributed Dynamic Channel Allocation Algorithm," in proceedings of *the 3rd Asia-Pacific Conference on Communications* (APCC'97), Sydney, Australia, Dec. 1997, pp. 459-463.

P3 Shaoji Ni, "Distributed Channel Allocation Algorithm with Power Control," in proceedings of *the 8th IEEE International Symposium on Personal, Indoor and Mobile Radio Communications* (PIMRC'97), Sept. 97, Helsinki, Finland, pp. 406-410.

P4 Shaoji Ni, Yong Liang, Sven-Gustav Häggman, "Outage Probability in GSM-GPRS Cellular Systems with and without Frequency Hopping," *Wireless Personnel Communications,* Vol. 14, Sept. 2000, pp. 215-234.

P5 Shaoji Ni, Yong Liang, Sven-Gustav Häggman, "Outage Probability for GPRS over GSM Voice Services," in the proceedings of *IEEE Vehicular Technology Conference 1999-Fall* (VTC'99-Fall), 19-22 September, Amsterdam, The Netherlands, pp. 839-843.

P6 Shaoji Ni, Sven-Gustav Häggman, "GPRS performance estimation in GSM voice and GPRS shared resource system," in the proceedings of 1999 *IEEE Wireless Communication and Networking Conference* (WCNC'99), Sept. 22-24, 1999, New Orleans, USA, pp. 1417-1421.

P7 Shaoji Ni, "GPRS network planning on the existing GSM System", in the proceedings of *IEEE GLOBECOM 2000*, Nov. 27-Dec. 1, 2000, San Francisco, USA, Vol. 3, pp. 1432-1438.

CONTRIBUTIONS OF THE AUTHOR

In all the publications, the contributions of the author include the research ideas, theoretical analysis, construction of the simulation software and the writing of manuscripts.

In publications 4 and 5, the co-author, Ms. Yong Liang, contributed on some part of the simulation software construction and some computer simulations during the work on her Master’s thesis under the instruction of the author.

In publications 1, 4, 5, and 6, Prof. Sven-Gustav Häggman, being the supervisor of the author, contributed valuable technical comments.

1. INTRODUCTION

1.1 Motivations of the Work

The radio resource management includes the selection of: 1) the access port or base station—where the mobile should be camped; 2) waveforms or channels—which channels should be used for carrying the information; 3) the power of the transmitter—how much amount of the power should be used for transmitter. For voice services, the system capacity is defined by the number of simultaneously served subscribers per area (volume) unit at some predefined level of quality of service (QoS), e.g., call blocking and dropping rates, while for data services, it is defined by the throughput at some predefined level of QoS (e.g., time delay, error rate, reliability). The capacity and the QoS interact on each other and are the key parameters of mobile networks. In recent years, as the number of subscribers to the mobile radio systems has been growing rapidly, in order to increase the capacity of a network the efficient utilization of the scarce radio resources becomes one of the most important and difficult issues in the mobile communications. The capacity of existing F/TDMA systems could be improved by increasing the spectrum utilization efficiency, e.g., to replace the fixed channel allocation (FCA), with the more efficient dynamic channel allocation (DCA). In one of the third generation (3G) of mobile telecommunications systems, e.g., the universal mobile telecommunications system (UMTS) - time division duplexing (TDD) mode, the DCA has been selected [1] to be the method of the radio resource allocation. Another way of increasing the spectrum utilization efficiency could be to utilize those traffic channels not being used by voice services to provide a packet data service, like general packet radio service (GPRS) and cellular digital packet data (CDPD).

Since the number of parameters for evaluating the QoS of a system could be from 10 to 20 according to the suggestion of International Telecommunications Union (ITU), in this thesis, the parameters of QoS include the call blocking probability, call dropping probability, call handover rate, call outage probability (rate) and time delay (for GPRS).

The network capacity improvement is limited by the QoS supported by network. For example, DCA allows the entire set of channels to be reused at any cell site provided that the signal to interference ratio constraint is satisfied. However, parameters such as the quality deterioration of the on-going calls, the call dropping rate and the call reestablishing rate, caused by admitting a new call into the network, are very important parameters of QoS and affect the efficiency of the system. The efficiency of a DCA algorithm is constrained by the call setup time and the limited amount of the information of the coming mobile available for the system. More specifically, the system neither knows the exact position of the coming mobile nor allows to locate its position[1] before the channel allocation due to the constraint of a short call setup time. In addition since the traffic in the network is dynamically variable and the number of states for such a process is very large (number of cells multiply number of channels), it is impossible to get an optimal solution for channel allocation by computing in a short time.

[1] The location services (LCS) specified by UMTS is a complex system in the radio aspect and only is used for locating of emergency calls and the LCS subscribers. It can not be applied for the radio resource management.

For GPRS, according to the "capacity on demand" principle [2], the physical channels unused by voice services are allocated dynamically to the GPRS according to the actual needs for packet transfers. However, the introduction of GPRS into GSM networks without allocating new spectrum will increase the interference probability of voice services. In addition, the physical channel allocated to GPRS is shared by a few data users simultaneously. The cochannel interference to the voice users might vary rapidly and dramatically in the time interval from 20 ms to a few seconds depending on the transmitted packet data size, because the locations of those packet data users could be largely different. This effect could drive the system into an unpredictable and unstable situation besides increasing the interference probability of voice services. Therefore, the introduction of GPRS into GSM networks may affect the QoS of voice services.

In summary, the capacity of the existing and the future systems could be increased by improving the utilization of radio resources with DCA. The introduction of GPRS will have impacts on GSM voice services. It is necessary to develop new methods for network to control the radio resources and for operators to plan their networks efficiently. Those are the motivations of the thesis work.

1.2 Previous Work

Compared with FCA, DCA is potentially able to provide substantial capacity increase and eliminates frequency planning in network planning. During the past twenty years, many DCA algorithms have been proposed [3-12], and many claims have been made on improvement of the system capacity, but the range of capacity gains is still unclear. Due to the dynamic nature of frequency reuse, the quality of a radio link can be very sensitive to the DCA algorithms and system architecture. The main idea of all DCA schemes is to evaluate the cost of each candidate channel, and select the one with the minimum cost provided that certain interference constraints are satisfied. The selection of the cost function is what differentiates DCA schemes. The selected cost function might depend on future blocking probability in the vicinity of the cell [18], on the usage rate of the candidate channel [3][19], on the reuse distance [3][8][19], on the channel occupancy distribution under current traffic conditions [11], on radio channel interference measurement of individual mobile users [10, 12, 13], or on the average blocking probability of the system. The maximum packing DCA [4-5] assumes that the system has infinite capacity to rearrange every on-going call, without regard to the number of reassignments involved. In this method a new call is accepted if there is any possible reassignment of channels to calls in progress which results in a channel that is free within the interference region of the new call's cell. Because the maximum packing DCA requires complete knowledge of existing calls in the entire system and a very large computing capacity, it is impractical at all. The channel segregation DCA strategy [6-7] is a self-organized dynamic channel assignment scheme and does not need to measure the propagation information in advance. By learning from statistical data of carrier sense results, each base station takes its favorite channel independently so that unnecessary interference will not occur. As a result, each base station captures its favorite channels for dedicated use. The reuse partitioning [8-9] is a simple DCA allocation scheme. In this method several overlaid fixed channel allocation plans are used. Mobiles with the high received signal levels are assigned channels with small reuse factor, whereas mobiles with low received signal quality get channels with large cluster sizes. The effect of reuse partitioning is equivalent to increasing frequency reuse by sectoring cells into multiple concentric rings. A distributed DCA algorithm for a hierarchical cellular structure has been proposed in [55]. With this DCA algorithm, the resources are enabled to be shared not only between cells of the same hierarchy, but between layers. The implementation study of DCA to GSM system has been presented in [56]. It shows that the expected increase of system capacity of the selected DCA can be achieved by carefully choosing the values of the characteristic parameters of the considered DCA. A more detailed review of the DCA can be found from [51].

Recently, the cochannel interference computation result shows [60] that the capacity improvement with interference adaptive DCA schemes is better than that with either traffic adaptive or location adaptive DCA schemes. For the interference measurement based DCA schemes, e.g., first available (FA) and maximum Signal to Interference Ratio (SIR) schemes, channels are allocated without checking how the candidate channel will affect the SIR value of on-going calls in its neighboring cells. Such an allocation may cause the SIR values of some existing links to fall below the acceptable value. When the deterioration occurs, the power control (if the function exists) is requested to restore their quality. If it is unsuccessful, the intracell call handover is requested. If there is no free channel for handover, those calls will be dropped. Even if we can find a new channel for handover, the call handover may cause another established link to request handover or to be dropped. Such successive feedback phenomena may produce deadlocks [50] and increase the load on switching systems due to intracell handover, and create instability of network. Thus, the efficiency and the quality of service (QoS) of the network will be decreased due to causing an increase in the burstiness to the overall arrival rate and overall departure rate of a cell [57].

For the FA based DCA scheme, according to simulations [15] inadvertent dropping of calls caused by originating calls can occur so often that all unsuccessful (blocked or dropped) calls are unintentionally dropped calls and not blocked calls. In addition, an exhaustive search and too frequent intracell handover access (a successful call experienced average 2.2 times of handover reported in [15]) will decrease the system capacity and make it difficult to implement in real networks. The “active-link-protection” channel access scheme [24] tries to predict for a few initial channel probing iterations whether the admission process will be successful or not. However, this prediction needs the pathloss information between the new link and active links. That makes the proposed method difficult to be implemented.

The admission control is normally used by the network for protecting the QoS of existing users. Since the current cellular network are evolving toward soft capacity limited systems, the integration of channel allocation, admission control, power control into one radio resource management scheme would maximize the utilization of radio resources [68-75]. A general survey of researches in the integration of power control and admission control can been found in [68]. A distributed constrained power control algorithm has been proposed [69] for the noninteractive admission control, where the admission decision is instantaneously made based on system state, and interactive admission control, under which the new mobile is permitted to interact with one or more potential channels before a decision is made. However, the noninteractive admission control scheme is needed to maintained a SIR target depended on the arrival calls and the interactive admission control is at a slow convergence rate. The recent researches in [70-71] has shown the integration of channel allocation and admission control can better utilize the radio resources and guarantee the QoS. However, the complexity of those algorithms required large computing capacity let it difficult to be used in practical network. Therefore, first part of this thesis will focus on the development in the integration of channel allocation, admission control, power control into one radio resource management scheme.

For the maximum SIR (MSIR) or least interference based DCA scheme, an very interesting result is shown in [58] that the call handover rate increases steeply and decreases gradually as the traffic increase. It seems that selecting the least interference channel for channel allocation does not guarantee to have a low call handover rate. Different SIR threshold values, e.g., admission, readmission, target, minimum reassigned and dropped threshold values, are used to decrease the effect of the channel allocation in [59], however, such a consideration can not bring the maximum performance in capacity.

In the interference measurement based DCA schemes [10, 12, 13, 15, 57-59], the channel is allocated without evaluating how the candidate channel will affect the on-going calls in its neighboring cells. It seems that the researchers consider the least interference free channel to be least likely to cause excessive interference to other active users. This judgment is not very correct. The least interference channel does not mean directly that it will cause the least effect to other active users (in the next chapter, the author will show the result in more detail).

Therefore, in the first part of this work, the author will propose two DCA schemes, the minimum call outage (MCO) scheme and the cost-function based scheme, which considers not only the capacity improvement but also the quality deterioration of the on-going calls, the call dropping rate and the call reestablishing rate, caused by admitting a new call into the network.

The introduction of packet data services should have no any effect or negligible effect on the existing voice services. In order to guarantee the quality of service (QoS) of voice services, it may be necessary to allocate dedicated channels for the packet data services, especially for the GPRS with multiple applications and multiple class services. However, such a scheme with dedicated channels will reduce the number of channels provided for voice services and increase the blocking probability to an undesirable level. The dynamic sharing of the channels between voice services and packet data services seems not to have much impact on the capacity of voice services and creates an additional capacity for packet data services [46]. However, the system performance may be degraded, e.g., outage probability increase, due to the additional interference contributed by packet data transmission [47-49].

The studies of the impact on voice services due to overlaying packet data services found in the literature are mainly focused on the American advanced mobile phone standard (AMPS) [46-48]. The discussion in [49] about the effect of GPRS on GSM is only considering a simple case. The power control (with error), discontinuous transmission (DTX) and frequency hopping have not been considered in those discussions. Until now, it seems that there is no published paper to give an analysis method which can be applied for the discussion of the GPRS impact on the GSM voice services. In the second part of this thesis the author will present a method to calculate the outage probability of the GSM-GPRS network for both the non-frequency hopping and frequency hopping systems. The GPRS impact on the GSM voice services is discussed by analyzing the outage probability of GSM-GPRS network.

As the first trial GPRS services will be available soon, the GPRS network planning becomes an imminent issue for GSM network operators. Because GPRS network is planned on the top of GSM network, obviously, we cannot directly apply the GSM network planning method into GPRS network planning. A very short general description of GPRS radio network planning had been given in [61], however, the most important issue of GPRS radio network planning, e.g., the remaining capacity of a GSM radio network, has not been mentioned. Until now it has not found a published paper discussing this issue in detail yet. The studies of the spectrum efficiency [62-63] for GPRS show there is an optimal combination pair of the frequency reuse factor and the coding scheme. Earlier studies of GPRS performance found in the literature [33-36, 62-63] are simulated with a fixed number of channels used for data transmission. The GPRS performance in "on-demand" radio resources has shown in [64], but the availability of radio resources due to the interaction of voice services is still not considered. The essential point that GPRS is operated on the top of GSM radio network seems to have been ignored by many researchers. The throughput and delay performance of GPRS mainly depends on the number of radio resources allocated and the system interference level which determines the coding scheme adopted. However, the number of channels available to GPRS is a random variable depending on how many channels are used by voice services and the interference level of the network. In this thesis the author provide new results of the GPRS performance in varying radio resources and present guidelines or principles of GPRS network planning.

1.3 Objective of the Work

In this thesis, the author will firstly propose two DCA schemes where the effect of a candidate channel on existing calls is modeled by the call outage rate or a cost function. Thus, the algorithms consider not only the capacity improvement, but also the quality deterioration of the on-going calls, the call dropping rate and the call reestablishing rate, caused by admitting a new call into the network. Secondly an analysis method will be developed to investigate the GPRS impact on the GSM voice services. Finally, the GPRS downlink performance is discussed and some guidelines or principles for GPRS network planning are presented.

The objectives of this thesis work are: 1) to investigate the relationship between the network capacity and the quality of service in order to increase the utilization of radio resources; 2) to develop algorithms or methods for network control to manage the radio resources efficiently and for radio network planners to plan the network optimally.

1.4 Original Contributions

For those SIR measurement bases DCA schemes mentioned in Section 1.2, channels are allocated only on basis of the local signal and interference measurement without checking how the candidate channel will affect the QoS of on-going (cochannel) calls of neighboring cells. Such allocation may cause deadlocks for the network and high rates of the call reestablishment and the call dropping. The researchers may consider the least interference free channel to be least likely to cause excessive interference to other active users. This consideration is not very correct. The least interference channel (downlink interference) does not mean directly that it will cause the least effect (uplink interference) to other active users. Therefore, in *publications* 1-3 [P_1, P_2, P_3], the author contributes the research by the uplink interference consideration and introduces two new methods to minimize the possible effects of channel allocation on existing calls and optimizes the performance of the capacity and QoS. *Publication* 1 [P_1] introduces a method to estimate the outage effect of the candidate channels on existing calls. A channel pre-selection is done by the outage rate estimation for on-going calls causing by those candidate channels, before a channel is selected by measurements. In order to simplify the estimation method of the cost for candidate channels presented in *publication* 1, a cost-function is further introduced in *publication 2* [P_2] for channel pre-selection and the corresponding channel allocation algorithm integrated with power control is proposed in *publication* 3 [P_3].

Recently, the research of the GPRS impact on the GSM voice services has drawn the attentions of operators. However it has not been found a published paper to give an analysis method which can be applied for investigating this issue. In *publication* 4 [P_4], the author develops a new analysis method to calculate the outage probability of the GSM-GPRS network for both non-frequency hopping and frequency hopping systems. This method takes into account the Rayleigh fading, power control (with error), discontinuous transmission, and frequency hopping. The GPRS impact on GSM voice services is investigated by the outage probability calculation, which is discussed in *publication* 4 and 5 [P_4, P_5].

Earlier studies of GPRS performance are simulated with a fixed number of channels used for data transmission. The research result based on such an unrealistic condition may mislead operators. The essential point that GSM voice services affect the radio resource available to GPRS seems to have been ignored by many researchers. In *publication* 6 and 7 [P_6, P_7], the GPRS performance in a dynamically varied number of channels is discussed. An approximation method for the performance estimation of the single-slot service is discussed *publication* [P_6]. The analysis method may be useful for the preliminary designing of the GPRS system “buffer size”. GPRS downlink performance considering link adaptation of coding schemes in different traffic models is investigated in *publication* [P_7] and has shown its difference from the uplink performance published in papers [35-36]. The guidelines or principles of GPRS network planning are presented in *publication* 7 [P_7].

1.5 Outline of the Thesis

The thesis is organized as follows. In chapter 2, two DCA algorithms are proposed. The effect of a candidate channel on existing calls is discussed and modeled by the call outage rate or a cost function. In chapter 3, a method to calculate the outage probability of the GSM-GPRS network for both the non-frequency hopping and frequency hopping systems is presented. The impact of GSM voice services by the introduction of GPRS is discussed. In chapter 4, an approximation method for the performance estimation of the single-slot service is discussed. GPRS downlink performance considering coding schemes link adaptation for different traffic models is investigated and new results are presented. Some guidelines or principles of GPRS network planning are presented too. Finally, in chapter 5, the conclusions of the thesis are made and the comments for further research are given.

2. DYNAMIC CHANNEL ALLOCATION ALGORITHMS

2.1 Introduction

2.1.1 Frequency reuse concept

Frequency reuse is the core concept of the cellular mobile radio system. In the frequency reuse system, users in different geographic locations (different cells) may simultaneously use the same frequency channel. The frequency reuse system can drastically increase the spectrum efficiency, but if the system is not properly designed, serious interference may occur. Cellular radio systems rely on an intelligent allocation and reuse of frequency channels in a coverage region. Each cellular base station is allocated a group of radio channels to be used within a cell. Base stations are assigned channel groups which contain completely different channels from those in neighboring cells. Figure 2.1 illustrates the concept of cellular frequency reuse, where cells labeled with the same letter use the same group of channels. A cell cluster is a basic unit of the cellular networks, which is outlined in bold. The cluster size N is the number of cells in the basic unit. In this example, N is equal to seven.

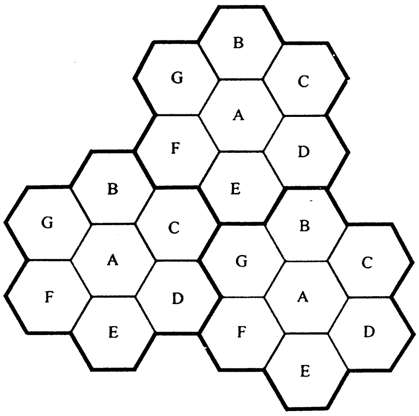

Figure 2.1 Illustration of the cellular frequency reuse concept. Cells with the same letter use the same set of frequencies.

2.1.2 Interference and system capacity

Interference is the major limiting factor in the performance of cellular radio system. It has been recognized as a major bottleneck in increasing capacity and is often responsible for dropped calls. Frequency reuse implies that in a given coverage area there are several cells that use the same set of frequencies. These cells are called co-channel cells, and the interference from users with the same channel in the other co-channel cells is called co-channel interference. Unlike thermal noise which can be overcome by increasing the signal-to-noise ratio (SNR), co-channel interference cannot be combated by simply increasing the carrier power of a transmitter. This is because an increase in carrier transmit power will increase the interference to neighboring co-channel cells. To reduce co-channel interference, co-channel cells must be physically separated sufficiently by a distance, called as reuse distance. For a network with the limited amount of frequency channels, a large reuse distance (cluster size) can guarantee a high QoS to the system, but the capacity will be decreased due to the decrease of the channel number in a cell.

Another interference of cellular systems is the adjacent channel interference. Adjacent channel interference results from imperfect receivers' filters which allow nearby frequencies to interfere the using frequency channel. Adjacent channel interference can be minimized through careful filtering and channel assignment. Since each cell is given only a fraction of the available channels, a cell need not be assigned channels which are all adjacent in frequency. By keeping the frequency separation between each channel in a given cell as large as possible, the adjacent channel interference may be reduced considerably. However, if the frequency reuse factor is small, the separation between adjacent channels may not be sufficient to keep the adjacent channel interference level within tolerable limits.

2.1.3 Fixed channel allocation and dynamic channel allocation

In a fixed channel allocation (FCA), each cell is allocated to a predetermined set of channels based on the frequency reuse concept. Any call attempt within the cell can only be served by the unused channels in that particular cell. In a dynamic channel allocation (DCA), channels are not allocated to different cells permanently. Instead, the base station allocates a channel to the call following an algorithm that takes into account the likelihood of future blocking within the cell, the frequency of use of the candidate channel or other cost functions. Compared with FCA, DCA has following properties of efficient spectrum usage, which are potentially able to provide substantial capacity increase [54]:

No trunking efficiency losses — Instead of subdividing the available spectrum into fixed groups, DCA allows the entire set of channels to be reused at any cell site. All radio channels could be made available everywhere for every call.
Average case interference scenario — In contrast to FCA, which requires pessimistic scenario (e.g. worst case) consideration, DCA allows a channel reuse factor relying on average co-channel interference. This transforms into lower margins for lognormal shadow loss.
Adaptive bandwidth sharing — In a dynamic multi-operator environment, DCA allows usage of channels to be proportional to each operator’s market share.

Besides improving capacity, DCA eliminates frequency planning in network planning. DCA not only allows significant capacity gains due to superior frequency reuse, but also mitigates efficiency losses arising from the traffic variation with spatial and time dependency. The main idea of all DCA schemes is to evaluate the cost of each candidate channel, and select the one with the minimum cost provided that certain interference constraints are satisfied. The selection of the cost function is what differentiates DCA schemes.

DCA can be divided into centralized and distributed schemes according to the control type employed. In centralized DCA schemes, the centralized controller collects the information it needs from all base stations in order to find a suitable channel for assignment. Simulation results have shown [3, 4] that centralized DCA schemes produce near-optimum channel allocation, but at the expense of the overload of the controller. As a result it is very complex and impractical for implementation. On the other hand, distributed DCA schemes [10-13] “distribute” the functions of the centralized controller into the base stations. Each base station makes its own decision on the channel assignment, using either local information about the usage state of channels in its own cell and in its nearby cells or the measurement information about the signal and interference. Since the distributed DCA schemes can adapt to the dynamic traffic situation and decrease the system complexity in channel allocation, it can be implemented more easily in F/TDMA systems. However, for distributed DCA, the signaling traffic load of the core network might increase and limit the capacity improvement.

For measurement based distributed DCA schemes in early studies, channels are allocated without evaluating how the candidate channels will affect the SIR value of on-going calls in its neighboring cells. Such channel allocation may cause the drop of the existing calls or a series of intracell call handover which will produce the decrease of the system efficiency and degrade the quality of service (QoS) of the network. The channel with least interference (downlink interference) or with maximum SIR does not mean directly that it will cause the least effect (uplink interference) to other active users. Therefore, in this chapter, the author contributes the research by the uplink interference consideration and introduces two new methods to minimize the possible effects of channel allocation on existing calls and optimizes the performance of the capacity and QoS. The minimum call outage (MCO) scheme and the cost-function based scheme, which will be introduced here, considers not only the capacity improvement, but also the quality deterioration of the on-going calls, the call dropping rate and the call reestablishing rate, caused by admitting a new call into the network.

2.2 Minimum call outage (MCO) dynamic channel allocation scheme

2.2.1 Estimation of the average effect of the channel allocation on existing calls

The performance of a channel allocation algorithm in uplink and downlink may have a little difference because of a slight difference in interference between the two links, however, the its performance characteristic and trend in uplink and downlink should be similar. Therefore, we only consider the downlink situation in the following for reducing the simulation time. For simplicity, we assume each base station to have the same transmitted power P on each channel and with omnidirectional antenna. Fast fading is not considered. The propagation model is assumed to have an average path loss exponent η. Then the received power of desired signal for mobile p in cell i and that of the interference signal from the link of mobile q in the k-th cell, respectively, are

$$S_{ip} = A \cdot P \cdot r_{ip}^{-\eta} \cdot 10^{\frac{\xi_{0ip}}{10}}, \qquad (2.\ 1)$$

$$I_{kq} = A \cdot P \cdot d_{kp}^{-\eta} \cdot 10^{\frac{\xi_{kp}}{10}} \beta_{pq}, \qquad (2.\ 2)$$

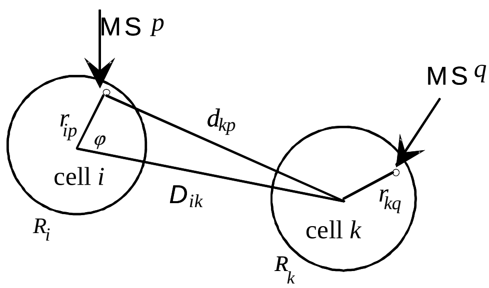

Figure 2.2 Illustration of interference cells.

where A is a proportional constant; r_{ip} is the distance between mobile p and the base station of the serving cell i (Figure 2.2), d_{kp} is the distance between mobile p (in cell i) and the base station of the k-th interfering cell; η is the an average path loss exponent; ξ_{0ip} and ξ_{kp} are log-normal random variables[1] (in dB) which refer to the slow fading of the desired signal of mobile p and the interference signal from the base station of interfering cell k respectively; β_{pq} is the attenuation factor (with respect to channel q to channel p) of interference by receiver filter.

The attenuation factor, β_{pq}, is introduced to describe the co-channel interference and the adjacent channel interference with one formula. For the co-channel interference, β_{pq} becomes 1, but for the adjacent channel interference, β_{pq} depends the receiver filter characteristic and the channel separation between channel p and channel q.

The desired signal and interfering signals, and different individual interfering signals are all assumed to be statistically mutually independent. From Eq. (2.1) and (2.2), the signal to interference ratio (SIR) is obtained:

$$SIR = \frac{S_{ip}}{\sum_{k\in\{n_i\}}\sum_{q\in\{m_k\}} I_{kq}} = \frac{r_{ip}^{-\eta}\cdot 10^{\frac{\xi_{0ip}}{10}}}{\sum_{k\in\{n_i\}}\sum_{q\in\{m_k\}} d_{kp}^{-\eta}\cdot 10^{\frac{\xi_{kp}}{10}}\beta_{pq}}$$

$$= \frac{1}{\sum_{k\in\{n_i\}}\sum_{q\in\{m_k\}} \frac{d_{kp}^{-\eta}}{r_{ip}^{-\eta}}\cdot 10^{\frac{\xi_{kp}-\xi_{0ip}}{10}}\beta_{pq}} = \frac{1}{\sum_{k\in\{n_i\}}\sum_{q\in\{m_k\}} w_{kq}}, \tag{2. 3}$$

where $\{n_i\}$ denoted the interfering cell set for host cell i; $\{m_k\}$ is the assigned channel set in the k-th interfering cell. Let

$$w_{kq} = (\frac{d_{kp}}{r_{ip}})^{-\eta} 10^{\frac{\xi_{kp}-\xi_{0ip}}{10}}\beta_{pq}\,. \tag{2.4}$$

The term w_{kq} is the contribution to the SIR reciprocal of call p in cell i by the admission of call q in cell k. The effect on the SIR value of the existing call p due to allocating channel q in cell k can be calculated from the term of w_{kq}.

The distribution of mobiles within a cell has effect on the capacity. Considering the non-uniform distribution of mobiles within a cell, we need to take into account of the distribution of call arrival, the distribution of the calls' moving directions and the speed of mobiles. It is so complicated that it is difficult to consider practically and theoretically. Fortunately, the power control can dilute the capacity effect of mobiles non-uniformly distributed in a cell. Therefore, it is assumed here that mobiles uniformly distributed within a cell to simplify mathematical calculation. It needs to note that the assumption does not imply the uniform traffic within a network. The traffic load in each cell in a network can be different. When a new call is arrival, the base station (BS) does not have the exact location of this mobile station (BS). Therefore

[1] In publication [P_1], the slow fading component is denoted with $10^{(-\xi/10)}$. Because the slow fading ξ is log-normal distribution, there is no difference in statistical characteristics between the slow fading in [P_1] and that here.

we simplify the MS location distribution with a uniform distribution and obtain the average effect the SIR of a candidate channel on existing calls. Applying probability density function (PDF) of uniform distribution, $f(s)=1/(\pi R^2)$ into Eq. (2.4), where R is the radius of a cell, we have

$$\overline{w_{kq}} = \iint\limits_{A} E\left[10^{\frac{\xi_{kp}-\xi_{0ip}}{10}}\right] \cdot \beta_{pq} \cdot \left(\frac{d_{kp}}{r}\right)^{-\eta} \cdot f(s)ds$$

$$= E\left[10^{\frac{\xi_{kp}-\xi_{0ip}}{10}}\right] \beta_{pq} \int_0^{2\pi}\int_0^{R} \frac{\left[(D_{ik})^2 + r^2 - 2rD_{ik}\cos\varphi\right]^{-\frac{\eta}{2}}}{r^{-\eta}} \frac{1}{\pi \cdot R^2} rdrd\varphi$$

$$= E\left[10^{\frac{\xi_{kp}-\xi_{0ip}}{10}}\right] \frac{\beta_{pq}}{\pi} \int_0^{2\pi}\int_0^{1} \frac{\left[(\frac{D_{ik}}{R})^2 + x^2 - 2x\frac{D_{ik}}{R}\cos\varphi\right]^{-\frac{\eta}{2}}}{x^{-(\eta+1)}} dxd\varphi \tag{2.5}$$

where D_{ik} is the distance between centers of cell i and cell k. Eq. (2. 5) can be integrated by numerical calculation.

Assuming that channel q in cell k is a channel which will be assigned to a new call, then $\overline{w_{kq}}$ can be taken as the average effect on the SIR of the existing call p in cell i from this assignment. Hence, if the channel q is allocated, The SIR value of call p, SIR_{old}, will be approximately changed to

$$SIR_{new} \approx \frac{1}{\frac{1}{SIR_{old}} + \overline{w_{kq}}} \tag{2.6}$$

The proposed MCO scheme is based on the SIR estimation with Eq. (2.6).

2.2.2 MCO dynamic channel allocation scheme

The timid DCA algorithm is such an algorithm that a channel is allocated only if there are no nearby interfering cells using this channel. It has been shown in previous work [11] that the timid DCA outperforms the FCA at very light traffic load. However, for heavy traffic load, the timid DCA performance in call blocking probability shows its inefficiency. On the other hand, the aggressive DCA algorithm is such that the mobile station may take a channel providing the SIR constraints are satisfactory even if it is being used in nearby interfering cells. The call blocking performance can be improved significantly by aggressive DCA, however, this improvement is obtained at the expense of high call reconfiguration (intracell handover) and call dropping rates, especially for heavy traffic load. Consequently, the aggressive DCA algorithm may cause signaling overload and poor QoS of the network, when it is operated in a practical system. It would be more attractive that the DCA algorithm improves the call blocking performance on basis of keeping the call handover and call dropping rates at an acceptable level. This is the main purpose of the MCO scheme.

A. Signal to Interference Ratio (SIR) Table

In the MCO scheme, the distributed DCA strategy is adopted. Each base station is a controller executing a DCA scheme for making decisions on channel assignment within that cell. Information exchange is limited to its neighboring interfering cells only. At each base station, a SIR table is maintained, which contains the SIR values of occupied channels of its own cell and its interfering cells (Table 2.1). After a channel is allocated to a call, the host base station changes the channel state from “free” into “busy” by updating corresponding SIR measurement value of this channel into the SIR table. In addition, for the existing calls (affected by this new allocated call) in interfering cells, their new calculated SIR values are updated into the SIR table with Eq. (2.6) as well.

At the same time, the channel allocation information, such as the cell number, the channel number and its SIR value, is exchanged to its interfering cells. The neighboring interfering cells will proceed with the same updating procedure.

In contrast to channel allocation, after a channel is released, the base station changes the channel state from “busy” into “free” by updating the SIR table with zero value for this channel. In addition, for those existing calls (affected by this new allocated call) in interfering cells, their new calculated SIR values are updated into the SIR table with:

$$SIR_{new} \approx \frac{1}{\frac{1}{SIR_{old}} - \overline{w_{kq}}}, \tag{2.7}$$

The efficiency of the MCO scheme depends on the updating frequency of those SIR tables besides the updating in call set-up and release. If we often update those SIR tables with measurement values, we can evaluate the effect on existing calls by a candidate channel more correctly. However, if the SIR information is frequently exchanged between base stations, the load of the core network will be increased significantly.

	channel				
cell	1	2	3	...	M
i_0	12.5				
i_k		15.0			11.2
...					
i_n		12.0			10.3

Table 2.1 An example of the SIR table at cell site i_0. The first row represents SIR values of all channels at cell site i_0; the other rows represent those at neighboring interfering cells of cell i_0.

B. Minimum Call Outage Scheme

The First Available (FA) and maximum SIR (MSIR) (also called Least Interference) schemes mentioned in [25], can be classified as aggressive schemes, because the deterioration of SIRs of on-going calls is not considered in channel selection. When a channel is assigned to a call, it is possible that this allocation will cause the SIR of already established links to deteriorate. We usually call such a deteriorated link whose SIR is lower than the threshold value γ as “outage links (calls)”. The call outage rate P_{outage} is defined as the ratio of the number of outage call to the number of on-going calls. If the call outage rate is minimized, the probability of call dropping will decrease and the quality of service will increase. The algorithm of the Minimum Call Outage (MCO) scheme is

- Each BS maintains a SIR table. The values of this table are updated when a call is allocated or released in its cell and neighboring interfering cells.
- When a call arrives in a cell, the base station searches through all free channels for those channels whose SIR values are larger than the threshold value γ. Then, the base station evaluates the deterioration of the SIR of on-going calls (only within its interfering cells) according to its SIR-table and equation (2.6) for those channels. The channel that causes the minimum call outage rate is assigned to the new call. If there is more than one channel giving the same the call outage rate, we randomly choose one of these channels, i.e., the channel p to be assigned should satisfy:

    (1) $SIR^{(p)} > \gamma$,
    (2) Channel p minimizes the outage rate of existing calls.

- If there is no channel with SIR larger than the threshold value γ available, then the call is blocked.

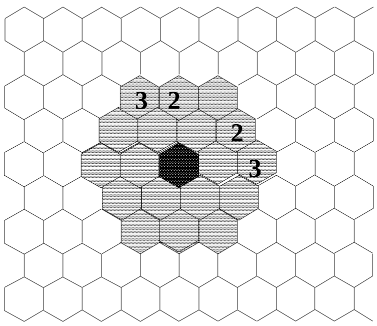

Figure 2.3 Simulation network.

-host cell, - interfering cells.

2.2.3 Simulation results and discussions

A. Simulation Model

In the MCO scheme we try to maximize the number of mobiles that can be admitted into the network and minimize the outage rate of on-going calls. The nature of this DCA problem does not allow any simple analytical optimization. So the scheme is investigated by simulation. In the simulation, the network model is a two-dimensional regular hexagonal grid with 81 cells (9×9) (Fig. 2.3). In order to avoid the boundary effect, the left-most and the right-most columns are “neighbours” with each other, and so are the top and the bottom rows. Thus, the results are representative of an infinite system, and therefore can be applied to a large network. Around a host cell, only two tiers of cells (6 cells in the first tier and 12 cells in the second tier) are considered to be the interfering cells. The number of channels available to the system is 60. The co-site channel separation constraint is 4 channels. The attenuation factor β is assumed to be 1.0, 0.05, 0.001, 0.00005, for the channel separation = 0, 1, 2 and 3 respectively. For channel separation larger than 3, we set β to be 0. The propagation model is assumed to have an average path loss with an inverse fourth power (η = 4) distance dependency, and lognormal slow fading with a σ = 8 *dB* standard deviation. The term of $E[10^{(\xi_{kp}-\xi_{0ip})/10}]$ in Eq. (2. 5) is calculated with the method given in [14], but the value is sensitive to the integral range of $\xi_k - \xi_0$ adopted in the calculation. If the integral range is still adopted in the range of (-4σ, 2σ), according to the calculation, for a σ = 8 *dB* standard deviation of slow fading, the value is approximately 4. The threshold value of SIR for assigning a channel to a new call is chosen as γ = 12 *dB*. If the SIR of an on-going call is deteriorated below 10 *dB*, the call is dropped. To investigate the nature of the algorithm, we do not consider any intracell and intercell handover and power control in the simulation system. Hence, the movement of mobile stations is ignored here, and the "call outage" is created by the new call's admission. The call arrivals in each cell constitute independent Poisson processes. The duration of each call is exponentially distributed with mean $1/\mu$ = 180 s. The call traffic is spatially uniformly distributed.

B. Simulation Results [P_1]

To evaluate the system performance two extra parameters are defined as:

$$\text{drop probability} = \frac{\text{number of dropping calls}}{\text{number of admitted calls}} \qquad \text{outage probability} = \frac{\text{number of outage calls}}{\text{number of admitted calls}}$$

The performance of FA, MSIR and MCO schemes is shown as the blocking probability, dropping probability and outage probability (Fig. 2.4–2.6). The blocking probability for FCA (N = 7) is also shown in Fig. 2.4. With uniform traffic distributions, each DCA scheme performs significantly better than the FCA scheme as expected. For a 1% blocking probability, the DCA schemes can offer about 40% - 60% of capacity gain. Among those three DCA schemes, the First Available scheme is a fast setting up algorithm and the blocking probability performance is best, however, the probability of call dropping is the highest, and especially the probability of call outage (10 *dB* ≤ SIR < 12 *dB*) is very high. Even though for the traffic load with a 1% call blocking probability, its outage probability is still as high as 30%. Therefore, for such high outage probability, it is very easy to understand the simulation result [15] that with FA scheme the unintentional dropping of calls caused by originating calls can occur so often that all unsuccessful (blocked or dropped) calls are unintentionally dropped calls and not blocked calls. For the MSIR scheme, the probability of call dropping is lowest, but this obtained at the expense of highest probability of call blocking. For a 1% blocking probability, it has approximately 9% traffic load less than the other two schemes. For the MCO scheme, the call blocking probability is almost similar to that of the FA scheme, but it has much more improvement in the performance of probability of call dropping and probability of call outage.

The comparison of performance for the three DCA schemes with spatially non-uniform traffic (hot spots' traffic) has been shown in [P_1]. The performance shows the similar characteristics as that obtained with uniform traffic.

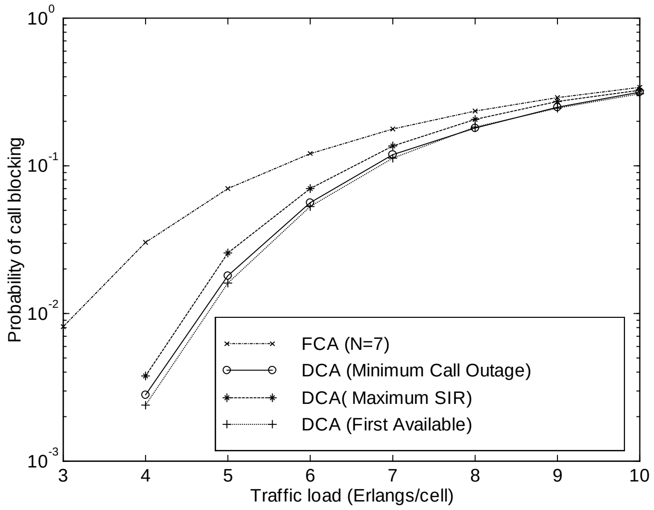

Figure 2.4 Probability of call blocking with uniform traffic.

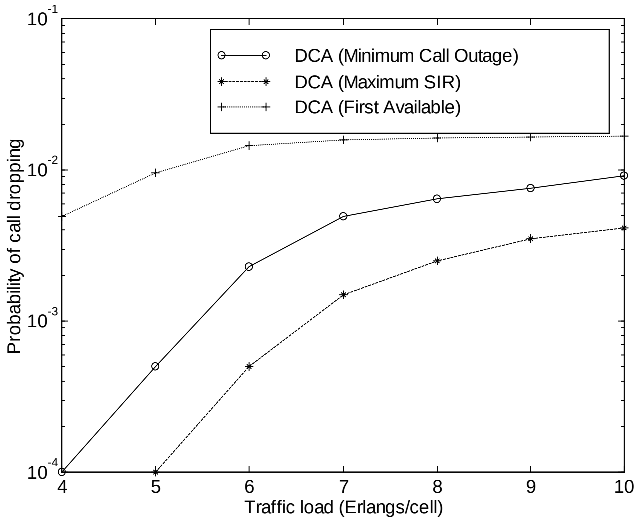

Figure 2.5 Probability of call dropping with uniform traffic.

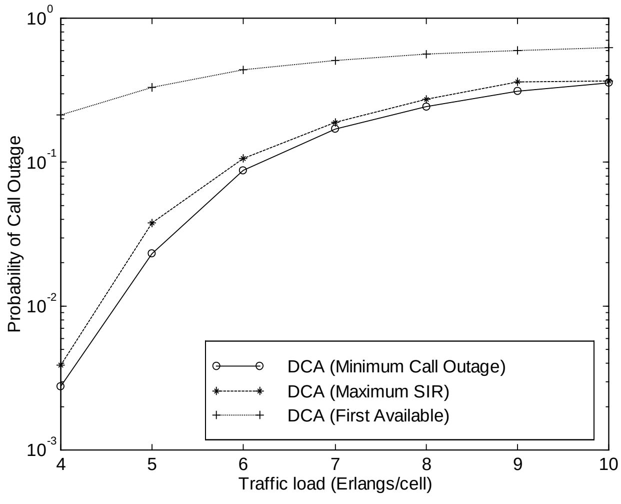

Figure 2.6 Probability of call outage with uniform traffic.

C. Discussions

The MCO scheme minimizes the call outage probability and consequently reduces the call dropping probability when assigning a channel to a new call. It shows some improvement in QoS compared with that of FA and MSIR schemes. However, the MCO scheme has following limitations for implementation:

- Due to the limitation of network signaling load, the SIR values of calls cannot be exchanged very frequently. There is a trade-off between the frequency of SIR exchange and the performance of MCO. More frequent the SIR exchange, the better the performance of MCO. The determination of the frequency of SIR exchange should consider the capacity of network signaling load and user traffic load. Consequently, the SIR values in a SIR-table may be obsolete values. The use of obsolete SIR values may have a wrong call outage rate and may cause a wrong decision on channel allocation.
- The exact position of users within cells cannot be taken into account when the cost (or call outage rate) of a candidate channel is evaluated.
- Even though many algorithms have been proposed [16-17] for SIR estimation, the SIR measurement in existing cellular system is based on the Bit Error Rate (BER) measurement and the error of SIR measurement might be more than 1 *dB*. That may bring more error to the estimation of the call outage rate.
- The algorithm needs to evaluate the outage probabilities of all free channels. The average call setup time of the MCO scheme cannot be guaranteed to be shorter than that of First Available and Maximum SIR schemes.
- The MCO scheme given here is based on the homogeneous network model with equally sized hexagonal cells. In the practical system, different cell size and cell geometry would co-exist in a network and the transmitted power for different call would be different. As an approximation, we can use the average transmitted power of each cell in place of the constant power to evaluate the average SIR effect of existing calls. The estimation of the call outage rate will be much more complicated.

In the following section, we will introduce a cost function to improve the evaluation of the cost of a candidate channel. The proposed cost-function based DCA algorithm is simpler and easier to implement.

2.3 A Cost Function Based Dynamic Channel Allocation Algorithm

2.3.1 Cost Function

The main idea of all DCA schemes is to evaluate the cost of each candidate channel, and select the one with the minimum cost provided that certain interference constraints are satisfied. The selection of the cost function is what differentiates DCA schemes. Here we introduce a cost function based on the average interference.

When a new call is accepted into the network, it might cause quality deterioration of ongoing calls. The cost of a call admission depends on the assigned channel l, the distance D_i between cell centers of co-channel users of channel l, the location (r_i, θ_i) of all cochannel users, and the transmitted power set $P \in P_i$ $(i \in n)$ of all cochannel users. The cost function is:

$$C = C(P_i, r_i, \theta_i, l, D_i), i \in n \tag{2.8}$$

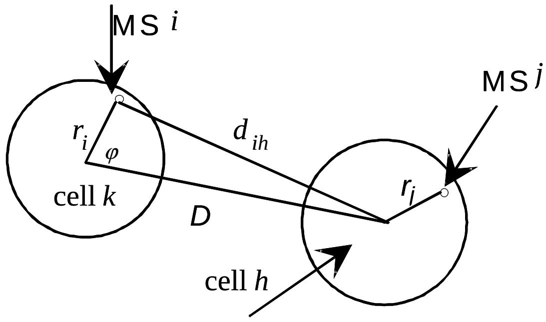

Figure 2.7 Downlink interference.

The performance of a channel allocation algorithm in uplink and downlink may have a little difference because of a slight difference in interference between the two links, however, the its performance characteristic and trend in uplink and downlink should be similar. Therefore, we only consider the downlink situation in the following for reducing the simulation time.. For any cell, normally, only two tiers of cells are considered as its interfering cells. If a cochannel user is assigned in the host cell, for the second tier of interfering cells (Fig. 2.3), the co-channel interference to cells of type 3 ($D/R = 2\sqrt{3}$) is different from that to cells of type 2 ($D/R = 3$). As shown in Fig. 2.7, the downlink cochannel interference power of mobile *i* in cell *k* from the link of mobile *j* in cell *h* is:

$$I_j = A \cdot P_j d_{ih}^{-\eta} 10^{\xi_j/10}, \tag{2.9}$$

where $d_{ih} = (D^2 + r_i^2 - 2Dr\cos\varphi)^{1/2}$; P_j is the downlink transmitted power to mobile *j*; ξ_j is the slow fading variable with log-normal distribution and parameter *A* is a proportional constant. Assuming that the mobile users are uniformly distributed within a cell, the average interference power with respect to the whole cell area in cell *k* is:

$$\bar{I} = \iint_s I_j f(s) ds$$

$$= A \cdot R^{-\eta} P_j \int_0^{2\pi} \int_0^{R} E\left[10^{\xi_j/10}\right] \left(\frac{R}{d_{ih}}\right)^{\eta} \frac{1}{\pi R^2} r dr d\varphi . \tag{2.10}$$

By numeric calculation, for η = 4, the average interference for $D/R = 3$ and $D/R = 2\sqrt{3}$ is respectively:

$$\bar{I} = A \cdot R^{-\eta} P_j E\left[10^{\xi_j/10}\right] \cdot \frac{C}{\pi}, \tag{2.11}$$

$$\text{where } C = \begin{cases} 0.1363 & \text{for} \quad D/R = 3 \\ 0.0736 & \quad\quad\;\; D/R = 2\sqrt{3} \end{cases} .$$

For a scheme with no power control, the transmitted power is assumed to be the same for all users. For a scheme with power control, because the power control (SIR-balancing scheme) is to locally minimize the interference probability in the aspect of the transmitted power [21-22], the transmitted power is not considered as a factor in our cost function. If the difference of $E[10^{\xi/10}]$ between different cells is ignored, and only the variable *D* in Eq. (2.11) is considered, it can be seen that the average interference with the distance $D = 3R$ is approximately twice as much as that in the distance $D = 2\sqrt{3}R$. That means that, if assigning a new user in the host cell, the cost to the cochannel user in type 2 interfering cells (Fig. 2.3) is almost twice as much as that in type 3 interfering cells. That implies that if the cost of this channel allocation to calls in type 3 interfering cells is considered as 1, then the cost to calls in type 2 cells should be considered as 2. In addition, the average interference to the third tier's cells is no more than 1/3 of that to type 3 interfering cells. Any DCA normally has the property to avoid using the same channel in neighboring cells because of the high cochannel interference. The number of the common active channel set between the first tier's cells and the third tier's cells will be much high than that between the second tier's cells and third tier's cells. Even if the first tier cells and the third comment active channels, if a high cost for which the same channel is assigned in neighboring cells (first tier cells) have been considered, it is reasonable to ignore the effect of channel allocation on the active calls in the third tier's cells.

We use the average interference to construct a cost function for channel allocation. For cell x, let $F(x)$ denote the available channel set in cell x, $H(k)$, the occupied channel set in interfering cell k, $I_1(x)$, the set of the first-tier of interfering cells, $I_{22}(x)$, the set of the second-tier of type 2 interfering cells, and $I_{23}(x)$, the set of the second-tier of type 3 interfering cells (Fig. 2.3). Based on the interference analysis, we define the cost to the interfering cell $k \in \{I_1(x), I_{22}(x), I_{23}(x)\}$, due to allocating channel $l \in F(x)$ in cell x, as

$$C_x(k,l) = \begin{cases} 0 & \text{if } \; l \notin H(k) \\ c & \text{if } \; l \in H(k) \; \text{ and } \; k \in I_1(x) \\ 2 & \text{if } \; l \in H(k) \; \text{ and } \; k \in I_{22}(x) \\ 1 & \text{if } \; l \in H(k) \; \text{ and } \; k \in I_{23}(x) \end{cases}, \tag{2.12}$$

where the constant c is defined as a large value in order to avoid assigning a cochannel user in the first-tier of interfering cells as much as possible. The overall cost function to its interfering cells for channel l is

$$C_x(l) = \sum_{k \in \{I_1(x), I_{22}(x), I_{23}(x)\}} C_x(k,l) \; . \tag{2.13}$$

From Eq. (2.12), we can calculate the cost of each available channel in cell x. The cost of a channel roughly describes its effects on the on-going calls, if this channel is allocated. We will use the cost to decide the priority of a channel for allocation.

2.3.2 Performance of the cost-function in channel allocation

The abilities of an algorithm quickly making decisions about the access channel and transmitted power are key issues of the quality of service (QoS). Before the cost function in Eq. (2.13) is combined with the power control algorithm to be used for network resource management in the following section, the performance of the cost function in channel allocation (without power control) must be evaluated. We need to know what is the benefit of the cost function based algorithm, compared with some traditional channel allocation algorithm and what is the time efficiency of the cost function based algorithm.

A. Cost Function Based Channel Allocation Algorithm

To reduce the call dropping rate and to shorten the call set-up time, SIR values are measured only in a limited number of channels, called optional channels, chosen according to the priority list. The cost function in Eq. (2.13) is used to decide the cost of a channel. The lower the cost of a free channel, the higher the priority of the channel for allocation. The proposed algorithm is operated in the following way:

- For any cell, two tiers of cells are considered as interfering cells (Fig. 2.3). The channel state information (allocating or releasing) of each cell is locally exchanged to its interfering cells. Every cell maintains a list of the cost for all channels including the free channels and the occupied channels. The lower the cost of a free channel, the higher the priority of this channel for call set-up. The cost of a channel in a cell is updated (increased or decreased) in real time if a co-channel call is accepted or terminated (dropping and departure) in one of the cell's interfering cells. If we only consider assigning the cochannel users in the second tier of interfering cells, the maximum cost of a channel is 12. Hence, in order to avoid as much as possible assigning a cochannel user in the first tier of interfering cells, we choose the constant c in Eq. (2.12) as 13[*]. To reduce the aggressiveness of the algorithm, if the cost of a channel is higher than 23 (full of cochannel users in neighbour cells), the channel is marked in order not to allow its use for call set-up.
- When a call arrives in a cell, the free channel with the highest priority (lowest cost) is chosen for call set-up. When the measured SIR of this channel is higher than the predefined value, the channel is used for allocation. Otherwise, the SIR of another free channel with the next highest priority is measured until a channel is found for allocation or the new call is blocked. If all free channels have been measured, but the SIR requirement is still not satisfied, the call is blocked.
- When the SIR of an on-going call has fallen below the threshold value of call dropping, the call is dropped.

This algorithm tends to allocate a channel with minimum effects on on-going calls, hence, we call this algorithm the minimum cost algorithm.

A. Simulation Models

The performance of the cost-function based algorithm is investigated by simulation. In the simulation, the network model is same as the model in section 2.2 (Fig. 2.3). The channel model has an average pathloss with inverse fourth power (η = 4) distance dependency, and log-normal slow fading with zero mean value and a σ = 8 *dB* standard deviation. There are 36 orthogonal channels available in the system. The threshold value of SIR for assigning a new call is chosen as γ = 12 *dB*. The threshold value for call dropping is 10 *dB*. Omnidirectional antennas are assumed to be used in the system. The call arrivals in each cell constitute independent Poisson processes with uniform arrival rates. The duration of each call is exponentially distributed with mean $1/\mu$ = 120 s. The locations of calls are randomly generated within each cell. The mobility of mobiles is not considered. In order to evaluate the behavior of this cost-function in channel allocation in more detail, we do not consider the power control and handover and only simulate the performance of the cost-function based DCA scheme.

B. Simulation Results [P_2]

The purpose of this algorithm is mainly to maximize the number of mobiles that can be admitted into a network and to minimize the call dropping rate of on-going calls. In addition, the efficiency of the algorithm is also considered. The performance of first available (FA) and

[*] It may be chosen a lower value but the aggressiveness of this algorithm will increase.

maximum SIR (MSIR) schemes is also simulated as references to the cost-function based DCA (called the minimum cost) scheme.

The call blocking probability, call dropping probability, and unsuccessful (blocking and dropping) call probability for different traffic loads (uniform) are shown in Fig. 2.8–2.10, respectively. As shown in Fig. 2.8, the FA scheme has lowest call blocking probability; the minimum cost scheme has slightly higher call blocking probability than the MSIR scheme. However, Fig. 2.9 shows that the call dropping probability of FA scheme is the highest and much higher than that of the other two algorithms; the call dropping probability of the minimum cost scheme is the lowest. Obviously the low call blocking probability of the FA scheme is based on the high call dropping rate. The minimum cost scheme with a lower call dropping probability compared with FA and MSIR schemes shows its low aggressiveness. If this cost function is incorporated with power control and handover to manage the network resources, it may not cause high call reassignment rate and high signaling load. The parameter of the unsuccessful call probability which is defined as the sum of call blocking and dropping probabilities is more adequate to investigate the performance of an algorithm. Fig. 2.10 shows that the minimum cost scheme has the lowest unsuccessful call probability and the FA scheme has much higher unsuccessful call probability than the other two algorithms.

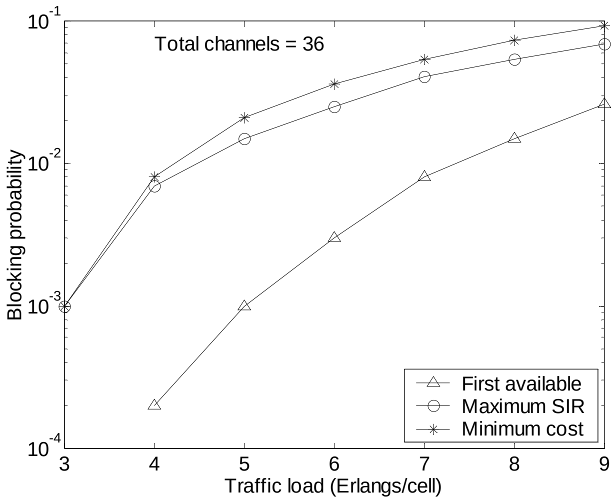

Figure 2.8 Call blocking probabilities with uniform traffic loads.

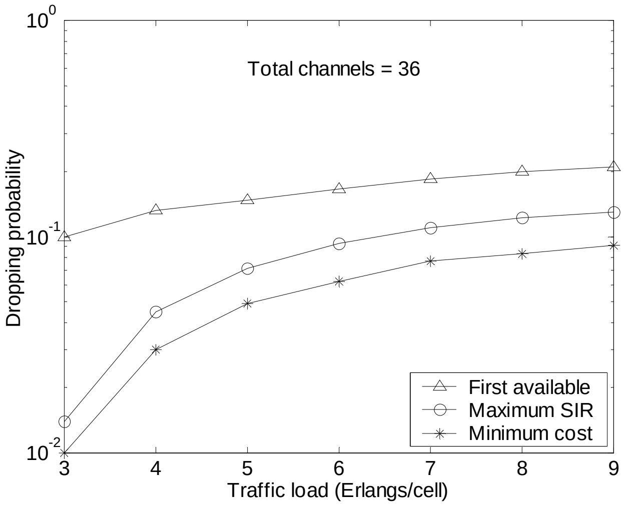

Figure 2.9 Call dropping probabilities with uniform traffic loads.

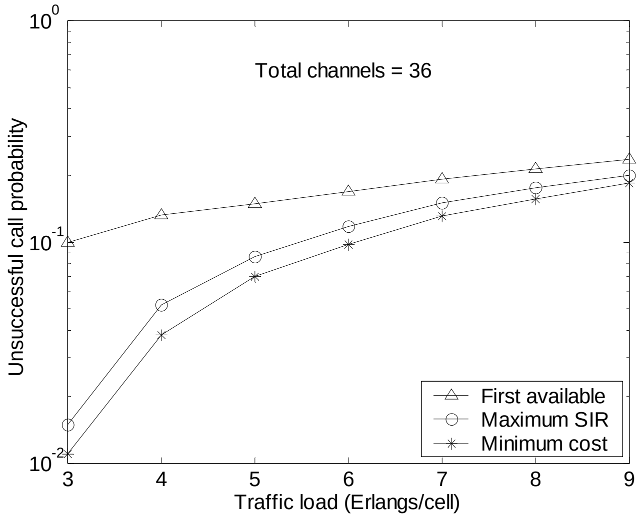

Figure 2.10 Unsuccessful call probabilities with uniform traffic loads.

The call set-up time is another important parameter for evaluating the performance of an algorithm. The more channels that have been measured, the longer the call set-up time. Because the MSIR scheme needs to measure all free channels in order to select the channel with maximum SIR value for allocation, of course, its call set-up time must be the longest one among the three schemes. In order to evaluate the set-up time of the minimum cost and FA schemes, we make a statistics in that the number of channels have been measured for each channel allocation and count the amount of such channel allocation, for which one channel has been measured, called as “once measurement” channel allocation, Fig. 2.11 shows the ratio of the number of the “once measurement” channel allocation to the total number of channel allocation for the minimum cost and FA schemes. We call this ratio as the “one-time-trial ratio”. The one-time-trial ratio for both minimum cost and FA schemes decreases with increasing the traffic load. That implies that the average call set-up time increases with increasing traffic load for both schemes. However, the one-time-trial ratio of minimum cost scheme is about 13.5–16.4% higher than that of FA scheme. At high traffic loads the difference of the two ratios is larger. Hence, the average call set-up time of the minimum cost scheme is shorter than that of FA scheme.

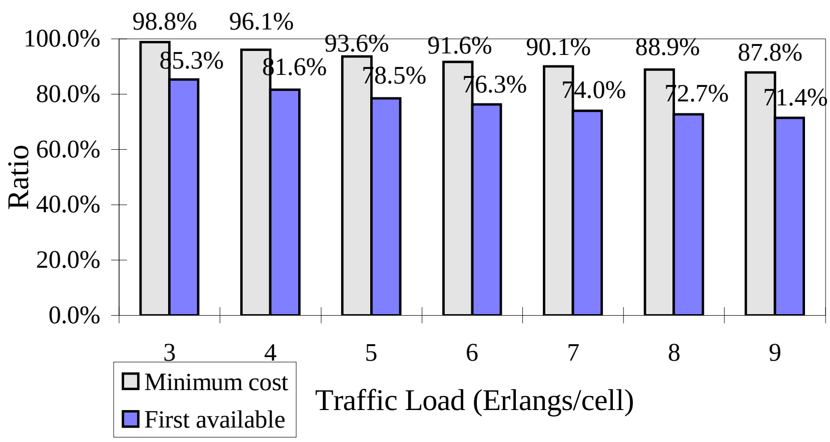

Figure 2.11 The “one-time-trial” ratios with uniform traffic loads for the minimum cost and first available schemes.

The cost of a channel introducing in this work roughly describes its effects on the on-going calls, if this channel is allocated. By using the cost-function to select a channel for call set-up, we find that cost function based DCA algorithm has lower call dropping probability and unsuccessful call probability than the FA and MSIR schemes; the FA scheme achieves the low call blocking probability by intensive call dropping. Therefore, if FA and MSIR schemes are combined with other techniques, such as power control and handover, it may cause intensive intracell handover rate and high signaling load, because of its high aggressiveness. In addition, for the call set-up time, the minimum cost scheme has better performance than FA and MSIR schemes.

2.3.3 A cost-function based distributed DCA algorithm with power control

The power control can suppress the adjacent channel interference (for non-orthogonal channels), the cochannel interference (for orthogonal channels), and minimize power consumption to extend terminal battery life. Undoubtedly, the power control can raise the network capacity. Some power control algorithms [20-21] based on the idea of balancing the SIR of all radio links have been proposed, but the final SIR achieved by those algorithms may be unsatisfactory for some of the links. Some calls must be dropped in order to keep the SIR of other calls higher than the predefined threshold value. Obviously, the efficiency of radio resource management is dependent on the channel assignment and the power control. The combination of DCA and power control to obtain some substantial capacity gains has been reported in [15], however, because no channel pre-selection is done before the channel probing procedure, inadvertent dropping of calls caused by originating calls can occur so often that all unsuccessful (blocked or dropped) calls are unintentionally dropped calls and not blocked calls. In addition, an exhaustive search and too frequent intracell handover access (a successful call experienced with average 2.2 handovers as reported in [15]) will decrease the system capacity and make the algorithms difficult to implement in real networks. Here, a distributed DCA algorithm with power control is proposed. In this algorithm, the cost function introduced in section 2.2.1 is used for channel pre-selection, and the power to be assigned is probed on the pre-selected channels.

A. Power Control

It is assumed N co-channel users are served in a system. One of them, i, transmits with the power P_i and the corresponding signal to interference ratio (SIR) is γ_i. The following distributed power control algorithm is used to search for a locally optimal power for the ongoing calls, which is similar to the algorithm used in [21]:

$$P_i(k+1) = \max\{ P_{min}, \min\{ \delta_i(k) * P_i(k), P_{max}\} \}, \qquad (2.14)$$

$$\text{and } \delta_i(k) = \gamma / \gamma_i(k),\ i = 1, 2, ..., N,$$

where $P_i(k)$ denotes the k-th discrete time transmitted power of call i; P_{max} and P_{min} are the maximum and minimum transmitted power respectively; γ is the SIR threshold value. The algorithm tends to set the SIR value of every call to the threshold value. The convergence of this algorithm has been proven [22] and the convergence properties hold [22-23] even when each user is allowed to have different power updating rate and when these updates are asynchronous.

B. Dynamic Channel Allocation Algorithm with Power Control

The channel assignment and power assignment will be integrated into a distributed channel access algorithm here. In this algorithm, the cost function introduced in Section 2.3.1 is used for channel pre-selection, and the assigned power is probed on the pre-selected channels. The proposed algorithm is operated in the following way:

- For any cell, two tiers of cells are considered as interfering cells (Fig. 2.3). The channel state information (allocating or releasing) of each cell is locally exchanged to its interfering cells. Every cell maintains a list of the cost for all channels. The cost function in Eq. (2.13) is used to decide the cost of a channel. The cost of a channel in a cell is updated (increased or decreased) in real time if a co-channel call is accepted or terminated (dropping and departure) in one of the cell's interfering cells. If we only consider assigning the cochannel users in the second tier of interfering cells, the maximum cost of a channel is 12. Hence, in order to avoid as much as possible assigning a cochannel user in the first tier of interfering cells, we choose the constant c in Eq. (2.12) as 13[*]. To reduce the aggressiveness of the algorithm, if the cost of a channel is higher than 23 (full of cochannel users in neighbour cells), the channel is marked in order not to allow its use for call set-up.
- When a call arrives in a cell, the free channel with highest priority (lowest cost) is chosen for call set-up and the call power probing process is activated. The procedures of the power probing for a new (or handover) call are:

    1. Assigning the minimum transmitted power P_{min} to the new call p.
    2. Measuring the SIR value γ_p of the call.
    3. If $\gamma_p < \gamma$, adjusting the power of the call according to Eq. (2.14) and going back to step 2; if $\gamma_p \geq \gamma$, and $P_{min} \leq P_p(k) \leq P_{max}$, this call is admitted into service with this power and the call power probing process is ended.
    4. If a power can not be found in the range of $[P_{min}, P_{max}]$ with which the SIR value $\gamma_p \geq \gamma$ or the probing iteration number (10 is used here) is larger than a pre-assigned value, the probing is moved to the next highest priority channel. Actually, an exhaustive searching is not allowed in a system. Hence, we prescribe that if four channels have been evaluated, but the SIR requirement is still not satisfied, the call is blocked.
- If a call is in service, the power control algorithm in Eq. (2.14) is used to maintain its quality. Each base station monitors its own served calls at some time interval. We assume that all base stations are synchronized (actually the algorithm works asynchronously either). When the SIR of a call falls below the target value, the power control procedure is requested. However, if the maximum transmitted power is requested or the number of iterations of power level adjustment is larger than the allowed value, but the SIR is still below a specified value (e.g., the call dropping threshold value), the handover procedure is requested. The "call set-up" procedure will begin to search for a channel for handover. If a channel is found, the call is moved to this channel. Otherwise, the call is dropped.

C. Simulation Models

The performance of this algorithm is investigated by simulations. In these simulations, the network model is the same as the model in Section 2.2 (Fig. 2.3). The channel model, traffic model and antenna model are the same as those used in Section 2.3.2. There are 36 orthogonal

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[*] It may be chosen a lower value but the aggressiveness of this algorithm will increase.

channels available in the system. The cell radius is 5 km. The maximum and minimum transmitted powers are assumed to be 20 and 0.02 Watts respectively (30 *dB* range). It is assumed that all channels in all cells have a noise level of -120 *dBm* at the receivers. The threshold value of SIR for assigning a channel to a new call and the target value of ongoing calls are both chosen to be $\gamma = 12$ *dB*. The threshold value for call dropping is 10 *dB*.

D. Simulation Results [P_3]

Because in Section 2.3.2, the performance of the (pure) cost-function based DCA scheme has shown that the call dropping and call unsuccessful rates can be reduce and the call set-up can be speeded up, compared with that of FA scheme and MSIR scheme, we do not make such comparisons again here. The purpose of this algorithm is mainly to maximize the number of mobiles that can be assigned into a network and minimize the call dropping rate of ongoing calls. In addition, the efficiency of the algorithm is also considered. To evaluate the system performance, two extra parameters, R_h, the intracell handover rate, and R_{uh}, the unperformed-handover rate, are defined as:

$$R_h = \frac{\text{number of requests of intracell handover access}}{\text{number of admitted calls}},$$

$$R_{uh} = \frac{\text{number of unperformed handover calls in successful calls}}{\text{total number of successful calls}}.$$

The physical meaning of R_h is the average number of intracell handover accesses caused by admitting a new call. The R_{uh} is the ratio of the successful calls which the intracell handover is not needed to total successful calls. In following simulations, the number of call arrivals varies from 175,000 to 218,000 depending on the traffic load. The SIR of each service call is measured once per second. In the meantime the power control procedure is activated. In power control, if the maximum transmitted power is requested or the number of iterations of power level adjustment has been 10, but the SIR is still below 10 *dB*, the handover procedure is requested. If there is no qualifying channel for handover, the call is dropped.

The blocking and dropping probabilities as function of traffic load (uniform) are shown in Figure 2.12. The blocking probabilities of FCA with a reuse factor of three (N = 3) are also shown. We find that at the load of approximately 9.4 Erlangs our scheme performs with 1% blocking and 1.2% dropping probabilities while FCA shows about 9.7% blocking probability. The system capacity has been largely improved compared with FCA. Because a large power range (88 *dB*) is used in the simulation of the paper [15], we can not make a direct comparison of the performance shown here with the performance shown in [15]. However, from the simulations in Section 2.3.2 which have shown that the (pure) cost-function based DCA scheme outperforms the FA scheme, we at least can infer that the performance in call handover and call setup time will be better than their result. The simulation results of the handover rate will give more support to the statement.

The intracell handover rate is shown in Fig. 2.13. It varies from 34.3% to 44.1%. A very interesting observation is that the handover rate initially increases with increasing traffic load, but for traffic loads exceeding 11 Erlangs it decreases with increasing traffic load. One reason is that the dropping and blocking probabilities increase with increasing traffic load; for high traffic load, while the number of calls in network reaches a certain value, the system controls the call admission (due to the property of cost function) and only those less aggressive calls are allowed into the network (Fig. 2.12 shows that there is a cross-point between the curves of the blocking probability and the dropping probability). The unperformed handover rate (Figure 2.13) gives more evidence for this explanation. The unperformed handover rate varies slightly from 87.5% to 91.4%. The figure of the two "rates" implies that most of admitted calls are not aggressive and successful calls experiencing multiple handover is only a few percent.

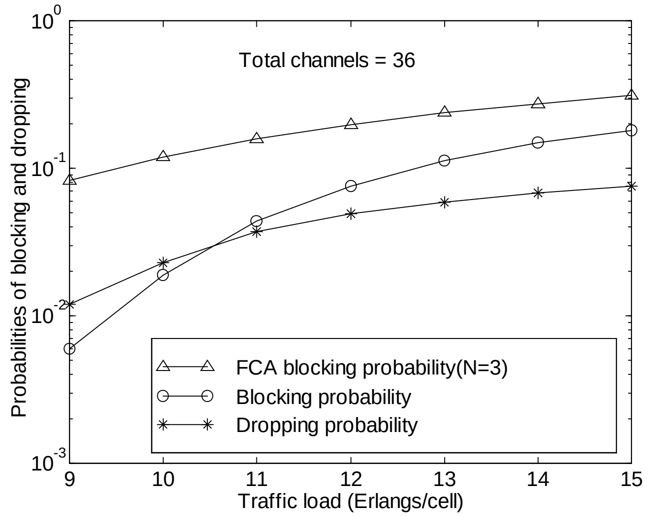


Figure 2.12. Probabilities of blocking and dropping with uniform traffic loads.

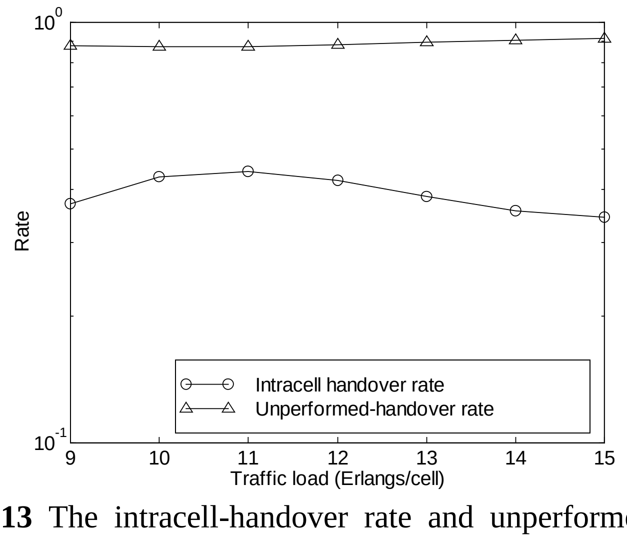


Figure 2.13 The intracell-handover rate and unperformed handover rate with uniform traffic loads.

In order to evaluate the speed of call setup, we simulate how many allocated channels (allocated to new calls and handover calls) are chosen from free channels with the highest priority (called number 1) and the second highest priority (called number 2). Figure 2.14 shows the percentage of allocated channels in the sequence of the priority list with different traffic loads. More than 98% of allocated channels are chosen from the highest and next highest priority channels. The number of allocated channels from the highest priority channels slightly increases at high traffic loads. This algorithm is designed to search channels for call set-up from the four highest priority channels whose cost is not larger than a specified value, but most of them are from the first two highest priority channels. Hence, this algorithm performs with a short call setup time.

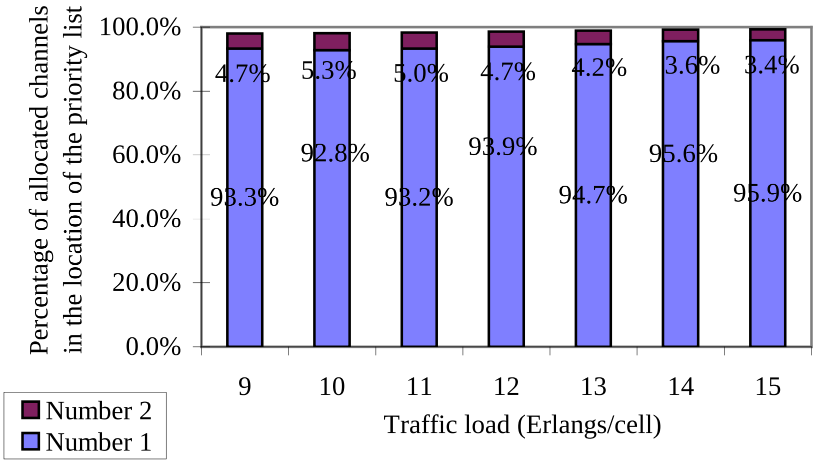


Figure 2.14 Percentage of allocated channels in the sequence of the priority lists. Number 1 denotes the highest priority channels and number 2 denotes the next highest priority channels.

Comparing the performance of our algorithm with the first available (FA) based algorithm [15] which does not have any priority channels for the power searching and the power is just searched from a free channel randomly chosen. If a satisfying SIR value is not found in the chosen channel, another free channel is randomly chosen to perform the same power searching procedure until a satisfying SIR value is found or the call is blocked. We have simulated the latter case at 15 Erlangs traffic load. Even though the latter algorithm has 0.5% blocking probability and 4.2% dropping probability, the intracell handover rate is 696% and the unperformed handover rate is 66.3%. That means that in average 6.96 intracell handover accesses are caused by an admitted call and almost 34% of successful calls have experienced multiple handover accesses. Obviously, the low blocking probability for the FA based scheme is obtained by intensive call reassignment. However, because of the limitation of processing capacity and signaling load in the radio network, such huge intracell handover rate might not be acceptable and must cause much higher call blocking and dropping probabilities if it is implemented than those given by simulation. In addition, the ratio which the calls are set up on the first chosen channel is 72.8% and the ratio which the calls are set up on the second chosen channel is 10.2%. It means that the FA based scheme has a longer call setup time comparing our proposed scheme. Hence, the intracell handover rate should be a factor to consider when evaluating the performance of an algorithm.

2.4 Chapter Summary

In this chapter, two new dynamic channel allocation algorithms have been proposed. In the proposed DCA algorithms, the effect of the channel allocation on existing calls is considered with the call outage rate or a cost function. The effect of the time advance error for TDMA system is not considered in the work.

In the first proposed algorithm, called the Minimum Call Outage (MCO) algorithm, a method of estimating the average variation of on-going calls' SIR due to the assignment of a coming call has been developed. The MCO scheme minimizes the call outage rate of the existing calls when assigning a channel to a new call. The MCO scheme improves the capacity or QoS performance compared with the First Available and Maximum SIR schemes. However, the implementation of the MCO scheme is limited by the estimation error of the call outage rate.

In the second proposed algorithm, a cost function has been introduced to estimate the cost of a channel. By comparing the performance of the (pure) cost-function based DCA scheme with that of FA scheme and MSIR scheme we find that the cost-function can be introduced for channel pre-selection to reduce the call dropping and handover rate and to speed up the call set-up. Therefore, we have presented a distributed channel access algorithm combining the channel assignment and power assignment. The cost function is used for channel pre-selection, and the assigned power is probed on the pre-selected channel. This algorithm greatly increases the capacity compared with the FCA. It does not cause a high intracell handover rate and performs with a short call setup time.

3. GPRS IMPACT ON THE QoS of GSM VOICE SERVICES

3.1 Introduction

General packet radio service (GPRS) is designed for transmitting packet data and is overlaid on GSM system. It is supposed to take its radio resource from the pool of channels unused by GSM voice services. The physical channels unused by voice services are allocated dynamically to the GPRS according to the needs for actual packet transfers which is referred to as the “capacity on demand” principle [2]. Considering the radio interface, the QoS of GSM voice services may be affected by GPRS. For example, the interference probability of voice services will increase due to the additional interference contributed by GPRS transmission. Besides probably causing a degradation of the QoS of voice services, the GPRS might affect the system stability because a few users may be multiplexed in a channel and the cochannel interference to the voice users might vary rapidly and dramatically in the time interval from 20 ms to a few seconds depending on the transmitted packet data size.

The introduction of packet data services should have no any effect or negligible effect on the existing voice services. In order to guarantee the quality of service, it may be necessary to allocate dedicated channels for the packet data service, especially for the GPRS with multiple applications and multiple class services. However, such a scheme with dedicated channels will reduce the number of channels provided for voice services and increase the blocking probability to an undesirable level. The dynamic sharing of the channels between voice services and packet data services seems not to have much impact on the capacity of voice services and creates additional capacity for packet data services [46]. However, the system performance may be degraded, e.g., outage probability increase, due to the additional interference contributed by packet data transmission [47-49].

The investigation of the impact on voice services due to overlaying packet data services found in the literature is mainly focused on the American advanced mobile phone standard (AMPS) [46-48]. The discussion in [49] about the effect of GPRS on GSM is only considering a simple case. The power control (with error), discontinuous transmission (DTX) and frequency hopping have not been considered in those discussions. Until now, it seems that there is no any published paper to give an analysis method which can be applied for the discussion of the GPRS impact on the GSM voice services. In this chapter we will present a method to calculate the outage probability of the GSM-GPRS network for both the non-frequency hopping and frequency hopping systems. The GPRS impact on the GSM voice services is discussed by analyzing the outage probability of voice services in a GSM-GPRS network.

3.2 Outage Probability Analysis for GSM-GPRS System

All signals are assumed to have experienced Rayleigh fading with respect to a local mean signal strength, while the local mean signal strength experiences log-normal slow fading based on a mean value, which is determined by the propagation loss law of an inverse *n-th* power of distance. The desired signal and interfering signals, and different individual interfering signals are all statistically mutually independent. The slow fading is assumed to be uncorrelated and the variance is the same for all cells. Voice activity detection is assumed to be used in the system. The channel activity factor is assumed to be 40% for transmitting voice services and 100% for GPRS data services*. The power control algorithm is assumed to compensate the pathloss and slow fading partly ($0<\alpha<1$) [26-28]. The power control error is considered [29-30] as a lognormal variable with a zero mean and a standard deviation σ_e.

The received power of a desired signal S and the interference power I can be described as:

$$S = Ar_{00}^{-\eta}\xi_{00}Z_{00}P_{T_{00}} \tag{3.1}$$

$$I = \sum_{k=1}^{M}\sum_{i=1}^{N_k} I_{ik} = \sum_{k=1}^{M}\sum_{i=1}^{N_k} Ar_{ik}^{-\eta}\xi_{ik}Z_{ik}P_{T_{ik}}X_{ik} \tag{3.2}$$

where P_T is the transmitted power; ξ is the slow fading variable; Z is the fast fading variable; r is the distance from the transmitter to the receiver; η is the pathloss exponent. X is the channel activity factor; A is a proportional constant. The indexes i and k refer to the i-th mobile in the k-th interfering cell and $i = k = 0$ corresponds to the desired mobile and the desired cell. M and N_k are the number of interfering cells and the number of mobiles in the k-th cell respectively. One should note that, when a channel is allocated to GPRS, it is shared by a few data users. The cochannel interference to the voice users might fluctuate frequently in the time interval from 20 ms to a few seconds depending on the transmitted packet data size, since the locations of those data users could be very different.

The outage probability of a system is defined as the probability that the instantaneous signal power to interference power ratio (S/I) falls below a specified threshold γ and denoted as

$$P_o(outage) = \Pr\left\{S / I < \gamma\right\} \tag{3.3}$$

A method to calculate the outage probability of the GSM-GPRS network for both the non-frequency hopping and the frequency hopping systems has been presented [P4]. This method takes into account the Rayleigh fading, power control error, discontinuous transmission, and frequency hopping.

Non-Frequency Hopping System

According to [P4], the average outage probability for the non-frequency hopping system is:

$$\overline{P}_o(\text{outage}) = 1 - p^{-\frac{1+N_d}{2}} \sum_{l=1}^{n} w_l \Big\{ \prod_{k=1}^{N_v} \Big[1 - p_k + \pi^{-\frac{1}{2}} p_k \sum_{j=1}^{n} w_j \frac{1}{1 + d_k \exp(cx_j - ax_l)} \Big] \cdot \prod_{q=1}^{N_d}\sum_{i=1}^{n} w_i \frac{1}{1 + d_q \exp(cx_i - ax_l)} \Big\} \tag{3.4}$$

where w_i is the weight of the n-point Gauss-Hermite quadrature formula and x_i is the abscissa of the i-th zero of Gauss-Hermite polynomial [31-32]; N_v and N_d are the number of cochannels used by voice users and data users simultaneously in M interfering cells respectively,

$$\sigma_k = \sigma_q = \sqrt{(1+\alpha^2)\sigma_s^2 + \sigma_e^2}\,,\ \sigma_{00} = \sqrt{(1-\alpha)^2\sigma_s^2 + \sigma_e^2}\,, a = \sqrt{2}\sigma_{00}\ln 10/10,$$

$$c = \sqrt{2}\sigma_k \ln 10/10,\ d_k = \gamma(r_{00}^{1-\alpha} r_{0k}^{\alpha} / r_k)^{\eta},\ d_q = \gamma(r_{00}^{1-\alpha} r_{0q}^{\alpha} / r_q)^{\eta},$$

---

* Due to multiplexing, when a channel is used for GPRS, the idle time of this channel could be very small if the number of users requesting service is high enough. For the system planner's point of view, the network situation with a high GPRS load (worst case) should be considered to guarantee the QoS of voice services.

where r_{00} is the distance from the desired mobile to its host base station; r_k (or r_q) is the distance from the desired mobile to the k-th interfering base station of voice (or data) user in downlink, or the distance from the host station to location of the cochannel voice (or data) user in the k-th interfering cell in uplink; r_{0k} (or r_{0q}) refers to distance from location of the cochannel voice (or data) user in the k-th interfering cell its own base station. The outage probability is mainly dependent on the locations of mobiles, the frequency reuse factor and the channels' occupancy by GSM voice services and GPRS.

Frequency Hopping System

For frequency hopping systems, each channel in a cell is occupied by voice or data users with the same probability A_k,

$$A_k = \frac{N_v(k) \cdot V_f + N_d(k)}{N_t \cdot N_{hop}(k)} \tag{3.5}$$

where N_t is the number of time slots per TDMA frame and $N_{hop}(k)$ is the number of distinct frequency carriers in cell k; $N_v(k)$ and $N_d(k)$ are the number of channels using in cell k by voice users and data users respectively; V_f is the activity factor of voice users.

In GSM frequency hopping system, each channel within a cell is orthogonal. Because of the non-uniform call traffic in practical networks, the number of frequency carriers allocated to different cells may not be the same. Therefore, the frequency set used in the desired cell may not be exactly the same as the frequency sets in interfering cells. Let $N_{co}(k)$ be the number of frequency carriers used in both the desired cell and interfering cell k; N_{de} is the total number of frequency carriers used in the desired cell. We define a fractional factor f_k as:

$$f_k = \frac{N_{co}(k)}{N_{de}} \times \frac{N_{co}(k)}{N_{hop}(k)} \tag{3.6}$$

Therefore, the interference probability caused by all users in cell k is $p_k = A_k \cdot f_k$. The average outage probability with frequency hopping system [P4] is

$$\overline{P}_o(\text{outage}) = 1 - \pi^{-\frac{1}{2}} \sum_{l=1}^{n} w_l \{ \sum_{k=1}^{M} \left[1 - p_k + \pi^{-\frac{1}{2}} \frac{p_k}{N_k} \sum_{i=1}^{N_k} \sum_{j=1}^{n} w_j \frac{1}{1 + d_{ik} \exp(cx_j - ax_l)} \right] \} \tag{3.7}$$

where $d_{ik} = \gamma (r_{00}^{1-\alpha} r_{0ik}^{\alpha} / r_{ik})^{\eta}$; r_{00} is the distance from the desired mobile to its host base station; r_{ik} is the distance from the desired mobile to the k-th interfering base station of the i-th mobile in downlink, or the distance from the host base station of the desired mobile to the i-th mobile in the k-th interfering cell in uplink; r_{0ik} refers to distance from mobile i in interference cell k to its own base station; $N_k = N_v(k) + N_d(k)$.

3.3 Numerical Results

In this section, we will give some numerical results of the outage probability of the uplink affected by transmitting GPRS traffic in the GSM radio network resource. The goal of this work is to investigate how the performance of voice services are affected by the introduction of GPRS traffic into existing GSM networks. The main interest is the interference statistics of voice services affected by GPRS. Therefore, we assume that the call blocking probability of voice services is not affected by GPRS by considering a preemptive priority for voice services.

A. System model

An idealized system with equally-sized hexagonal cells and omnidirectional antennas is considered. A central cell which is taken as the cell with desired mobiles has six interfering cells. The propagation loss exponent η and standard deviation σ_s of the slow fading are same for all cells and assumed to be 4 and 8 *dB* respectively. The voice channel activity factor is assumed to be 0.4, and the packet data channel activity factor is 100%. For power control, the algorithm partly compensating the path loss and shadowing is chosen with α = 0.5 [28]. The threshold value of SIR is 10 *dB*.

The number of physical channels are 32 (4×8) available in a cell. It is assumed that 3 channels are reserved for network signaling, thus, only 29 channels are available for traffic in a cell. The GSM network is supposed to be operated at a blocking probability of 0.02 for voice services. For a cell with 29 traffic channels, an average traffic load of 21.04 Erlangs is then supported. The average number of voice calls in the system is $E(n) = \rho(1-P_b) = 21.04\times98\% = 20.62 \approx 21$. This average number of voice calls is considered in our calculations mainly. Based on the average situation with 21 channels used for voice services, a few channels are assumed to be allocated to GPRS and the outage probability is calculated. The system parameters are listed in Table 3.1.

TABLE 3.1. System parameters.

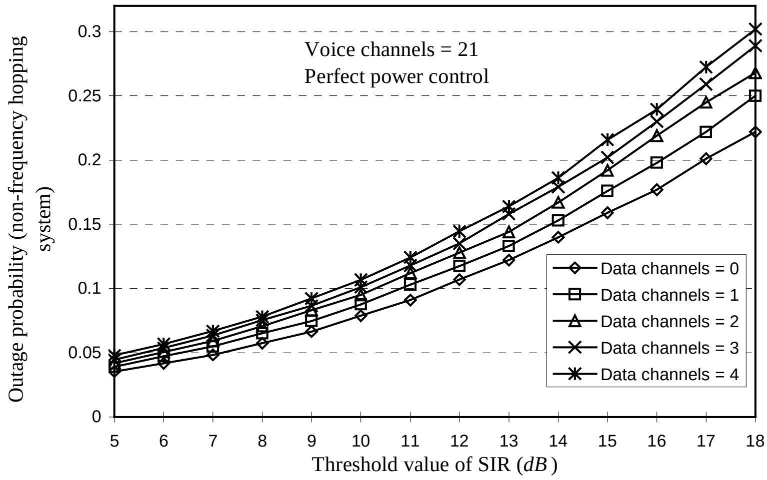
propagation loss exponent	$\eta = 4$
std. deviation of the slow fading	$\sigma_s = 8$ *dB*
voice channel activity factor	40%
GPRS channel activity factor	100%
power control (partly compensation)	$\alpha = 0.5$ [28]
std. deviation of power control error	σ_e=0 and 2 *dB*
SIR threshold value	γ =10 *dB*
total number of traffic channels	29
voice traffic load (P_b=2%)	21.04 Erlangs
channels simultaneously used by voice services	21

The mobile stations are uniformly distributed with an identical number in each cell. The outage probability is calculated through a series of Monte Carlo simulations based on generating a large number of snapshots. In each snapshot locations of users are randomly generated, and a pure random channel allocation algorithm is used to assign channels to users in the central (desired) cell as well as in each interfering cell, according to the considered number of simultaneous users. Due to the random channel allocation, for the non-frequency hopping system there always exist some "good" channels with less cochannel users and "bad" channels with more cochannel users, but for the frequency hopping system the quality of every channel is the same because of the interference diversity. From the system planner's point of view, the system should guarantee the quality of the "worst" channel with the largest number of cochannel users. Therefore, for the non-frequency hopping system the outage probability of each snapshot is calculated from the "worst" channel. In a simulation, 10000 snapshots are generated and the outage probability of each snapshot is calculated from (3.4) or (3.7). The distribution of the outage probability in a given cell depends on locations of the mobile stations in this cell and in its interfering cells. Instead of giving a cumulative distribution function (CDF) of the outage probability, we only show the 0.9 percentile value, called 90%

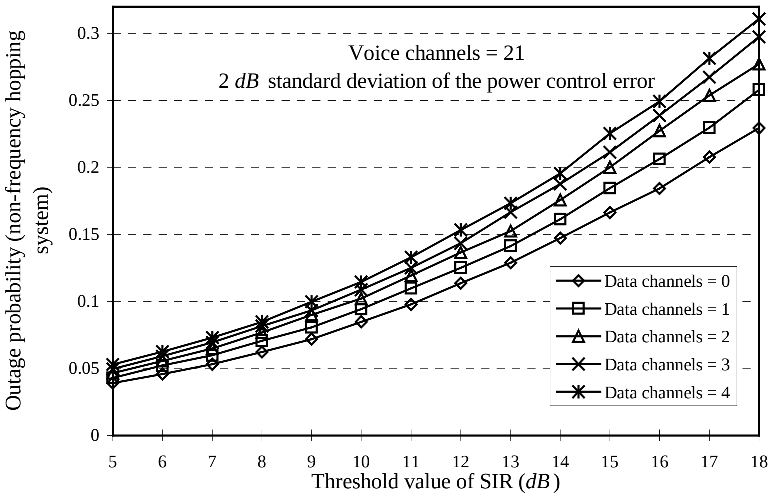
worst case outage probability. The 90% worst case outage probability is obtained by sorting those 10000 snapshots' values in increasing order and choosing the 9000th value.

B. Results

For an average situation of 21 traffic channels used by voice services in each cell simultaneously, Fig. 3.1 shows the 90% worst case outage probability of the non-frequency hopping system (with a reuse factor of 7) as function of the SIR value and with the number of channels used for GPRS as a parameter. Figure 3.1a and 3.1b correspond to perfect power control and 2 *dB* standard deviation of power control error in the system respectively. As seen from Fig. 3.1, with the 10 *dB* SIR threshold value, the outage probability increases by about 5%-10% whenever the number of channels used for GPRS increases by one. In addition, comparing Fig. 3.1a-b, about 0.5 *dB* degradation in average is found for the system with imperfect power control.



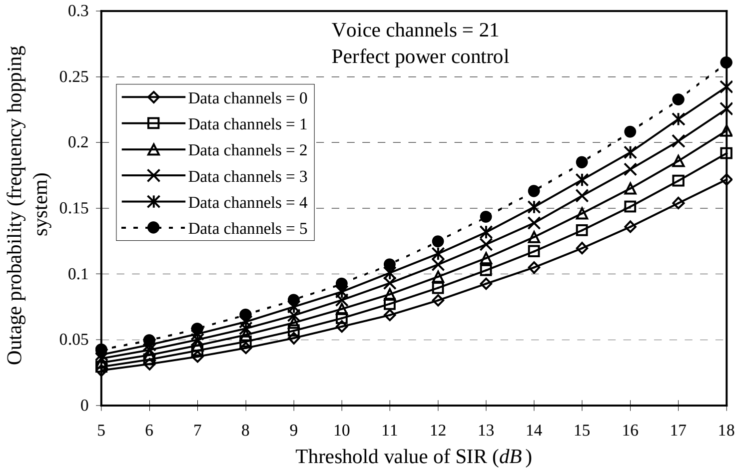
(a)



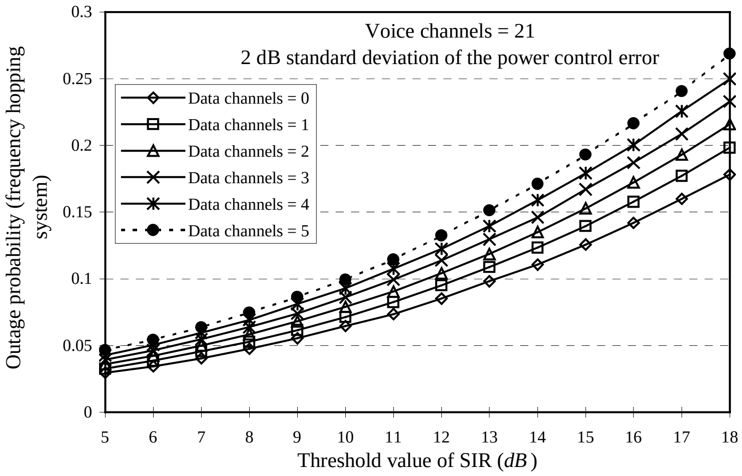
(b)

Figure 3.1. Outage probability (90% worse case) of non-frequency hopping system vs. SIR with GPRS channel occupancy as a parameter for perfect power control and imperfect power control respectively (21 channels used by voice services, a reuse factor of 7).

Similarly, Fig. 3.2a-b show the 90% worst case outage probability of the frequency hopping system (with a reuse factor of 7) as function of the SIR value and with the number of channels used for GPRS as a parameter. Fig. 3.2a and 3.2b correspond to perfect power control, and 2 *dB* standard deviation of power control error in the system respectively. The performance of the frequency hopping system in the outage probability is much better than that of the non-frequency hopping system. However, its characteristic of the outage probability affected by GPRS is similar to that of the non-frequency hopping system.



(a)



(b)

Figure 3.2. Outage probability (90% worst case) of frequency hopping system vs. SIR with GPRS channel occupancy as a parameter for perfect power control and imperfect power control respectively (21 channels used by voice services, a reuse factor of 7).

The power control error can cause the system has a higher outage probability. From Fig. 3.1 and 3.2, we find that when a high number of channels are used for GPRS, the increase percentage of the outage probability is higher than that when a low number of channels are used for GPRS. Therefore, the power control error has more impact on the system performance when more channels are allocated to GPRS.

Figure 3.3 shows the 90% worst case outage probability of the non-frequency hopping system (with a reuse factor of 7) as function of the number of channels occupied by voice services and GPRS. A SIR threshold value of 10 *dB* and perfect power control are assumed in the simulation. For the same number of channels used by GPRS, the outage probability does not vary very much from high channel occupancy to low channel occupancy of voice services. The main reason is that the outage probability is calculated with the worst channel in each snapshot here. This result implies that for the non-frequency hopping system the channels provided to GPRS are not much different between high channel occupancy and low channel occupancy of voice services. It must be aware that statement concluded from Fig. 3.3 is not always true when the intracell handover is considered. When the channel occupancy is low, if GPRS jeopardizes some “bad” calls, they can be switched to other free channels with proper intracell handover. Consequently, the system could provide more resources to GPRS than those shown in Fig. 3.3 when the channel occupancy of voice services is low.

Figure 3.4 shows the 90% worst case outage probability of the frequency hopping (with a reuse factor of 7) as function of the number of channels occupied by voice services and GPRS. For the same number of channels used by GPRS, the outage probability increases as the channel occupancy of voice services gets higher. Therefore, the quality of voice services may not be affected by the introduction of GPRS into GSM with proper admission control.

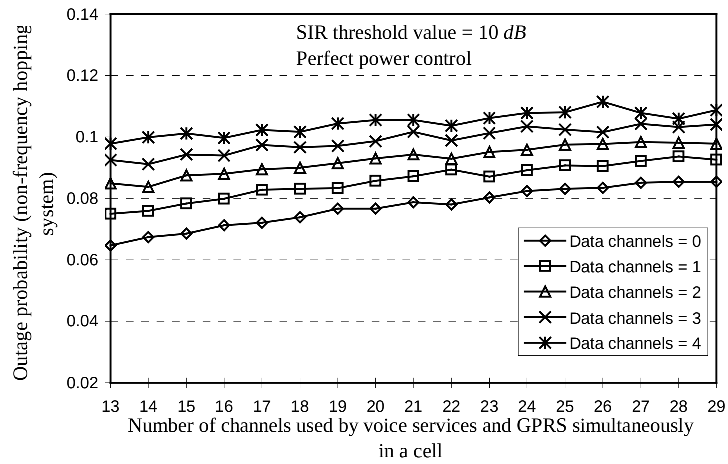


Figure 3.3. Outage probability (90% worst case) of non-frequency hopping system vs. different number of channels occupied by voice services and GPRS (SIR threshold value = 10 *dB*, a reuse factor of 7).

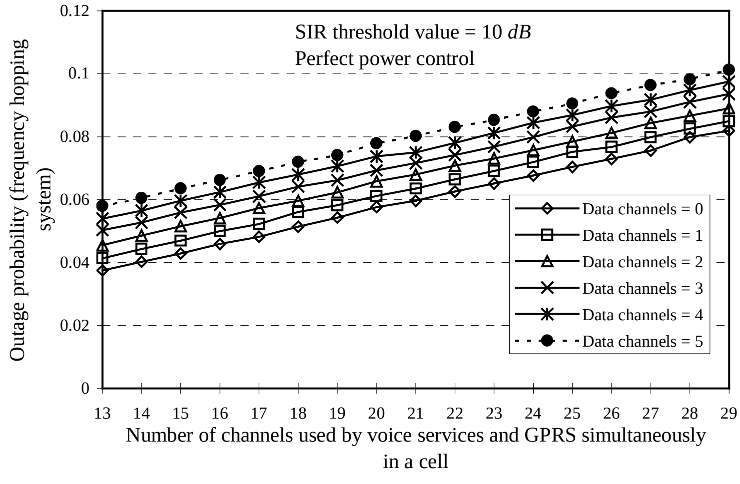


Figure 3.4. Outage probability (90% worst case) of frequency hopping system vs. different number of channels occupied by voice services and GPRS (SIR threshold value = 10 *dB*, a reuse factor of 7).

Cell coverage is normally determined by the received signal strength, however, the received value of the signal to interference ratio (SIR) is more relevant to describe the service area of a cell. Therefore, a parameter, the cell service area, is defined as the area over which a specified outage probability limit is achieved. In order to investigate the cell service area of existing voice services affected by GPRS, we simulate the outage probability as a function of the normalized radius (*r*/*R*, *R* is the cell radius) in the desired cell. In each simulation, the location of the mobile station in the desired cell is restricted to a circle with a radius *r*, and locations of mobiles in interfering cells are randomly generated in those cells. 21 simultaneous voice users are assumed in each cell and a SIR threshold value of 10 *dB* is used. Perfect power control is assumed in the system. Figure 3.5 shows the 90% worst case outage probability of the non-frequency hopping system as a function of normalized radius (*r*/*R*). With the same outage probability limit, Figure 3.5 shows that the cell service area decreases by about 10%~20% whenever the number of channels used for GPRS increases by one. As more channels are allocated to GPRS, the cell service area decreases dramatically.

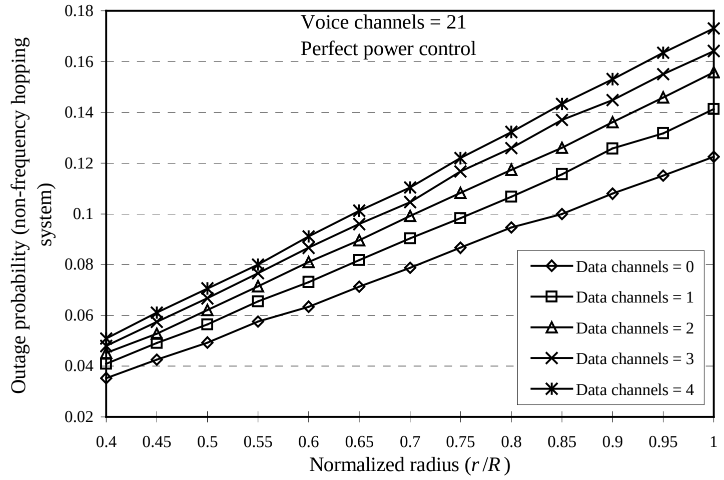


Figure 3.5. The 90% worst case outage probability of the non-frequency hopping system vs. normalized radius (*r*/*R*).

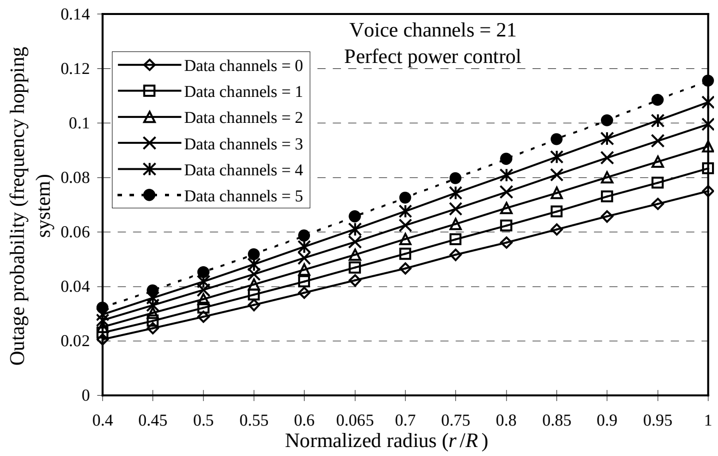


Figure 3.6. The 90% worst case outage probability of the frequency hopping system vs. normalized radius (*r*/*R*).

Figure 3.6 shows the 90% worst case outage probability of the frequency hopping system distributed with normalized radius *r*/*R* respectively. Though the frequency hopping system has better performance in the cell service area than the non-frequency hopping system, the characteristics of the cell service area shrinking as the introduction of GPRS is similar to that of the non-frequency system. The cell service area decreases by about 10-15% whenever the number of channels used for GPRS increases by one.

Since GPRS causes a decrease of the cell service area, consequently, the dropping rate of the intercell handover for the GSM network may increase due to the introduction of GPRS.

The effects of GPRS on the outage probability for systems with different frequency reuse factors have been further discussed in publication 5 [P_5]. According to [P_5], the simulation shows that the outage probability increase of the network with a small frequency reuse factor caused by GPRS is higher than that of the network with a large frequency reuse factor. Therefore, the GPRS affects the QoS of voice services of the network with small size reuse factor more. GPRS will reduce the cell service area, but the reduction percentage of the cell service area for the system with small reuse factor is higher than that for the system with large reuse factor.

C. *Stability of simulation results*

In this chapter, the outage probability of each single point in those Figures is the 0.9 percentile values, called 90% worst case outage probability, which is obtained from a series of Monte Carlo simulation with 10000 shapshots. More specifically, the 90% worst case outage probability is obtained by sorting those 10000 snapshots' values in increasing order and choosing the 9000th value. The 90% worst case outage probability distributed with the number of shapshots can be approximately considered as a normal distribution, therefore we can apply the method in [67] to estimate the confidence intervals for the results given in this chapter. Let parameter Y is estimated from the observation vector $X=[x_1, x_2, \ldots, x_m, \ldots, x_n]$. Normally, the sample mean $\mu=\sum x_i/n$ is used to estimate parameter Y if x_i, $i=1,..n$, distributes with a normal distribution. The factor of the confidence interval is [67] $\Delta\mu=z\cdot s/n^{1/2}$, where z is percentile of the standard normal density and s is the standard deviation.

In this chapter, we use one of the sample x_m to be the approximate estimation of Y (it can only be true when the difference between the observation value x_i is very small). The corresponding factor of the confidence interval is $\Delta=\mu-x_m\pm z\cdot s/n^{1/2}$.

One of curves in Figure 3.3 is used for the numerical test. The 0.9 percentile outage probability of the case with four GPRS channels is calculated respectively with 10000, 20000, 30000, …, 100000 shapshots. The comparison between the outage probability with 10000 shapshots ($P_{outage}(10000)$) and the mean with 99% confidence interval (which are estimated from the ten sample values of $P_{outage}(10000)$, $P_{outage}(20000)$, …, $P_{outage}(100000)$) is shown in Figure 3.7.

Figure 3.7 shows that most of the outage probability values from 10000 shapshots are out of the 99% confidence interval of the mean. However, the difference between the outage probability values from 10000 shapshots and mean plus the 99% confidence interval is very small (below 0.002) and can be ignore in the practical application. That is the maximum confidence interval is about 2% of the outage probability with 10000 shapshots for this case. Therefore, the outage probability from 10000 shapshots shown in this chapter are stable enough for the practical application.

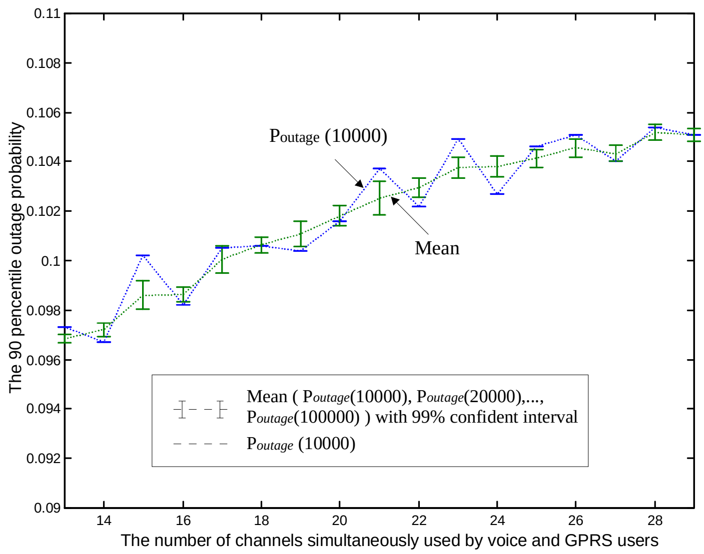


Figure 3.7. Comparison between the outage probability with 10000 shapshots ($P_{outage}(10000)$) and the mean with the 99% confidence interval (which are estimated from the ten sample values of $P_{outage}(10000)$, $P_{outage}(20000)$, …, $P_{outage}(100000)$ $P_{outage}(10000)$, $P_{outage}(20000)$, …, $P_{outage}(100000)$).

3.4 Chapter Summary

GPRS is designed for transmission of packet data and supposed to take its radio resources from the pool of channels unused by GSM voice services, in order to increase the effective capacity of the digital cellular system. However, introducing GPRS into the existing GSM network is at the expense of the degradation in either voice quality or voice capacity. In this chapter, an analysis method has presented to calculate the outage probability of the GSM-GPRS network for both the non-frequency hopping and frequency hopping systems. This method takes into account the fast (Rayleigh) fading, power control (with error), discontinuous transmission, and frequency hopping (if applied), and can be utilized to the evaluation of network performance and network planning for GSM-GPRS cellular systems.

The effects on the quality of voice services due to the introduction of GPRS into GSM network are evaluated by calculating the outage probability. Obviously, GPRS increases the outage probability of existing GSM voice services. With the 10 *dB* SIR threshold value, the outage probability increases by about 5%-10% whenever the number of channels used for GPRS increases by one. The power control error has more impact on the system performance when more channels are allocated to GPRS. The system with frequency hopping may provide more channels to GPRS at the low channel occupancy, however, for non-frequency hopping system, some voice users may need to have multiple intracell handovers in order to provide more channels to GPRS at the low channel occupancy. The cell service area is decreased by about 10%~20% whenever the number of channels used for GPRS increases by one. As more channels are provided to GPRS the cell service area decreases dramatically. Consequently, the dropping rate of the intercell handover for GSM network may increase due to the introduction of GPRS.

The simulation shows that the outage probability increase of the network with a small frequency reuse factor caused by GPRS is higher than that of the network with a large frequency reuse factor. Therefore, the GPRS affects more on the QoS of voice services of the network with small size reuse factor. GPRS will reduce the cell service area, but the reduction percentage of the cell service area for the system with small reuse factor is higher than that for the system with large reuse factor.

Therefore, channels unused by voice services might not all be used for carrying GPRS traffic. The number of channels allocated to GPRS depends on the difference between the outage level of the existing GSM network and the maximum acceptable level. Therefore, in order not to damage the GSM voice services the remaining capacity of an existing network must be correctly evaluated.

4. ASPECTS FOR GPRS RADIO NETWORK PLANNING

4.1 Introduction

In chapter 3, we have discussed the possible effects of GPRS on the QoS of GSM voice services and presented a method to calculate the remaining capacity of an existing network considering the interference probability constraint. In this chapter we will discuss the GPRS performance and principles of GPRS network planning.

The throughput and delay performance of GPRS mainly depends on the amount of radio resources allocated and the system interference level which determines the coding scheme adopted. Earlier studies of GPRS performance found in the literature [33-36] are simulated with a fixed number of channels used for data transmission. However, the number of channels available to GPRS is a random variable depending on how many channels are used by voice services and the interference level of the network. The service statistics is a movable boundary Markov process [37-39]. The multi-rate service process sharing the same resource can be modeled by a multi-dimensional Markov chain. For a small system (e.g., a few channels) the exact solution of the system performance could be approached by matrix methods [40] and moment generating functions [39]. However, for a large system, since those methods are computationally intractable, the approximation method or simulation may be the only ways to investigate the system performance.

In this chapter, we will investigate the GPRS downlink performance in a dynamically varying number of resources. The protocols [2] in the uplink packet transmission is different from those in the downlink transmission, where there is no any contention phase and the network initiates the packet transfer to an MS in the Ready state. The medium access control (MAC) is operated to provide efficient multiplexing in order to get GPRS users to share a common transmission medium on both uplink and downlink. On the downlink the multiplexing is controlled by a scheduling mechanism. Here, the first in first out (FIFO) scheduling principle is assumed. We firstly discuss the single-cell case without considering the network interference level. An approximation method for the single-slot service is discussed. The analysis method may be useful for the preliminary designing of the system “buffer size”. The multiple-slot services are discussed by simulations. Secondly we evaluate the GPRS performance in multiple cell network considering the network interference level. By this work new results about the GPRS performance are provided and will be useful for network planning and operation. In the latter part of this chapter the guidelines for GPRS network planning are given.

4.2 GPRS Performance Evaluations

4.2.1 Single-Cell Case

For a system with m physical channels, m_v channels are shared by voice and GPRS services and m_d channels ($m_d \geq 0$) are dedicated to GPRS (Fig. 4.1). In the pool of m_v channels, when channels are not used by voice services, those channels may be used for GPRS transmission. The voice services own preemptive priority over GPRS, i.e., whenever channels used for the GPRS service are needed by voice services, the GPRS transmissions in some of those borrowed channels are stopped and the channels are released to voice services. The GPRS users whose transmissions were interrupted will get service when resources are available again.

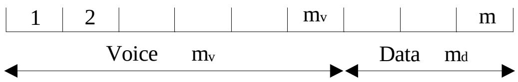


Figure 4.1. Illustration of the radio resource structure allocated to GSM voice services and GPRS. The number of m_v channels is shared with voice and GPRS services and the number of m_d channels is dedicated to GPRS.

Assume that voice users arrival is a Poisson process with a rate of λ_v and the call service time is exponentially distributed with a mean of $1/\mu_v$. All GPRS users share the physical channels unused by the voice services. The arrival of GPRS data users is assumed to be a Poisson process with rate λ_d and the service time is exponentially distributed with a mean of $1/\mu_d$. The maximum number of data users accepted into the system (in service and in queue) is N. GPRS users are served according to the first in first out (FIFO) principle. The arriving GPRS user is allowed to transmit data if a sufficient number of free channels is available; otherwise it is queued or blocked.

For such a voice/packed data mixed system, the blocking probability of voice services is not affected by GPRS, however, the resource provided to GPRS is a random variable which depends on the channel occupancy state of the voice services. The service statistics is a movable boundary Markov process. The exact analysis solution for such a system could be unsolvable especially as multiple classes of users are supported in GPRS. Therefore, here we use an approximation method for single-slot service and simulations for multiple-slot services to investigate the GPRS performance.

A. An Approximation Method for Performance Evaluation

The voice services are independent of GPRS. Because GPRS is mainly designed to transmit intermittent and burst data, the service time of GPRS is rather smaller than that of voice services. For such traffic characteristics, the steady state behavior of the voice/data mixed system could be decomposed [38-39] into long term (voice) and short term (data) behavior. In order to be accurate for the decomposition approximation it is necessary that each group of states should achieve equilibrium in isolation. The essential of this decomposition technique is to use the probability distribution of voice services to describe the interaction of voice services with GPRS. Thus, the GPRS performance with the dynamically varying resource is obtained by combining this distribution with its performance with a fixed resource.

For the voice services, the probability of n users in service is [53]

$$G_n = G_0 (\frac{\lambda_v}{\mu_v})^n \frac{1}{n!}, \quad n = 1, 2, ..., m_v, \text{ where } G_0 = \left[\sum_{n=0}^{m_v} (\frac{\lambda_v}{\mu_v})^n \frac{1}{n!} \right]^{-1} \tag{4.1}$$

The channels unused by the voice services may be used for the data services. The probability of x channels available for the data services is equal to the probability of m_v-x channels are used by voice services and is obtained from (4.1):

$$g(x) = G_0 (\frac{\lambda_v}{\mu_v})^{m_v - x} \frac{1}{(m_v - x)!}, \quad x = 1, 2, ..., m_v \tag{4.2}$$

Due to the interference constraint, in chapter 3 we have concluded that channels unused by voice services may not all be utilized by GPRS. It is assumed that the maximum number of channels (not including the dedicated channels) allocated to GPRS is L.

For the transmission of the single slot service in a fixed number of C channels, the average queueing time can be obtained from the $M/M/C/N$ queueing system, where N is the maximum number of users in system (in service and in queue). The steady-state probability p_n is [53]:

$$p_n = \begin{cases} p_0 \frac{\rho^n}{n!}, & n < C \\ p_0 \frac{\rho^n}{C! C^{n-C}}, & C \le n \le N \end{cases} \tag{4.3}$$

where n is the number of active users in the system, $\rho = \lambda_d/\mu_d$ and

$$p_0 = \left[\ 1 + \sum_{n=1}^{C-1} \frac{\rho^n}{n!} + \sum_{n=C}^{N} \frac{\rho^n}{C!} \frac{1}{C^{n-C}}\ \right]^{-1} = \left[\ \sum_{n=0}^{C-1} \frac{\rho^n}{n!} + \frac{\rho^C (1 - (\frac{\rho}{C})^{N-C+1})}{C!(1 - \rho/C)}\ \right]^{-1}$$

A new arrival is accepted into the system only if the number of GPRS users in the system is below the maximum accepted number N. Otherwise, the new arrival is blocked. The blocking probability is

$$P_N(C) = p_0 \frac{\rho^N}{C! C^{N-C}} \tag{4.4}$$

The average number of users in the system is obtained as

$$W(C) = \sum_{n=1}^{N} n p_n = p_0 (\sum_{n=1}^{C} \frac{\rho^n}{(n-1)!} + \frac{C^C}{C!} \sum_{n=C+1}^{N} \frac{n \rho^n}{C^n})$$

$$= p_0 \{ \sum_{n=1}^{C} \frac{\rho^n}{(n-1)!} + \frac{C^C}{C!} \cdot \frac{(C+1)(\frac{\rho}{C})^{C+1} - C(\frac{\rho}{C})^{C+2} - (N+1)(\frac{\rho}{C})^{N+1} + N(\frac{\rho}{C})^{N+2}}{(1 - \frac{\rho}{C})^2} \} \tag{4.5}$$

Combining (4.2) with (4.4) and (4.5), the average blocking probability, throughput and average queueing time of single slot GPRS in a dynamically varying resource are obtained as following expressions respectively:

$$\overline{P}_B = \sum_{x=0}^{L-1} g(x) P_N(x + m_d) + P_N(L + m_d) \sum_{x=L}^{m_v} g(x), \quad L \le m_v \tag{4.6}$$

$$\overline{TH} = \lambda_d (1 - \overline{P}_B) \tag{4.7}$$

$$\overline{T} = \frac{1}{\lambda_d (1 - \overline{P}_B)} (\sum_{x=0}^{L-1} g(x) W(m_d + x) + W(m_d + L) \sum_{x=L}^{m_v} g(x)) - \frac{1}{\mu_d}, \quad L \le m_v \tag{4.8}$$

Equations (4.6-8) are useful to design the “buffer size”. For multiple class (rate) GPRS services, it is very difficult to analyze the state distribution probability of queueing system even in a fix number of transmission channels because of the large size of the state space. Here, we only use simulation to evaluate the GPRS performance for multiple-slot services.

B. Numerical Results

In order to know the accuracy of the approximation, the numerical results are compared with the simulation results. In the numerical calculations and simulations, 4 carriers, i.e., 4×8=32 channels in a cell are assumed, from which 1 channel is reserved for GPRS data and 31 channels are shared by voice services and GPRS. In the simulations, when a new voice call arrives, if no free channel is available and the number of voice calls in service is below 31, one of GPRS calls is interrupted in order to allocate one channel to the new voice call. When resources are available, the interrupted GPRS calls have higher priority to be allocated resource than the queueing calls. The average interruption time and probability of interruption are simulated. The average service time of voice services is exponentially distributed with a mean of 180 s. The traffic load of voice services is 22.83 Erlang corresponding to 2% blocking probability for 31 channels. The maximum number of GPRS users allowed into network is 40. Coding scheme CS-2 corresponding with a transmission rate of 13.4 kb/s is assumed to be used in the simulation. The GPRS message size is exponentially distributed with means of 2×13.4 kb, 5×13.4 kb, 10×13.4 kb respectively, corresponding to the mean service time ($1/\mu$) of 2s, 5s and 10s with single slot transmission. The approximation results of single slot service are calculated from Eq. (4.6) and (4.8).

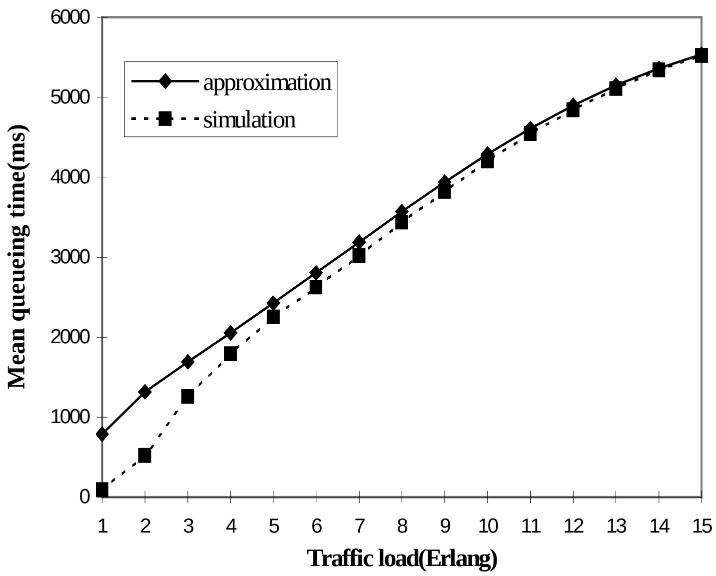


Fig. 4.2. The mean queueing time of single-slot service for the average service time ($1/\mu$) of 2 s.

Fig. 4.2-4.3 show the mean queueing time of single-slot service distributed to the traffic load for the average service time ($1/\mu$) of 2s, 5s and 10s respectively. The average interruption time is included into the mean queueing time here as well as in the latter part of this chapter. Comparing the approximation results with simulation results, we find that the approximation method could be used for evaluating the GPRS performance for high load traffic when the average service time of voice services is much longer than that of GPRS, e.g., μ_d/μ_v>100. For the characteristic of GPRS traffic, this requirement can be satisfied. As the average service time of GPRS increases, the error becomes larger. The reason of the error is that the approximation method over-estimates the blocking probability [P_6]. According to the paper [38], the error of this approximation decreases as the number of total channels increases and increases as the "queueing size" increases. Therefore equations (4.6-4.8) are only useful in the preliminary designing work of the "buffer size" *N*.

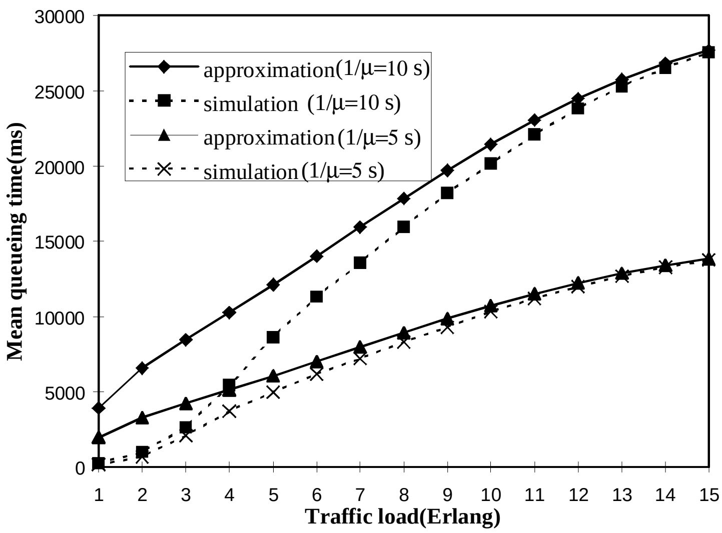


Fig. 4.3. The mean queueing time of single-slot service for the average service time ($1/\mu$) of 5 s and 10 s respectively.

Multiple-Slot Services Performance

In the simulations of multislot services, the GPRS traffics in Table 4.1 are used. Two schemes are used for resource allocation to GPRS calls.

- *scheme*-1 (fixed multislot scheme): On the arrival of a multislot (two or three slots) call from the queue, if the available channels are not enough for the requirement of a call, it still stays in queue until its resource requirement is fulfilled.
- *scheme*-2 (adaptive multislot scheme): On the arrival of a multislot (two or three slots) call from the queue, if the available channels are not enough for the requirement of a call, the available channels are allocated to the call.

TABLE 4.1: GPRS traffic

Traffic-1	All arrival traffic requires single-slot service.
Traffic-2	In the arrival traffic, 70%, 20%, 10% of calls require single-slot, two-slots and three-slots services respectively, i.e., $\lambda=\lambda_1/0.7=\lambda_2/0.2=\lambda_3/0.1$.
Traffic-3	In the arrival traffic, 50%, 30%, 20% of calls require single-slot, two-slots and three-slots services respectively, i.e., $\lambda=\lambda_1/0.5=\lambda_2/0.3=\lambda_3/0.2$.

Fig. 4.4-4.5 show the mean queueing time (average for all served calls) and blocking probability with resource allocation *scheme*-1 and *scheme*-2 distributed with the call arrival rate for the average message size of 2×13.4 kb. The performance of scheme-2 is much better than that of scheme-1. From those figures, we find that for traffics with the same arrival rate (λ) the performance of the adaptive multislot allocation scheme (*scheme*-2) in the mean queueing time and blocking probability does not depend on the composition of the individual

arrival rate (λ_k). In addition, the performance of the adaptive multislot allocation is almost same as that of a system which only provides single-slot service. That implies that, when the adaptive multislot allocation scheme is used, the GPRS performance can be evaluated with the performance of the single-slot allocation. Therefore, the preliminary design of the "buffer size" *N* for adaptive multislot service can used the Equations (4.6-4.8).

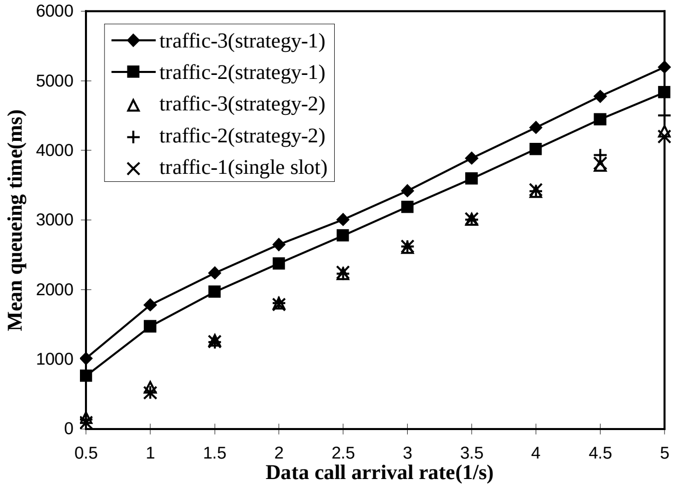


Fig. 4.4. The mean queueing time (average for all served calls) with resource allocation *scheme*-1 and *scheme*-2 for the average message size of 2×13.4 kbits.

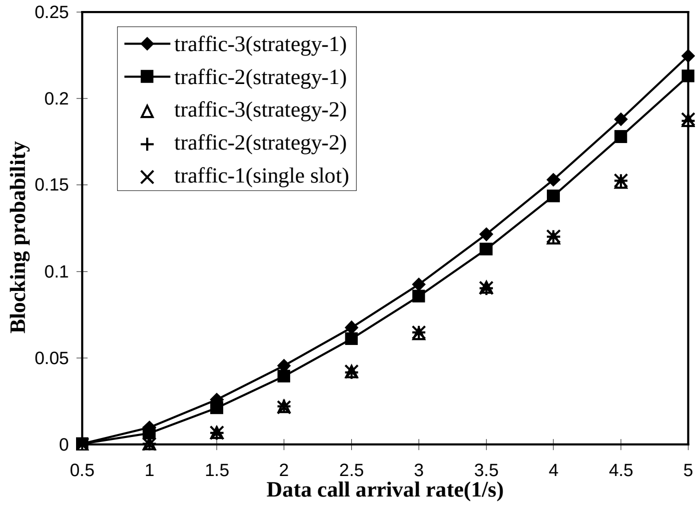


Fig. 4.5. The average blocking probability with resource allocation *scheme*-1 and *scheme*-2 for the average message size of 2×13.4 kbits.

The simulations of the multislot services also show that the interruption probability of GPRS calls depends on the average message size more strongly than on the traffic load. In the publication 6 [P_6], it has been shown that the served rate of the multi-slot services decreases as the increase of both call arrivals and the average packet size. Therefore, the adaptive multislot allocation not only can reduce the blocking probability and delay caused by fixed multislot services, but also can adapt to the GPRS traffic to provide the optimal performance.

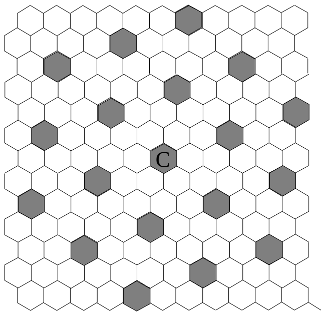


Figure 4.6. Simulation network.

4.2.2 Multi-Cell Network Case

The GPRS downlink performance shown in follows is from the simulation of a system with equally-sized hexagonal cells (a reuse factor of 7) and omnidirectional antennas. To reduce the border effect we simulate in 19 cochannel cells (Fig. 4.6) and only the performance in center cell *C* is shown here. The voice traffic load is 21.04 Erlangs corresponding to 2% blocking probability for 29 traffic channels (four carriers while three channels for signaling). The system parameters given in table 4.2 are from the reference [41]. A parameter of $\alpha = 0.5$ is chosen in power control which partly compensates the path loss and shadowing [28].

TABLE 4.2. Simulation parameters.

Pathloss (GSM900)	L=132.8+38log(*d*), *d* in km
BTS power (max.)	43 *dBm*
BTS power (min.)	13 *dBm*
Cell radius	2 km
Std. dev. of slow fading	σ_s =8 *dB*
Range of slow fading	± 15 *dB*
Noise level	-140 *dBm*
Power control parameter	$\alpha = 0.5$
SIR threshold (voice user)	γ =12 *dB*
Date error rate of a packet	0.05
Max. num. of users in queue	60

TABLE 4.3. The required SIR values for GPRS coding schemes.

Coding scheme	Code rate	Data rate (kbps)	Req. SIR without FH	Req. SIR with FH
CS-1	1/2	9.05	13 dB	9 dB
CS-2	≈2/3	13.4	15 dB	13 dB
CS-3	≈3/4	15.6	16 dB	15 dB
CS-4	1	21.4	19 dB	23 dB

Four coding schemes are used for GPRS in the simulation. The SIR threshold values for those coding schemes are shown in Table 4.3 [42]. In chapter 3 we have concluded that channels unused by voice services might not all be used for carrying GPRS traffic. The number of channels allocated to GPRS depends on the difference between the outage level of the existing GSM network and the maximum acceptable level. Here we assume the following policy to be used for GPRS resource allocation:

$$E = \begin{cases} Y & Y \le 5 \\ \max\{5,\ \mathrm{int}(75\%Y)\} & Y > 5 \end{cases} \tag{4.9}$$

where E is the number of channels which can be allocated to GPRS; Y is the number of channels unused by voice services. When a voice user is arriving and no free channel is available, one of channels used by GPRS must be released to the voice user. The multiple rate services consider the single-slot, two-slots and three-slots services, while the single rate service refers the case where only a single slot service is provided by the network. We use the adaptive timeslot allocation scheme for GPRS resource allocation, which implies that when the number of available channels is not large enough for the requirement of a multislot call, the available channels will be allocated to the call. For all the arriving GPRS calls, we assume that the distributions of single-slot, two-slot and three-slot services are 50%, 30%, 20% respectively (it has been shown in section 4.2.1 that the network performance in the mean queueing time and blocking probability does not depend on the amount of percentage of each group traffic for the adaptive timeslot allocation scheme). FUNET, Mobitex, and Railway traffic models [52], are used in the simulation. The packet size X (in kbytes) of the FUNET model conforms to a truncated Cauchy distribution, i.e.,

$$X = \tan(\pi(U - 0.5)) + 0.8,$$

where $0<X<10$ kbytes; U is a random variable uniformly distributed in (0, 1). The Mobitex model is uniformly distributed within the range of 15~45 bytes in uplink and 58~172 bytes in downlink respectively. The packet size X of the Railway model is a truncated exponential distribution, i.e.,

$$X = -170\log(U), \ 0<X<1000 \text{ bytes}.$$

In order to remove the chaotic of the simulation result we do not use traffic with packet size below 15 bytes in FUNET and Railway models.

In the downlink packet transfer, because the network initiates the transmission without any contention, and the multiplexing is controlled by a scheduling mechanism, the throughput is limited not only by available resources but also by the queueing length in network. The queueing length of GPRS users is assumed to be 60. The packet error rate is assumed as 0.05*. For a message with a size of x and an error rate of r, the total required transmitted data volume is $x/(1-r)$ for infinite retransmission with the selective ARQ protocol. As an approximation, we use the total required transmitted data volume model as the retransmission consideration.

We give following parameters, mean throughput, mean normalized delay and mean queueing time to show the capacity of a network. The mean throughput is the amount of data per second that has been successfully transmitted over the air interface. The queueing time is the time that a call has waited in the queue. The normalized delay is the time including the transmitted time of a message, the queueing time and the interrupted time due to releasing its channel to a voice user, normalized by the message size.

Figure 4.7 shows the mean throughput of the multi-rate services in downlink. The downlink throughput performance is different from that of the uplink [35-36] and does not exhibit a decrease as more traffic is arrival because there is no contention phase in downlink packet

* The required SIR values in Table 4.3 corresponding to a BLER (block error rate) value not more than 10% for systems with and without frequency hopping (FH).

transfer. In addition, the difference in the mean throughput between the multiple-rate services and the single-rate service is very small as shown by the simulation results.

Figures 4.8 and 4.9 show the mean queueing time of the multi-rate services and single-rate service for the three traffic models respectively. The mean queueing time of the FUNET model is much higher than that of the two other models because the average packet size (about 1.73 kbytes) of the FUNET model is much higher that of the Railway and Mobitex models (0.178 kbytes and 0.112 kbytes respectively). There is not much difference (below 100 ms) in the queueing time between the multi-rate services and the single-rate service. For the Railway and Mobitex models, the queueing time of the single-rate service is lower than (not more than 70 ms) that of the multi-rate services when the traffic load increases. That implies that the resource of the multiple-slot allocation can not be efficiently used by the traffic. As a result, the multiple-slot allocation causes more calls in queue when the traffic load increases.

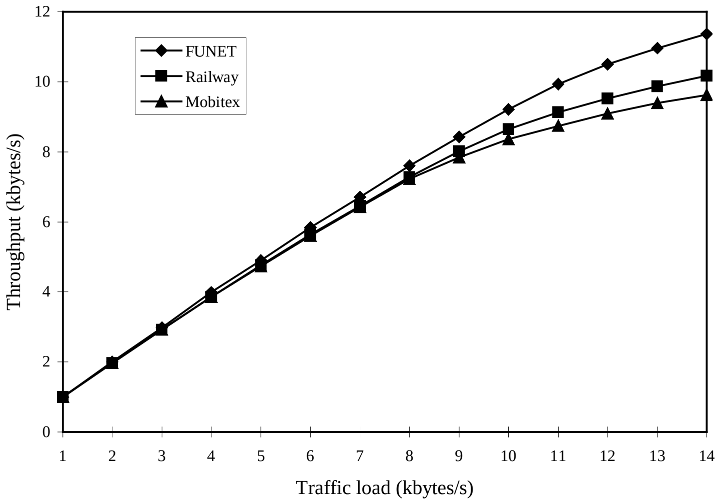


Figure 4.7. The mean downlink throughput of multi-rate services.

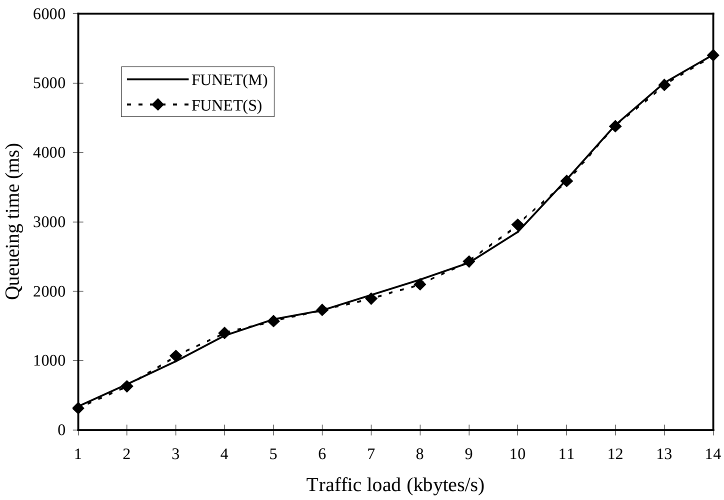


Figure 4.8. The mean queueing time of multi-rate services (M) and single-rate service (S).

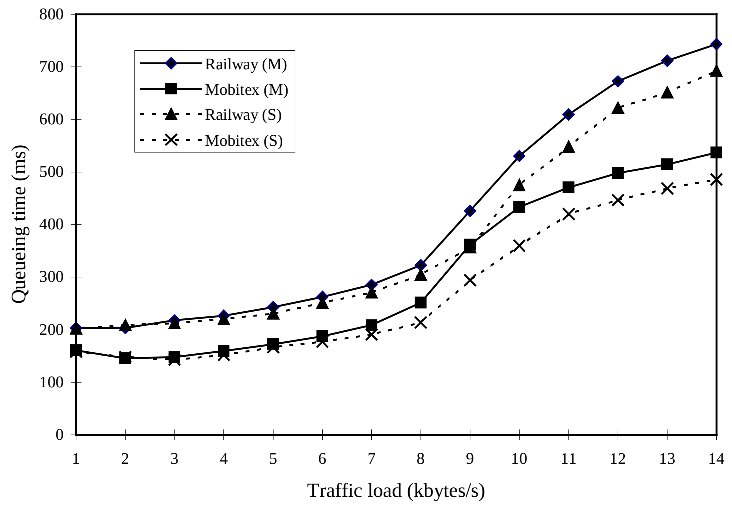


Figure 4.9. The mean queueing time of multi-rate services (M) and single-rate service (S).

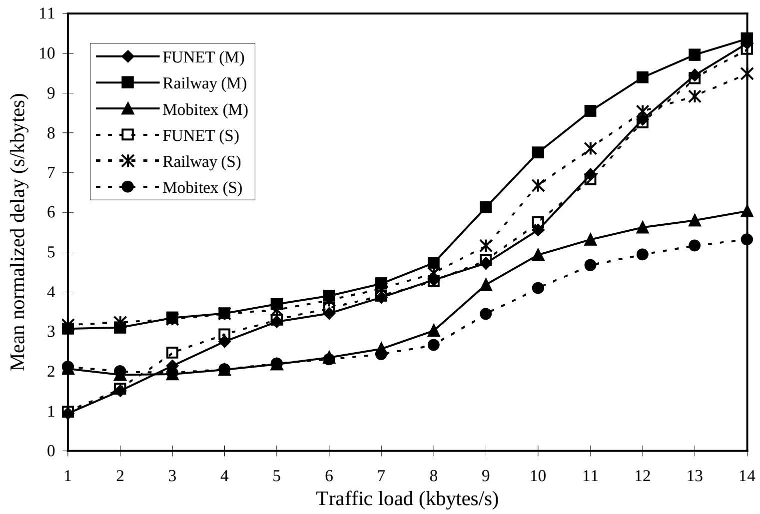


Figure 4.10. The mean normalized delay of multi-rate services (M) and single-rate service (S).

Figure 4.10 shows the mean normalized delay of the multi-rate services and the single-rate service. The normalized delay of the Railway model is highest in the three models. The reason is that because the Railway model has a high percentage of the low packet size. The characteristic of delay performance in downlink is quite different from that in uplink shown in [35], especially for the Mobitex traffic (with shorter packet sizes in uplink). For the FUNET model, the multi-rate services has the similar normalized delay as the single-rate service, however, for the Railway and Mobitex models the single-rate service has a lower delay than the multi-rate services when traffic load increases. This characteristic further shows the inefficiency of the multiple-slot allocation.

Figure 4.11 shows the comparison of the 95 percentile normalized delay of the single-slot allocation in the multi-rate services and the single-rate service. From Figure 4.11, we find that the delay of the single-slot allocation in the multi-rate services is the major contribution to the network mean delay. The multi-slot allocation significantly increase the delay of the single-slot service, which may be the “basic service” provided by the network and have the largest number of users due to the limitation of the network capacity.

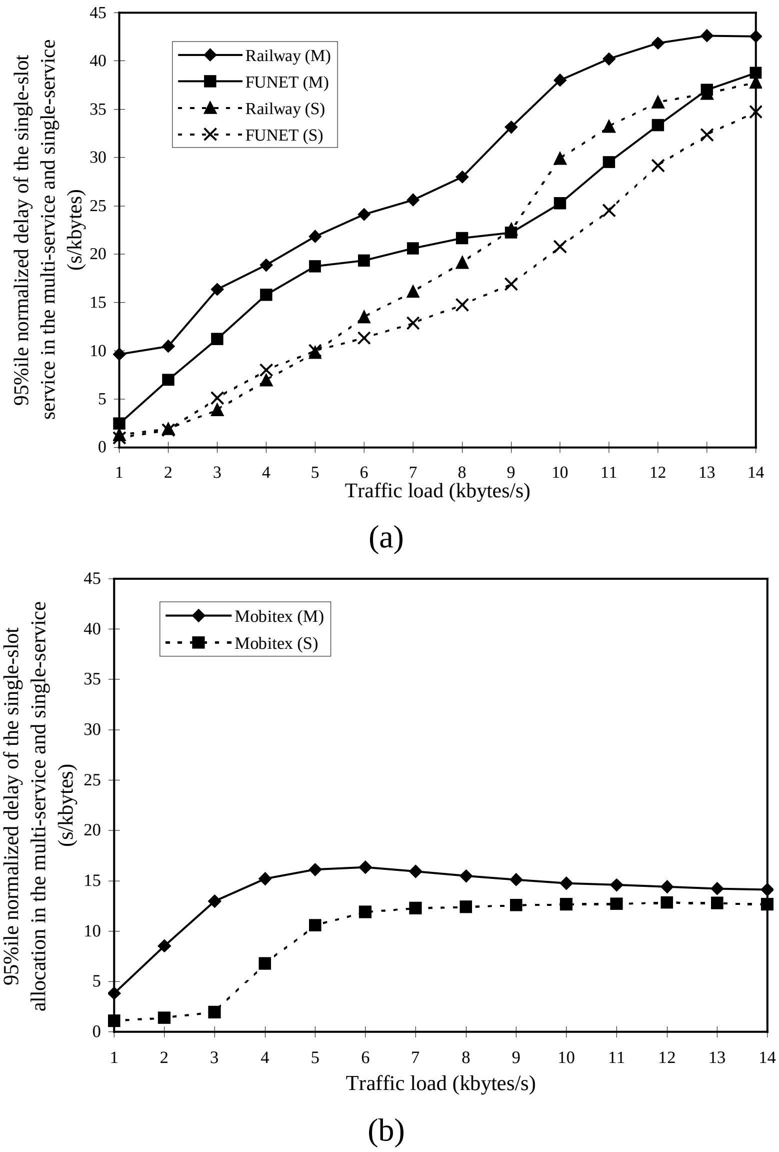


Figure 4.11. Comparison of the 95% percentile normalized delay of the single-slot allocation in the multi-rate services (M) and the single-rate service (S).

Due to the dynamic variation of free channels available from voice services and the interference limitation, the multiple-slot allocation does not show a gain of the mean throughput neither a decrease of the mean delay. This result is different from the result of the uplink performance [36]. Therefore, in the GPRS capacity planning, a compromise is needed to be done between the uplink capacity and downlink capacity. In multi-rate services, the multi-slot allocation significantly increases the delay of the single-slot service.

4.3 Aspects of GPRS network planning

GPRS is a new packet data service over the GSM and will be operated on the base station sites of existing GSM networks. Two principles are needed to be considered when a GPRS system is planned. The first one is that the effects on the existing voice services due to the introduction of GPRS, such as blocking probability and call dropping rate, should be kept as small as possible. The second one is that the reconfiguration of the existing network due to the introduction of GPRS must be avoided as possible, i.e., minimizing the additional GPRS deployment cost. In this section, the issues of GPRS network planning are discussed.

4.3.1 Service requirements

GPRS should support wireless applications such as retrieving information from wireless information centers and mobile offices, including World Wide Web surfing, file transfer, remote network login, sending and receiving e-mail, cash and fund transfers, stock market information transfer, credit card validations, lottery transactions, utility reading and electronic monitoring. Applications such as news, weather and traffic reports, taxi and fleet management may be included. The traffic of those applications exhibits the characteristic [27] from frequent transmission of small volumes of data, to intermittent and non-periodic (bursty) transmission of median volumes of data, and infrequent transmission of large data volumes.

The traditional voice traffic model is not valid for GPRS data traffic anymore. ETSI has suggested [52] some traffic models such as FUNET model, Mobitex model and Railway model for GPRS performance evaluations. For GPRS network planning we can apply those models. However, a traffic model of WWW surfing should be included. A model suggested by D-ETR SMG [43] is a simple model for GPRS planning.

GPRS provides both connectionless and connection-oriented point-to-point (PTP) packet mode transmission as well as point to multipoint (PTM) services including the multicast, group call and IP multicast transmission. The amount of radio resources allocated to GPRS is adapted to local data traffic conditions and can be increased and decreased when it is needed. This functionality gives a possibility that users acquire their services on basis of online negotiation. For the PTP and PTM services, the quality of service (QoS) parameters, such as the user data throughput, QoS class (transfer delay, priority) and reliability of transmission, may be negotiated or set to default values. For mean throughput, the mean bit rates can be negotiated to a value up to 111 kb/s [27]. Transfer delay and priority are combined into one parameter, the QoS class (Table 4.4). The GPRS planning should provide the dynamic range of the resource enabled to be used for GPRS and define the service classes supported by the network on basis of the system performance parameters, e.g., throughput and delay. In addition some information should be provided for the service negotiation and operation.

TABLE 4.4. GPRS QoS class.

packet size	128 octets		1024 octets	
	mean delay	95%ile delay	mean delay	95%ile delay
class 1 (predictive)	0.5 s	1.5 s	2 s	7 s
class 2 (predictive)	5 s	25 s	15 s	75 s
class 3 (predictive)	50 s	250 s	75 s	375 s
class 4 (best effort)	unspecified			

4.3.2 Remaining capacity of an existing network

In chapter 3, it has been shown that the resources unused by voice services might not all be used for carrying GPRS traffic due to the constraints of the interference probability or the outage probability of a GSM network. In order to guarantee the quality of service for GSM voice services the network can allocate resources to GPRS only if it does not jeopardize the voice services and the maximum extra resources enabled to allocate to GPRS are called the remaining capacity of an existing network here. In GPRS radio network planning the planners are first needed to answer what is the maximum resource that can be used for GPRS without causing the quality of voice services under the acceptable limit.

Hence, in the procedures of the GPRS system planning, an evaluation of the signal to interference ratio (SIR) level in the existing network must be done. The SIR evaluation can be done by measurement, simulation or both.

The calculation of the outage probability of a GSM-GPRS network can provide some guidelines to decide the maximum number of channels allocated to GPRS. The method presented in chapter 3 can be used to calculate the outage probability of the GSM-GPRS network for both the non-frequency hopping and the frequency hopping systems. As an example the remaining capacity of a hexagonal cell's network with a reuse factor of 7 is estimated in following by the calculation of the outage probability. In the calculation, the system parameters in table 3.1 are used except for the standard deviation of the power control error with 1 *dB*.

Figure 4.12 shows the 90th percentile outage probability as function of the number of channels used by voice and GPRS services and with the number of channels used for GPRS as a parameter. The figure shows that the number of channels enabled to be allocated to GPRS is depended on the voice traffic, however the channels unused by voice services may not be all used for GPRS. Up to 5 channels can be allocated to GPRS without degrading the network outage probability above the limit of 10%. When the number of free channels is higher than 5, the percentage of those free resources that can be allocated to GPRS is about 50%-80%. For example, considering the number of voice users to be 16 (thus the number of free channels is 13), for Figure 4.12 we find that the number of free channels that can be allocated to GPRS is at most 8. The more free channels, the lower percentage of those free channels can be allocated to GPRS. A simple admission control policy for allocating those free channels to GPRS could be constructed on the basis of such kind of simulation results.

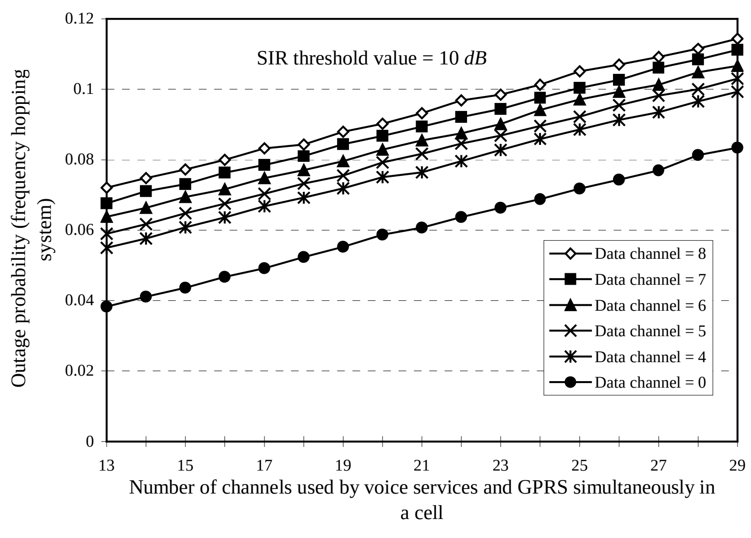


Figure 4.12. The 90th percentile outage probability as function of the number of channels used by voice and GPRS services.

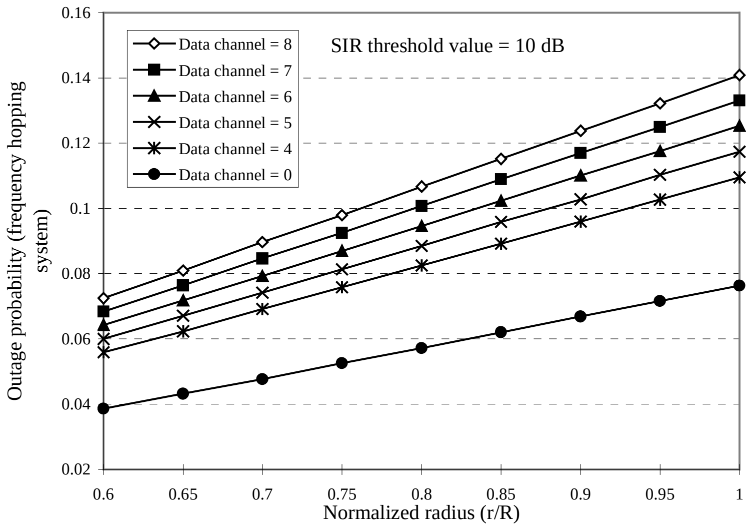


Figure 4.13. The 90th percentile outage probability as function of the normalized radius (r/R) of a cell.

Figure 4.13 shows the 90th percentile outage probability as function of the normalized radius (r/R) of a cell and with the number of channels used for GPRS as a parameter. The figure shows that the service area of a cell shrinks after introducing GPRS. This implies that GPRS may cause the system to have a higher dropping rate for call handover. When up to 5 channels are used for GPRS, the SIR value on the border may be below the threshold value.

In GPRS network planning we need to evaluate the remaining capacity of a network. On the basis of that evaluation, we can have a general picture for resources enabled to be allocated to GPRS. This is a trade-off process between the quality of voice services and the capacity gain of GPRS.

4.3.3 Capacity planning

The next question of GPRS radio network planing is how much average capacity is obtained by overlaying GPRS onto an existing GSM network. As GPRS applications vary from high-bit-rate services provided for business applications to low-bit-rate services like the normal file transfer, the resource occupied by a GPRS user may vary from 1 to 8 timeslots. The traffic density generated will depend on the environment characteristic (downtown, suburban areas, business areas, rural area), the mix of terminal types, the demographic situation, the penetration factor of GPRS services and the ranking or the popularity of applications. According to those factors the distribution of the traffic demand can be estimated. The traffic can be sorted into groups according to the basic bandwidth or timeslots required for each group.

Since the bursty characteristic of the traffic, the variable data rate and the packet switched transmission, the GPRS traffic capacity required can no longer be specified by the single unit of voice services. The transmission rate could be used as the unit of traffic capacity. For example, the traffic can be represented as kilobits per second per square kilometer (kb/s/km^2).

The main tasks of the GPRS capacity planning are not only to try to meet the capacity required by the traffic with an acceptable blocking probability and delay, but also to provide decision criteria to network for online QoS negotiation. To minimize the effort and cost for the network operator, the original network configuration including cell planning, frequency planning, setting of power and other cell parameters, should not require extensive modification. In the initial phase of the GPRS launch, the network reconfiguration might not be needed. As the GPRS getting more and more popular the reconfiguration may not be avoided, e.g., more base stations are needed to be installed.

Due to the bursty characteristic of the GPRS traffic and multi-rate parallel services provided, the Erlang B or Erlang C formula can not be applied in capacity planning. It does not require permanently allocated packet data channels (PDCHs). The operator can decide to dedicate permanently or temporarily some physical channels for GPRS traffic. In addition, rather than having a dedicated packet common control channel (PCCCH), the GPRS may utilize the existing GSM paging and control channels. For such a dynamically variable resource and the bursty traffic, the capacity offered by the network may only be able to obtain from simulations.

The resource that can be allocated to GPRS depends on the outage level of an existing network, the blocking probability target for voice services and the number of the transmitter-receiver units (TRXs) available in the base station. After the maximum number of channels allocated to GPRS is decided from the last planning procedure, the next step is to simulate the system performance in order to obtain the throughput and delay parameters. In the practical GPRS planning, it should be done by computer automatically. In section 4.2.2 we have shown the capacity performance of downlink differs that of uplink shown by previous studies. Therefore, in the GPRS capacity planning, a compromise is needed to be done between the uplink capacity and downlink capacity. On basis of the system performance parameters, e.g., throughput and delay, and the service classes supported by the network are defined.

4.3.4 Coverage planning

The main purpose of coverage planning is to achieve the required radio coverage with specified time and location probability. In traditional voice services it is achieved by the link budget within the range of the transmitted power level. Since GPRS is deployed on top of an existing GSM network, the same link budget as that of voice services may be used for GPRS. However, we need to consider that the outage probability near to the cell border area will be increased as more channels used for GPRS. If the outage exceeds the network target it will reduce the real served area of a cell and may cause a higher dropping rate of intercell handover after the introduction of GPRS.

Four coding schemes are used for GPRS. For those four coding schemes, the specification of GSM 05.05 [42] has defined the required SIR values (Table 4.3) corresponding to a BLER (block error rate) value not more than 10% for systems with and without frequency hopping. Those SIR values have been included an implementation margin of 2 dB. The coding scheme CS-1 is mainly used for the GPRS signaling and is the same scheme which is used in the GSM signaling channel, e.g., the standalone dedicate common channel (SDCCH). Almost every required SIR value for those coding schemes is higher than the 9 dB target value required for GSM voice services. In addition, when more channels is allocated to GPRS, it has an impact on the quality both of voice services and GPRS. Therefore, the GPRS service might not have coverage in some areas in a cell for some networks. The network planning should at least try to cover the areas with a high service demand. If the degradation of service coverage occurs temporarily, it will not be a problem for GPRS because the transmission can be delayed and retransmitted. However, if the effect is in long term we have to face the network reconfiguration problem, e.g., installing new base stations, or having a larger reuse factor and more frequency carriers.

The varying SIR target values for different coding schemes may let the users to achieve a higher data rate near the base station and a lower date rate near the cell border in generally. We do not need to define which coding scheme should be used beforehand.

4.3.5 Frequency planning

The operator needs to decide if some physical channels are dedicated to GPRS traffic permanently or temporarily. Normally no operator is willing to dedicate one frequency carrier permanently to GPRS because of the limitation of frequency carrier, therefore no frequency planning is needed. However, a small change in the original frequency planning might be needed in order to meet the capacity demand and the new SIR requirement for GPRS.

The intelligent underlay-overlay (IUO) [44-45] used for frequency reuse partitioning in GSM network still can be used for GPRS, but the SIR threshold values should be different from those of voice services. The IUO can provide flexible data rate for GPRS due to the option of several coding schemes. However, GPRS will reduce the original capacity gain from IUO in GSM network, because the probability of using “super layer” channels is reduced and the probability of using “regular layer” channel is increased. If more channels are used for GPRS, the whole cell may only be able to use the “regular layer” channels.

4.4 Chapter Summary

In this chapter we have investigated the GPRS downlink performance with a dynamically varying number of channels and outlined the principles of GPRS network planning. In the investigation of GPRS downlink performance, we have discussed both the single-cell case without considering the network interference level and the multi-cell network case when considering the network interference level.

The motivation for studying the single-cell case is to provide simple methods for preliminary system design (e.g., designing the system “buffer size”) and performance evaluation. An approximation method is discussed for the evaluation of GPRS performance in single-slot allocation with the varying amount of the radio resources. The approximation method given here could be used for preliminary design and evaluation of GPRS performance when the average service time of voice services is much longer than that of GPRS, e.g., μ_d/μ_v>100. When the adaptive multislot allocation scheme is used, the blocking probability and the mean queueing time of the multiple services are near to those of the single-slot service. Therefore, the approximation method can also be applied for the preliminary system design and performance evaluation of multiple services.

GPRS capacity performance in downlink is quite different from that in uplink because of the difference in the transmission protocols. The GPRS transmission efficiency is highly affected by the packet sizes of the data traffic. The multiple-slot allocation does not show a gain in the mean throughput neither a decrease in the mean delay. This result is different from the result of the uplink performance [36]. The multi-slot allocation significantly increases the delay of the single-slot service, therefore, a control of the multi-slot services is needed and the implementation of the high number of multi-slot services may not be a good strategy for operator.

The principles of GPRS network planning on top of the existing GSM system have been outlined. Because it is planned on top of the existing network, the remaining capacity of an existing network should be evaluated in the aspect of interference constraints. Otherwise, GPRS service may damage the GSM voice services. The maximum radio resources allocated to GPRS are defined as the network remaining capacity. For coverage planning, we need to consider that the outage probability near to the cell border area will be increased as more channels used for GPRS. If the outage exceeds the network target it will reduce the real served area of a cell and may cause a higher handover dropping rate after the introduction of GPRS.

5. CONCLUSIONS AND FURTHER RESEARCH

5.1 Conclusions

The first objective of this thesis is to investigate the relationship between the network capacity and the quality of service in order to increase the utilization of radio resources. The second one is to develop algorithms or methods for network control to manage the radio resources efficiently and for radio network planners to plan the network optimally.

In order to approach the research objectives, in this thesis, the author have proposed two dynamic channel allocation (DCA) schemes and developed an analysis method to investigate the GPRS impact on the GSM voice services. In addition, the GPRS performance is investigated and guidelines for GPRS network planning have been presented.

In the proposed DCA algorithms, the effect of the channel allocation on existing calls is considered by the evaluation of the call outage rate or a cost function. In the first proposed algorithm, called the Minimum Call Outage (MCO) algorithm, in order to evaluate the call outage caused by those candidate channels the author has presented a method of estimating the average SIR variation of on-going calls. The MCO scheme minimizes the call outage rate of the existing calls when assigning a channel to a new call. The MCO scheme improves the capacity or QoS performance compared with the First Available and Maximum SIR schemes, however, there are some limitations in this algorithm, for example, a wrong decision on channel allocation may be made due to using obsolete SIR values and the error of SIR measurement.

In the second proposed algorithm a cost function has been introduced to estimate the cost of the assignment of a candidate channel. By simulation we find that the cost-function can be introduced for channel pre-selection to reduce the call dropping and handover rates and speed up the call set-up. In the proposed DCA algorithm which combines the channel assignment and power assignment, the cost function is used for channel pre-selection, and the power to be assigned is probed on the pre-selected channels. This algorithm largely increases the capacity compared with the FCA. It does not cause a high intracell handover rate and performs with a short call setup time.

In order to investigate the GPRS impact on GSM voice services, in this thesis, the author has presented an analysis method to calculate the outage probability of the GSM-GPRS network for both the non-frequency hopping and frequency hopping systems. This method takes into account fast (Rayleigh) fading, power control (with error), discontinuous transmission, and frequency hopping, and can be utilized for the evaluation of network performance and network planning for GSM-GPRS cellular systems. The effects on the quality of voice services due to the introduction of GPRS into GSM network are evaluated by calculating the outage probability. GPRS increases the outage probability of existing GSM voice services. The outage probability increase of the network with a small frequency reuse factor caused by GPRS is higher than that of the network with a large frequency reuse factor. That means the GPRS affects more on the QoS of voice services of the network with a small reuse factor. GPRS will reduce the cell service area, but the reduction percentage of the cell service area for the system with small reuse factor is higher than that for the system with large reuse factor.

By the investigation of GPRS impact on GSM voice service, it is found that those channels unused by voice services might not all be used for carrying GPRS traffic. The number of channels which can be allocated to GPRS depends on the difference between the outage level of the existing GSM network and the maximum acceptable outage level.

An approximation method has been discussed for the evaluation of GPRS performance in single-slot allocation with the varying amount of the radio resources. The approximation method given here can be used for the preliminary design and evaluation of GPRS performance when the average service time of voice services is much longer than that of GPRS, e.g., μ_d/μ_v>100. The simulation results show that the blocking probability and the mean queueing time, of multiple services with the adaptive multislot allocation, are neat to those of the single service with single-slot allocation. Therefore, the approximation method can be applied for the preliminary design and evaluation of multiple services also.

GPRS downlink performance of a multicell network with SIR adaptive coding schemes in dynamic variable radio resource has been investigated. The simulation results show the performance in downlink is quite different from that in uplink shown in previous studies because of no contention phase in downlink transfer. The GPRS transmission efficiency is highly affected by the packet sizes of the data traffic. It will cause both low efficiency of transmission and high signaling load if the data packet size is too small. The multiple-slot allocation does not show a gain in the mean throughput neither a decrease in the mean delay. This result is different from the result of the uplink performance [36]. The multi-slot allocation significantly increase the delay of the single-slot service, therefore, a control of the multi-slot services is needed in the network and the implementation of a high number of multi-slot services (e.g., more than 4 or 5 slots) may not be a good strategy for operator.

Because GPRS radio network planning is planned on top of an existing GSM network, the most important issue for GPRS planning is to evaluate correctly the remaining capacity of an existing network in the aspect of interference constraints. Otherwise, GPRS service may damage the GSM voice services. The remaining capacity of a existing network is defined by the maximum amount of radio resources in average which can be allocated to GPRS. The capacity planning is done by simulating the system performance parameters, e.g., throughput and delay, and defining the service classes supported by the network on basis of the network remaining capacity. For coverage planning, it needs to consider that the outage probability near to the cell border area will be increased when more channels are used for GPRS. If the outage increase is beyond the network target it will reduce the real served area of a cell and cause a higher handover dropping rate after the introduction of GPRS. GPRS will reduce the original capacity gain from IUO in GSM network, because the probability of using “super layer” channels is reduced and the probability of using “regular layer” channel is increased. If more channels are used for GPRS, the whole cell may only be able to use the “regular layer” channels.

5.2 Future Work

In the proposed DCA algorithms, the modeling of the effect of the channel allocation on existing calls is mainly based on a homogenous network with regular hexagonal cells. In the practical network, the cell geometry and the cell size might not be the same for all cells. It would be interesting and useful to extend the method to model the effect of the channel allocation on existing calls for a network with different cell geometry and cell size. For such a network, it can be foreseen that the cost of a channel to existing calls might have more classes and be non-linearly scaled.

Even though the adjacent channel interference has been considered in the first proposed DCA method, it has not be considered in the second proposed DCA method. For a TDMA system typically with a few frequency carriers, it is very difficult to avoid the adjacent channel interference, therefore a DCA algorithm should provide a mechanism to handle the adjacent channel interference. This is one of interesting topics for the future work of the DCA proposed in this thesis.

In this work, the GPRS impact on GSM voice services is investigated in a network which uses the fixed channel allocation for the radio resource management. Some interesting researches have been done [65-66] on packet data services with DCA method. It would be interesting to investigate what is difference when the dynamic channel allocation is applied for the radio resource management. Also, the difference of the GPRS capacity gain between last two cases of radio resource management would be interesting for study.

In chapter 4, The downlink simulation results of multiple-slot allocation do not show a gain in the mean throughput neither a decrease in the mean delay. Those results are different from the results of the uplink performance [36]. However, the simulation environments used in downlink are not exactly same with those used in uplink, it is interesting and necessary to do the uplink simulation in order to make a further conformation of this conclusion.

SUMMARY OF PUBLICATIONS

Publication 1: Dynamic Channel Allocation Based on SIR Estimation (MDMC'96)

When assigning a channel to a new call, it may cause the deterioration of on-going calls or call drop. In this paper, a method to estimate the average SIR variation of on-going calls due to the assignment of an incoming call, has been developed. The effect of a candidate channel on existing calls is considered by the evaluation of the call outage rate and the average SIR degradation can be estimated. A new dynamic channel allocation scheme, the Minimum Call Outage (MCO) scheme which takes into account the network capacity, the stability of the network and the quality of service, is presented. The MCO scheme minimizes the call outage probability and reduces the call drop probability when assigning a channel to a new call.

Simulation results show that the call block probability for the first available (FA) scheme is lower than that for the maximum SIR scheme, but the call drop probability is higher and especially the call outage probability is significant high. The MCO scheme can provide much improvement in the call drop probability and call outage probability compared with the FA scheme, but only with minor degradation on the call block probability. The MCO scheme gives trade-off between the FA scheme and the Maximum SIR scheme.

Publication 2: Cost-Function Based Distributed Channel Allocation (APCC'97)

In this paper, a simple cost function based on the average interference has been introduced to estimate the cost of the assignment of a candidate channel. A distributed channel access algorithm, called the minimum cost algorithm, is proposed. In this algorithm the cost-function is used to select a channel for call set-up.

By comparing the performance of the minimum cost scheme with that of first available (FA) and maximum signal to interference ratio (MSIR) schemes, we find that the minimum cost scheme has lower call dropping probability and unsuccessful call probability than the other two schemes. The FA scheme achieves the low call blocking probability by intensive call dropping. For the call set-up time, the minimum cost scheme has better performance than FA and MSIR schemes. The average call set-up time of the minimum cost scheme is shorter than that of FA and MSIR schemes.

Publication 3: Distributed Channel Allocation Algorithm with Power Control (IEEE PIMRC'97)

In this paper, we integrate the channel assignment and power assignment into a distributed channel access algorithm. The cost-function introduced in *Publication 2* is used to provide optional channels according to their cost for transmitted power level searching.

The simulation results show that this algorithm largely increases capacity compared with the fixed channel allocation (FCA). The proposed algorithm does not cause high intracell handover rate. It has a short average call setup time even at high traffic load. We suggest that the intracell handover rate should be a factor in evaluation of the performance of an algorithm, because high handover access will intensively increase the load of signaling load and cause much higher call dropping and blocking probabilities.

Publication 4: Outage Probability in GSM-GPRS Cellular Systems with and without Frequency Hopping (*Wireless Personal Communications,* Vol. 14, Sept. 2000)

In this paper, we present a method to calculate the outage probability of the GSM-GPRS network for both the non-frequency hopping and the frequency hopping systems. This method takes into account Rayleigh fading, power control (with error), discontinuous transmission, and frequency hopping (if applied), and can be utilized to the evaluation of network performance and network planning for GSM-GPRS cellular systems.

The effects on the quality of voice services due to the introduction of GPRS into GSM network are evaluated by calculating the outage probability. Obviously, GPRS increases the outage probability of existing GSM voice services. With a 10 *dB* SIR threshold value, the outage probability increases by about 5%-10% whenever the number of channels used for GPRS increases by one. The power control error has more impact on the system performance when more channels are allocated to GPRS. The system with frequency hopping may provide more channels to GPRS at the low channel occupancy, however, for non-frequency hopping system, some voice users may need to have multiple intracell handovers in order to provide more channels to GPRS at the low channel occupancy. The cell service area is decreased by about 10%~20% whenever the number of channels used for GPRS increases by one. As more channels are provided to GPRS the cell service area decreases dramatically. Consequently, the dropping rate of the intercell handover for GSM network may increase due to the introduction of GPRS.

Therefore, channels unused by voice services might not all be used for carrying GPRS traffic. The number of channels allocated to GPRS depends on the difference between the outage level of the existing GSM network and the maximum acceptable level. Therefore, in order not to damage the GSM voice services the remaining capacity of an existing network must be correctly evaluated.

Publication 5: Outage Probability for GPRS over GSM Voice Services (IEEE VTC’99-fall)

In this paper the effects on the quality of voice services due to the introduction of GPRS into GSM network are further investigated by calculations of the outage probability. Based on *Publication 4*, the outage probability of non-frequency hopping system and frequency hopping system with different frequency reuse factors has been calculated.

For both non-frequency hopping system and frequency hopping system, GPRS affects the QoS of voice services of the network with small reuse factor more than that of the network with large reuse factor. The power control error has more impact on the system performance when more channels are allocated to GPRS. GPRS will reduce the cell service area, but the reduction percentage of the cell service area for the system with small reuse factor is higher than that for the system with large reuse factor.

Publication 6: GPRS Performance Estimation in GSM Circuit Switched Services and GPRS Shared Resource Systems (IEEE WCNC'99)

In this paper, an approximation method is used for evaluation of the GPRS performance of single-slot service on varying amount of radio resources. The method could be used for preliminary design and evaluation of GPRS performance when the average service time of voice services is much longer than that of GPRS. The simulations show that the interruption probability of GPRS transmission due to release of channels to the upcoming demand of voice services depends on the average message size more strongly than on the traffic load. When the adaptive multislot allocation scheme is used, the GPRS performance in blocking probability and the mean queueing time with multislot allocation is near to the performance of the single-slot allocation and only depends on the total arrival rate of packet traffic rather than on the composition of the arrival traffic rate of each service group.

Publication 7: GPRS Network Planning on the Existing GSM System(IEEE Globecom'2000)

In this paper the guidelines and principles of GPRS network planning on top of the existing GSM system have been presented. Some important issues for GPRS network planning have been pointed out. Because GPRS is planned on the existing network, the GPRS planning differs somewhat from the normal cellular network planning. The most important issue for GPRS planning is to evaluate correctly the remaining capacity of an existing network. Otherwise, GPRS service may damage the GSM voice services.

The capacity planning is done by planning the system performance parameters, e.g., throughput and delay, and defining the service classes supported by the network on basis of the network remaining capacity. GPRS capacity performance in downlink is quite different from that in uplink because of the difference in the transmission protocols. The GPRS transmission efficiency is highly affected by the packet sizes of the data traffic. It will cause both low efficiency of transmission and high signaling load if the data packet size is too small. The multiple-slot allocation does not show a gain in the mean throughput neither a decrease in the mean delay. This result is different from the result of the uplink performance [36]. One of the reasons is that their result is from simulation with a fixed amount of resources. The multi-slot allocation significantly increase the delay of the single-slot service, which may be the "basic service" provided by the network and have the largest number of users due to the limitation of the network capacity. Therefore, a control of the multi-slot services is needed in the network and the implementation of a high number of multi-slot services (e.g., more than 4 or 5 slots) may not be a good strategy for operator.

For coverage planning, the same link budget as that of voice services may be used for GPRS. However, we need to consider that the outage probability near to the cell border area will be increased when more channels are used for GPRS. If the outage increase is beyond the network target it will reduce the real served area of a cell and cause a higher handover dropping rate after the introduction of GPRS. The frequency planning is not needed for GPRS normally. GPRS will reduce the original capacity gain from IUO in GSM network, because the probability of using "super layer" channels is reduced and the probability of using "regular layer" channel is increased. If more channels are used for GPRS, the whole cell may only be able to use the "regular layer" channels.

ERRATA

Publication 5:

Error 1:

Figure 3 and Figure 4 are misplaced each other. That is that Figure 4 is the outage probability for a system without power control error and Figure 3 is the outage probability for a system with 2 *dB* standard deviation of power control error. The correct figures are:

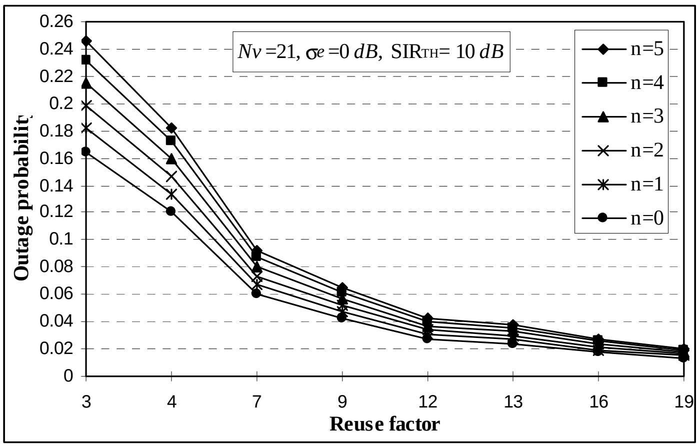


Fig. 3 The 90% worst case outage probability of the frequency hopping system (perfect power control σ_e=0 *dB*). N_v and n denote the number of channels simultaneously used by voice services and GPRS respectively.

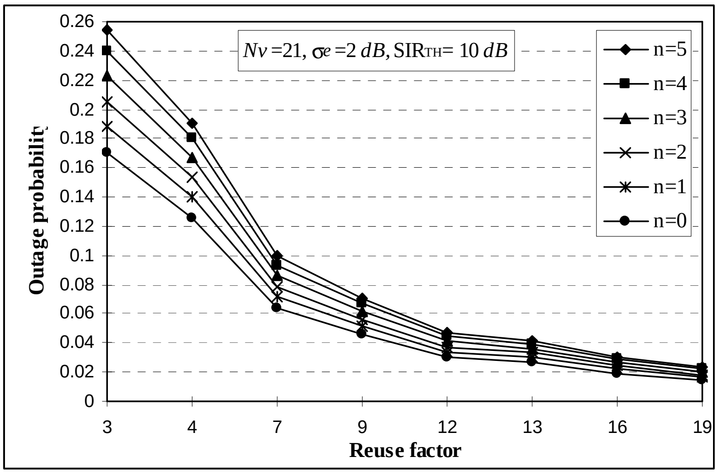


Fig. 4 The 90% worst case outage probability of the frequency hopping system (power control with an error of 2 *dB* standard deviation, σ_e=2 *dB*). N_v and n denote the number of channels simultaneously used by voice services and GPRS respectively.

Publication 6:

Error 1:

Figure 3: The legend in the figure is incorrect. The correct one is shown in following figure.

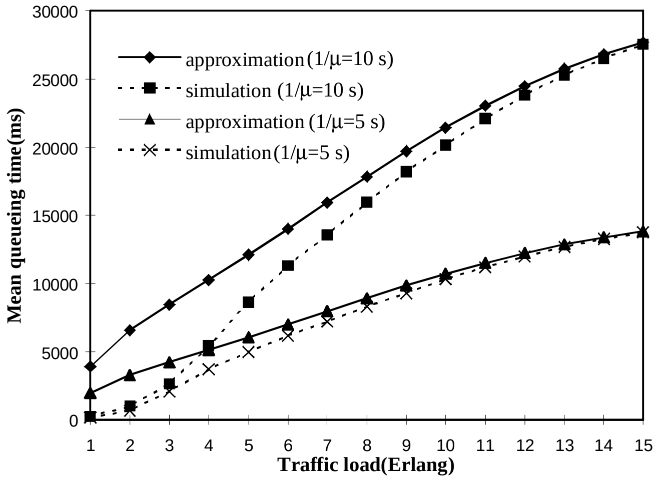


Fig 3. The mean queueing time of single-slot service for the average service time (1/μ) of 5 s and 10 s respectively.

Error 2:

Figure 11: The legend in the figure is incorrect. The correct one is shown in following figure.

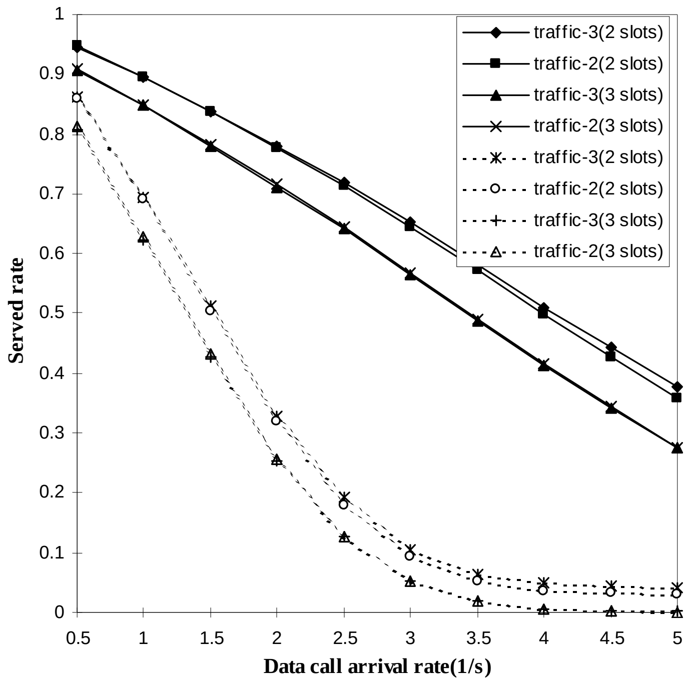


Fig. 11 The served rate of muitlslots services with resource allocation *scheme*-2 for the average message size of 2×13.4 kb and 5×13.4 kb respectively. The solid lines are used for the average message size of 2×13.4 kb and dashed lines for the average message size of 5×13.4 kb.

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PUBLICATIONS 1-7.