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CAPACITY AND PERFORMANCE ANALYSIS OF A MULTI-USER, MIXED TRAFFIC GSM NETWORK

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**A thesis submitted to the Faculty of Graduate Studies and Research
In partial fulfillment of the requirements of the degree of**

Master of Engineering in Electrical Engineering

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Studies and Research the acceptance of the thesis

**Capacity and Performance Analysis of a Multi User
Mixed Traffic GSM Network**

Submitted by Turki Darweesh
In partial fulfillment of the requirements for the degree of
Master of Engineering
in Electrical Engineering

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ABSTRACT

Mobile communication systems have experienced rapid growth in the number of users as well as the range of services provided during the last two decades. The Global System for Mobile communications, GSM, is a pan-European mobile communication system in the 900 MHz band which was first introduced in the early years of this decade. An increasing demand for data transmission services over the GSM system has been driven by the wide use of the internet applications.

In this thesis, a GSM network model is considered with a combined voice and data services where data service is supported for variable bit transmission rate. The objective of this thesis is to analyze a typical GSM network by means of modeling and simulation in order to determine the capacity limits for hybrid voice and data traffic under different conditions represented by a number of system parameters. These parameters include: load factor, voice/data users traffic ratio, transmitted power, frequency reuse pattern, frequency hopping, mobility, and transmission rate.

A computer simulation program was developed and used to determine the performance of the GSM network model. Results of the simulation are obtained and presented in the form of Cumulative Distribution Function (CDF) curves of the Bit Error Rate (BER). These results are discussed and analyzed in the last two chapters of this thesis.

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LIST OF ABBREVIATIONS & ACRONYMS

ARGH	Access Grant Channel.
ARQ	Automatic Repeat Request.
AUC	Authentication Center.
BCCH	Broadcast Control Channel.
BCH	Broadcast Channel.
BER	Bit Error Rate.
BS	Base Station.
BSC	Base Station Controller.
BSS	Base Station Subsystem.
BTS	Base Transceiver Station.
CCCH	Common Control Channel.
CCH	Control Channel.
CDF	Cumulative Distribution Function.
CDMA	Code Division Multiple Access.
CEPT	Conference Europe'ennee des Postes et Te'le'communication.
CH	Cyclic Hopping.
CIR	Carrier-to-Interference Ratio.
CIRF	Co-channel Interference Reduction Factor.
CRC	Cyclic Redundancy Check.
DCCH	Dedicated Control Channel.
EDGE	Enhanced Data rates for Global Evolution.
EIR	Equipment Identity Register.
ETSI	European Telecommunications Standard Institute.
FACCH	Fast Associated Control Channel.
FCCH	Frequency Correction Channel.
FDD	Frequency Division Duplexing.
FDMA	Frequency Division Multiple Access.
FFH	Fast Frequency Hopping.
GGSN	Gateway GPRS Super Node.
GMSK	Gaussian Minimum Shift Keying.
GOS	Grade Of Service.
GPRS	General Packet Radio Services.
GSM	Global System for Mobile communications.
GTP	GPRS Tunnel Protocol.
HLM	High Level Modulation.
HLR	Home Location Register.
HSCSD	High Speed Circuit-Switched Data.
IP	Internet Protocol.
ISDN	Integrated Services Digital network.

MAHO	Mobile Assisted Hand Off.
MOMENTS	MObile Media ENTertainment Services.
MS	Mobile Station.
MSC	Mobile Switching Center.
NSS	Network and Switching Subsystem.
OMC	Operational and maintenance Center.
OQPSK	Offset Quadrature Phase Shift Keying.
OSS	Operation Support Subsystem.
PCH	paging Channel.
PL	Path Loss.
PSTN	Public Switched Telephone Network.
QAM	Quadrature Amplitude Modulation.
RACH	Random Access Channel.
RH	Random Hopping.
RLC	Radio Link Control.
RXLEV	Received Signal Level.
RXQUAL	Received Quality Level.
SACCH	Slow Associated Control Channel.
SCH	Synchronization Channel.
SDCCH	Stand Alone Dedicated Control Channel.
SFH	Slow Frequency Hopping.
SGSN	Serving GPRS Super Node.
SIM	Subscriber Identity Module.
SMSC	Short Message Service Center.
TCH	Traffic Channel.
TCP/IP	Transmission Control Protocol / Internet Protocol.
TDD	Time Division Duplexing.
TDMA	Time Division Multiple Access.
UMTS	Universal Mobile Telecommunications System.
VLR	Visitor Location Register.

Chapter 1

INTRODUCTION

Mobile radio systems have gained increasing importance in commercial, business, and public safety sectors as well as for personal use. These systems enable their users to travel freely within the service area and simultaneously communicate with telephone, fax, data modem, and electronic mail subscribers anywhere in the world. The birth of the mobile radio technology have occurred in 1897 when Guglielmo Marconi was given the credit for the invention of the wireless telegraph [Kuc91]. Since then, mobile communications have evolved dramatically and continue to have an enormous impact on all sectors of economic activities.

A major part of this growth has been due to two factors. First, the adoption of digital modulation technology instead of the previously used analog modulation. Second is the concept of reusing the same allocated spectrum in different geographical areas (cells) which is known as *Cellular* concept. These two factors have facilitated the development of large scale cellular mobile networks with efficient spectrum utilization.

While voice was the primary service provided by early communication systems, current systems offer other transmission services. One of the most commonly used cellular systems is based on the Global System for Mobile communication, GSM, standard. In this thesis, a GSM network model with combined voice and data services will be analyzed and simulated in order to examine its capacity limits for hybrid voice and data traffic. This Chapter will briefly describe the objectives, motivations, organization, and contribution of the thesis.

1.1 Thesis Objective

The objective of this thesis is to study a GSM network that provides both voice services and data services where data service is supported for a variable transmission bit rates. The network tries to satisfy the bit rate requested by the users subject to resources availability. Performance as well as capacity of GSM network model is to be investigated and analyzed under different conditions and scenarios with regard to a number of system parameters of interest. These parameters include loading factor, transmitted power, frequency reuse plan, frequency hopping etc. The objective is to analyze the influence that each parameter has on the overall performance and capacity of the GSM network.

The objectives of this research can be summarized as follows:

1. To study the effect of voice users to data users occupancy ratio on the quality of transmission for data users.
2. To examine the influence of parameters such as load factor, frequency reuse scheme and frequency hopping on system performance.
3. To investigate the impact of changing the transmission characteristics of data users like transmitted power (by base station), mobility, and transmission rate on system performance.

The thesis objective is to be achieved by simulating a GSM network model with parameter values that make the model as realistic as possible. In simulation, only downlink (transmission from base station to mobile station) will be considered since this link tends to be the limiting factor if antenna diversity is assumed at the base station [Car94]

1.2 Thesis Motivation

The increasing demand for the wireless data services with high bit rates required that ETSI standardize a new mobile radio systems for providing required hybrid transmission

services to a maximum number of network subscribers based on the current GSM system. Some of these new systems are described in chapter 4.

The main motivation of this thesis is to complement the existing knowledge and performance results for the GSM network with mostly voice traffic with a new analysis and simulation to determine the performance of the network under hybrid traffic conditions.

1.3 Thesis Organization

This document is organized into eight chapters. Chapter 2 provides a brief overview of wireless communications, propagation characteristics in different radio channel environments, a description of cellular radio system concepts, tele-traffic theory and different air interface access schemes.

Chapter 3 introduces the GSM system and describes the different aspects associated with GSM services, network architecture, and specifications for logical channels and frame structure, power control, handoff, and frequency hopping.

Chapter 4 deals with data services over the GSM network as well as the future GSM evolution in that area (i.e. third generation technology).

Chapter 5 describes the model of GSM network to be implemented. All network specifications, model assumptions, and parameters are described and justified.

A description of the used simulation tool is provided in chapter 6. A flow diagram is fully presented and the methods used to collect data and display results are described. The last section of this chapter discusses the simulations that will be performed.

Chapter 7 presents the results of the research. Chapter 8 presents conclusions as well as some recommendations for future research work in this area.

1.4 Thesis Contributions

This Thesis contains a performance analysis of a mixed traffic GSM network in different conditions with regard to some system parameters of special interest as mentioned earlier.

A simulation model was developed to evaluate the performance of the GSM network model under consideration through simulation. As far as can be determined based on a review of the literature, this research has some general contributions that can be summarized as follows:

1. Analyzing the performance of a multi-user GSM network with a variable transmission rate for data users up to 76.8 kbps under different scenarios with respect to factors such as partial loading, frequency hopping, frequency reuse plan, and transmitted power.
2. Studying the effect of different voice-to-data traffic ratios on the performance of data users.
3. Observing the impact on network performance of different probability distributions of the transmission rate of data users.

Chapter 2

CELLULAR RADIO COMMUNICATIONS

In this chapter we will discuss the propagation and multipath characteristics of the radio channel such as path loss, fading , and shadowing. The fundamentals of cellular communications will also be discussed in later sections of this chapter.

2.1 Propagation in a Mobile Communication Environment

Unlike the wired communication, where the channel is stationary and predictable, radio channel of the wireless communication is extremely random and unpredictable. The reason for this is the limitations caused by the physical channel and the mobility of either the receiver or the transmitter or both. The transmission path between the transmitter and the receiver can be a simple line-of-sight or a complicated multi-path in which signals arrive at the receiver from different directions with different powers and time delays. As a result, it is not feasible to use a deterministic approach to characterize these kinds of channels. Rather, practical measurements with statistical analysis were found to be necessary to get a proper mathematical model for the mobile channel. These models are required to quantify two variables:

- 1-Average signal strength at any distance from the transmitter.
- 2-Signal variability which characterizes the fading nature of the channel.

2.1.1 Propagation Path Loss

Path Loss can be defined as the difference between the transmitted power and the received power. The simplest way to calculate the transmission path loss is by assuming a free space propagation model which can only be accurate if the transmitter and the receiver have a clear line-of-sight path in between. This assumption can always be true in the case of satellite communications. In the Free Space Model, the received power P_r depends on the T-R distance d and the transmitted power P_t as shown in equation 2.1[Rap96]

$$P_r(d) = \frac{P_t G_t G_r \lambda^2}{(4\pi d)^2} \quad (2.1)$$

Assuming unity antenna gains G_t and G_r , the path loss in dB is given by

$$PL(\text{dB}) = 10 \log \frac{P_t}{P_r} = 20 \log(4\pi d) - 20 \log(\lambda) \quad (2.2)$$

where λ is the wavelength in meters, Equation 2.2 can be written as

$$PL(\text{dB}) = 10. \alpha \log d + C \quad (2.3)$$

Where α is called the propagation exponent which basically depends on the environment and has the value of 2 in the free space case, C is a constant independent of the environment (a function of the wavelength)

There are few cases where the free space formula is applicable. This is due to the fact that in a real propagation environment there are always some mechanisms to obstruct the signal. The three basic mechanisms are *Reflection*, *Diffraction*, and *Scattering*. Reflection can be from the surface of the earth, as well as from buildings and trees. Diffraction happens when the path between transmitter and receiver is obstructed by a surface with sharp edges. Scattering can happen if the medium consists of a large number of obstacles with small dimensions compared to the wavelength. As a result of all the above mentioned propagation

mechanisms, each environment can have a different propagation exponent which can be determined by measurement in each environment. Environments can be classified into three main classes as shown below:

- 1- Urban areas which are dominated by tall buildings, offices, and other commercial structures.
- 2- Suburban areas which contain residential houses and parks.
- 3- Rural areas which contain open farmland with farm buildings, wooded areas and forests.

Since the mathematical modeling of the propagation of signals in real environment is so complicated, many empirical models to predict the propagation loss were established by different authors. The model established by M. Hata is considered here as an example of these empirical models.

* Hata's Model

Hata has built his model in order to describe the graphical data given by Okumura. This model is applied only to a certain range of parameters. The expression for urban areas will be shown below just as an example [Gar96]

$$PL = 69.55 + 26.16 \log(f_c) - 13.38 \log(h_T) - a(h_R) + [44.9 - 6.55 \log(h_T)] \log d \quad (2.4)$$

where : $150 \leq f_c \leq 1500$ MHz : carrier frequency,
 $30 \leq h_T \leq 200$ m : transmitter antenna height,
 $1 \leq d \leq 20$ Km : T-R separation, and
 $a(h_R)$ is the correction factor for the mobile antenna height and given by:

$$a(h_R) = (1.1 \log(f_c) - 0.7) h_R - (1.56 \log(f_c) - 0.8) \text{ dB} \quad (2.5)$$

where $1 \leq h_R \leq 10$ m : receiver antenna height.

It can be noticed that the propagation exponent in Hata's model is a function of the transmitter antenna height. For example, for an antenna height of 100 m, the propagation exponent will be 3.18.

2.1.2 Small Scale Fading

Small Scale fading refers to the fluctuation of the amplitude of the signal over a short period of time or travel distance. It is caused by the interference between many versions of the signal that arrive at the receiver with different time delays, or the so called *multipath* phenomenon. The effects of the small scale fading, or simply fading can be summarized in the following:

- a- Rapid changes of signal amplitude over a small distance or short period of time.
- b- Random frequency modulation caused by the *Doppler* shift.
- c- Time dispersion by the multi-path delays.

Fading can be affected by many physical factors in the propagation channel. Some of these factors are : speed of the mobile, speed of surrounding objects, and transmission bandwidth of the signal. Fading is mostly modeled by a *Rayleigh Distribution* which is resulted as the envelope of the sum of two quadrature Gaussian signals. The Rayleigh distribution has the following probability density function (pdf) [Par89]

$$P(r) = \frac{r}{\sigma^2} \exp\left(-\frac{r^2}{2\sigma^2}\right) \quad 0 \leq r \leq \infty \quad (2.6)$$

Where σ is the rms value of the received signal power before envelope detection, σ^2 is the average power of the received signal before envelope detection, and r^2 is the instantaneous power. The corresponding cumulative distribution function (CDF) is give by

$$P(R) = \Pr(r \leq R) = \int_0^R P(r) dr = 1 - \exp\left(-\frac{R^2}{2\sigma^2}\right) \quad (2.7)$$

Rayleigh Fading can be *frequency selective* when signals with different carrier frequencies experience different fading on the same channel. Nevertheless, closely spaced frequencies will have some correlation, i.e. they will suffer more or less the same fading. As a result, carrier frequencies of different signals (users) should be sufficiently spaced to avoid

any kind of fading correlation. This leads to the so called the *coherence bandwidth* of the channel which can be defined as the bandwidth over which two signals with different frequencies are not correlated. The coherence bandwidth of two received signals is given by

$$B_c = \frac{1}{2\pi\tau_d} \quad (2.8)$$

where τ_d is the mean delay spread which has the typical values given in table 2.1 [Lee93]

Type of Environment	Delay Spread τ_d (μ s)
Open Area	< 0.2
Suburban Area	0.5
Urban Area	3.0

Table 2.1 Delay Spread in Different Environments.

2.1.3 Large Scale Fading (Shadowing)

Large scale fading is called so because it characterizes signal strength over a large T-R separation distance. It also describes the variation in the mean signal strength due to the diffraction of the signal caused by the obstacles in the physical channel. Thus, it is also referred to as shadowing.

As opposed to the rapidly changing small scale fading, shadowing is more or less a local change in the average signal strength because it has a much slower variation around its mean value. Shadowing was found to have a *log-normal* distribution which implies that the measured signal level in dB, $r(d)$, at specific T-R separation has a normal distribution about a distance dependent mean $r_m(d)$ [Rap96]. The pdf of the log normal distribution is

$$P(r(d)) = \frac{1}{\sqrt{2\pi}\sigma_r} \exp - \frac{[r(d) - r_m(d)]^2}{2\sigma_r^2} \quad (2.9)$$

Where σ_r is the standard deviation of the random variable r (signal level) in dB. σ typically has values between 5 and 10 dB based on the environment. Typically, higher values of σ are more likely to be found in heavy built environments. The probability of the received signal level

exceeding some threshold r_o is given by the following

$$Pr(r(d) > r_o) = Pr\left(\frac{r(d) - r_m(d)}{\sigma_r} > \frac{r_o - r_m(d)}{\sigma_r}\right) = Q\left(\frac{r_o - r_m(d)}{\sigma_r}\right) \quad (2.10)$$

Where

$$Q(z_o) = \frac{1}{\sqrt{2\pi}} \int_{z_o}^{\infty} \exp\left(-\frac{r(d)}{2}\right) dr \quad (2.11)$$

It is quite often important to know the relation between the boundary coverage (at cell edge) and the probability of coverage within the area of the cell which is known as area coverage $U(\gamma)$. It can be defined as the percentage of the area with received signal above or equal some certain threshold γ . For a circular cell with radius R , it is shown in [Reu74] that $U(\gamma)$ is given by

$$U(\gamma) = \frac{1}{2} \left[1 + \operatorname{erf}(a) + \exp\left(\frac{1+2ab}{b^2}\right) \left\{ 1 - \operatorname{erf}\left(\frac{1+ab}{b}\right) \right\} \right] \quad (2.12)$$

Where $a = \frac{[\gamma - r_m(R)]}{\sigma_r \sqrt{2}}$ and $b = 10 \gamma \log(e) / \sigma_r \sqrt{2}$. Using equation 2.12 it is possible to determine the threshold γ that leads to a desired area coverage for some boundary coverage.

2.2 Cellular Concepts and Fundamentals

The traditional problem faced by mobile radio system designers has been how to balance the conflicting requirements of area coverage and user capacity. These requirements are said to be conflicting because to achieve a large coverage area, a single, high powered transmitter with a high antenna mounted on a tall tower must be used. Although a good coverage can be achieved by this approach, it is impossible to reuse the same allocated frequencies except in very distant location because of interference.

In 1970's, this scenario allowed Bell Mobile systems in New York city to support just 12 simultaneous calls. Taking into consideration the limited spectrum allocated by governments, it was a logical evolutionary step to come up with the cellular concept. It implies that instead of having a large area covered by a single transmitter, that area can be divided into smaller

coverage areas called cells, each with a low powered transmitter. This way, it is possible to reuse the same frequencies in different cells. In conclusion, maximizing the number of times each frequency may be used in a given geographical area while keeping the interference level within tolerable limits is the key to an efficient cellular system design.

2.2.1 The Frequency Reuse

Frequency reuse is the fundamental target of cellular radio systems. For design purposes, the area to be covered should be divided into cells in a regular fashion, therefore, cells need to be

Fig 2.1 Frequency Reuse Concept When $N = 3$.

represented by a regular polygon. The Hexagon was found to be an ideal choice. Each cell is assigned a group of radio channels which is completely different than those of the neighboring cells. The cells will form sets called *clusters* where cluster size N can take any value that satisfies the equation

$$N = i^2 + j^2 + ij \quad (2.13)$$

For any integer i and j . Figure 2.1 shows the case when cluster size is 3 where cells with the same number use the same group of channels. These cells are called *co-channel* cells. It is shown in [Gar96] that

$$N = \frac{D^2}{3R^2} \quad (2.14)$$

Where D is the distance between the centers of any two co-channel cells (i.e. having same number), and R is the cell radius.

2.2.2 Channel Allocation Strategies

A variety of channel allocation schemes have been developed to achieve an efficient utilization of the radio spectrum. These schemes can be divided into two main classes, *fixed* or *dynamic*. In a fixed channel allocation, a predetermined group of channels is assigned to each cell in a way that matches the frequency reuse plan made in earlier design stage. This scheme implies that calls within any cell can only be served by channels assigned to that cell.

In a dynamic channel allocation, all available channels are put in a pool administered by the MSC. Each time a call is made, the serving cell asks for a channel from the MSC following a certain algorithm and returns it back when call ends. This way, all available channels are accessible to all of the cells [Tek91].

2.2.3 Interference and System Capacity

Interference is one of the major limiting factors in the performance of cellular radio systems. Interference can be mainly classified as *co-channel* or *adjacent channel*. Co-channel interference is generated when the same frequency is being used by more than one user (in different cells) at the same time, which is just the case with the frequency reuse approach. On the other hand, adjacent channel interference results from signals assigned to adjacent frequencies because of the imperfect receiver filters. A better control of the adjacent channel interference is possible with the advanced Digital Signal Processing (DSP) technology and careful channel assignment by avoiding to assign channels that are all adjacent in frequency to the same cell. In referring to Figure 2.1, the ratio $q = \frac{D}{R}$ is used as a key parameter in analyzing the co-channel

interference. q is called the Co-channel Interference Reduction Factor (CIRF) related to the cluster size N by the following

$$q = \frac{D}{R} = \sqrt{3N} \quad (2.15)$$

To achieve a reliable communication, an acceptable signal level as well as an acceptable signal to interference ratio (CIR) are required. High value of q can lead to the required quality due to the low level of co-channel interference. On the other hand, this provides a smaller capacity since a high q value means a high cluster size N , therefore, a tradeoff should be made.

Considering Figure 2.1 once again, it will be noticed that there are six first tier co-channel cells. Neglecting other tiers, and assuming an equal distance to all six co-channel cells D , as well as an equal transmitted power from all transmitters, worst case scenario of Carrier to Interference Ratio, CIR, can be given by

$$\text{CIR} = \frac{R^{-\alpha}}{6D^{-\alpha}} = \frac{1}{6}q^{\alpha} \quad (2.16)$$

Where α was defined before as the propagation exponent [Lee86]. For a radio system, it is normal to specify a minimum required CIR_{req} threshold value based on some subjective tests.

This will lead to q value given by

$$q = [6 \text{ CIR}_{\text{req}}]^{1/\alpha} \quad (2.17)$$

This q value determines the minimum cluster size N that maintains the required quality level, which in turn gives the maximum possible capacity of the radio system as mentioned before.

2.2.4 Handoff and Power Control

Handoff (or Handover) is the process of reallocating the mobile user's voice and control signals to channels belonging to new base station (i.e cell) while the call is still in progress. There are many situations in which handoff is necessary:

- 1- When received signal strength or quality (CIR) drop below certain thresholds. This might be caused by a large T-R separation distance, or by a deep fading and is the most necessary handoff case to avoid call dropping.
- 2- When the interference level of other users can be improved by performing handoff.
- 3- When the blocking probability in some cell is higher than the required Grade Of Service (GOS) -even with an acceptable signal quality- and can be improved by moving some calls to other cells [Joh95].

The decision to perform handoff can be made either by the base station and MSC, or by the mobile itself as in Mobile Assisted Handoff (MAHO).

Power Control is similar to handoff in its objective to maintain the quality of service and overall performance but differs in the approach. While handoff changes the cell of the mobile, the power control mechanism works to ensure that each transmitter (mobiles and base stations) transmits the smallest power necessary to maintain the required quality in both forward (Mobile to BS) and reverse (BS to Mobile) links. In addition to improving the interference level, power control helps to extend the mobile's battery life.

2.2.5 Frequency Hopping

Frequency Hopping is a periodic change of the carrier frequency of the transmitted signal over a predefined set of frequencies called the *Hopset*. Time duration between hops is called a hop duration or hopping period. It has same duration as the frame duration which equals 4.615 ms in the case of GSM.

Frequency hopping can be classified as fast or slow. Fast hopping has a hopping rate that is higher than the information symbol rate, while the hopping rate is lower than information symbol rate in the case of slow frequency hopping.

In GSM specifications, frequency hopping is also divided into two schemes, *Random* or *Cyclic*. These schemes will be described in more detail later on.

2.3 Teletraffic Theory

Teletraffic Theory, or what also known as *Trunking Theory*, is the basic theory to serve a large number of users in a system that has limited resources using the sharing concept and the statistical behavior of users. This is just the case in cellular systems where a limited number of channels are available to accommodate an increasing number of subscribers. This theory was first developed by Erlang of Denmark in the 19th century. This is the reason the measure unit of traffic intensity bears his name. One *erlang* is the traffic intensity in a channel that is continuously occupied. Any consideration of traffic theory must include two main quantities, *Load* and *Grade of Service* GOS. Load is a dimensionless quantity measured in erlang and equals the product of the call request rate of the user λ and the call duration $(1/\mu)$. Total load is the summation of loads of all users in the system. GOS is a measure of the system congestion during the system's busiest hours. It can be expressed by the probability of a call being delayed beyond a certain time (as in Erlang C), or the probability of a call being blocked (as in Erlang B). Load and GOS are related through many formulas developed by Erlang. Erlang B formula gives the probability of call blocking and is given by the following [Par89]

$$Pr(\text{call blocking}) = \frac{\frac{A^C}{C!}}{\sum_{k=0}^C \frac{A^k}{k!}} \quad (2.18)$$

Where A is the total traffic load in erlang and C is the number of trunked channels. Erlang C formula gives the probability of a call experiencing a delay as

$$Pr(\text{call delay}) = \frac{A^C}{A^C + C!(1 - \frac{A}{C}) \sum_{k=0}^{C-1} \frac{A^k}{k!}} \quad (2.19)$$

Thus, the probability of a call being delayed for a time greater than t was found to be

$$Pr(\text{delay} > t) = \exp [-(C-A)\mu t] \quad (2.20)$$

Erlang B and C formulas were tabulated for various values of GOS and a number of channels.

2.4 Multiple Access Schemes

Multiple access techniques are required to enable a wireless communications service provider serving many users simultaneously with a limited allocated spectrum in a way that achieves the highest possible capacity while maintaining the required quality. Duplexing is the feature of the wireless system that allows the user to send and receive information simultaneously. This can be done by either *Frequency Division Duplexing* (FDD), where forward and reverse channels are assigned to two split simplex channels, or *Time Division Duplexing* (TDD), where time is used to provide both the forward and reverse channel.

The three main multiple access schemes are *Frequency Division Multiple Access* (FDMA), *Time Division Multiple Access* (TDMA), and *Code Division Multiple Access* (CDMA). These schemes are briefly described in the following [Rap96].

2.4.1 Frequency Division Multiple Access (FDMA)

In FDMA system, signals from various users are assigned different frequencies, so that each user is allocated unique channel on demand unshared by other users, as in figure 2.2. FDD is usually used with FDMA.

Fig 2.2 FDMA

Guard bands are required between adjacent channels to minimize cross talk. The number of the simultaneously available channels in the system is

$$N = \frac{B_t - 2B_g}{B_c} \quad (2.21)$$

where B_t is the total allocated frequency band, B_g is the guard band at the edge of the allocated band, and B_c is the channel bandwidth (200 kHz in GSM).

2.4.2 Time Division Multiple Access (TDMA)

In a TDMA system, each user transmits his/her data in a dedicated time slot that reoccurs every frame which is made up of several time slots (8 in GSM). Therefore, the transmission of each user is noncontiguous. During this time slot, the user accesses the whole frequency band allocated to the system (*Wideband* TDMA), see figure 2.3, or only part of the band (*Narrowband* TDMA), as in figure 2.4.

Fig 2.3 Wideband TDMA

The frame Duration should not be too long nor too short, since when long frames are used, the user will feel the delay caused by the discontinuity of transmission, while too short frames will not be capable of providing high bit rate with the required capacity.

TDMA can be used with either FDD or TDD. The number of users that can be supported by TDMA system is given by

$$N = \frac{m(B_t - 2B_g)}{B_c} \quad (2.22)$$

where m is the number of time slots per frame. Guard bands are required to prevent users at the edge of the spectrum from bleeding over into adjacent radio services.

Fig 2.4 Narrowband TDMA.

2.4.3 Code Division Multiple Access (CDMA)

The CDMA system uses spread spectrum techniques to spread the bandwidth of the transmitted signal over a spectrum much larger than the bandwidth of the signal. The signal is spread by a Pseudo-Noise (PN) code where a different code is assigned to each user. In CDMA, all users use the same frequency and can transmit simultaneously with a frequency reuse factor of one, Figure 2.5. That is the reason the CDMA system claims a higher capacity when compared with FDMA and TDMA. On the other hand, the CDMA system is more sensitive to interference and requires a robust power control technique to overcome this problem.

CDMA system can support number of users given by

$$N = 1 + \frac{W/R}{E_b/N_o} \quad (2.23)$$

where W is the total bandwidth, R is the baseband information bit rate, and E_b/N_o is the signal to Noise ratio SNR at the base station. W/R is called the processing gain.

Chapter 3

THE GSM SYSTEM

The first cellular radio system in Europe was in Scandinavia in 1981 and served only a few thousand subscribers. A few years later, six different and incompatible cellular systems were operating in 16 European countries to serve more than 1.2 million subscribers. However, it was impossible for a mobile to roam across the continent.

In 1982, the Group Special Mobile (GSM) committee of the Conference Europe'enne des Postes et Te'l'e'communication (CEPT) was assigned the task of specifying a pan-European public mobile communication system in the 900 MHz band. In 1991, after years of discussions, proposals, trials, testing, and validation, GSM was first introduced into the European market. A few years later, many countries in Asia, the Middle East and Australia had adopted GSM. At this time the new name "Global System for Mobile communications" was coined, and the European Technical Standard Institute ETSI began to be responsible for setting GSM standards. This chapter will focus on the services, network architecture, and specifications of the GSM system.

3.1 Services and Features

A communication services supported by any system are defined as a group of communication capabilities that a service provider offers to the subscriber who sees this group of services worth paying for. GSM services follow ISDN guidelines and are divided into three main classes: (1) Teleservices, (2) Bearer services, and (3) Supplementary services [Meh97].

Teleservices include regular telephony and emergency calls, in addition to voice messaging, Facsimile, Videotex, and Teletex.

Bearer services give the user the capacity required to transmit signals between certain access points. These services include packet switched protocols and data rates from 300-9600 bps. A connection can be made to other mobile stations or other data users on a circuit switched or packet switched network through the use of a data terminal or a computer connected to the mobile station, and a Packet Assembler/Deassembler (PAD) connected to the network. Bearer services also support the speech and data swapping during a call as well as the use of Automatic Repeat request (ARQ) in order to improve error rates.

The GSM supplementary services are defined as add-ons, additional features to both teleservices and bearer services. It can be summarized in: Number identification services, Calling Offering services, Call Completion services, Multiparty services, Community of Interest services, and Call Restriction services. All of these services are digital in nature and not available in analog mobile networks.

The most remarkable features of GSM are the *Subscriber Identity Module* SIM, and the on-the-air privacy feature. SIM is a memory device that stores user and network information without which all mobiles are identical and non-operational. Using SIM helps in making the distinction between subscriber identity and mobile identity, which enables the GSM network to route calls and perform billings based on the identity of the subscriber rather than on the identity of the mobile unit used.

Unlike the case with an analog systems, call privacy can be achieved in digital systems by encrypting the digital bit stream sent by a transmitter.

3.2 Network Structure

The three major subsystems of the GSM network Architecture are *Radio or Base Station Subsystem BSS*, *Network and Switching Subsystem NSS*, and *Operation Support Subsystem OSS*. Figure 3.1 [Ste92].

The BSS includes the equipment and functions related to the management of connections on the radio path including the management of handoff. It consists of many Base Station Controllers BSCs, each of which controls many base Transceiver Stations BTSs. A BSC controls such functions as handoff and power control, while the BTS performs the transmitting functions.

Fig 3.1 GSM system structure

Generally, the BSS is associated with the radio channel management, transmission functions, and radio link control.

The NSS includes the equipment and functions related to end-to-end calls, management of subscribers, switching, and communicating with other networks such as ISDN and PSTN. Units included in the NSS are the *Mobile Switching Center MSC*, *Home Location Register HLR*, *Visitor Location Register VLR*, *Authentication Center AUC*, and *Equipment*

Identity Register EIR. MSC is the central unit of the NSS, which provides call setup, routing, switching, handoff, and other functions such as billing. HLR is a centralized database that contains subscriber information and location information of all the users residing in the area of the MSC. VLR is a database of all roaming mobiles in the area of the MSC but not residing there. AUC is a database that provides HLR and VLR with authentication parameters and ciphering keys required for security purposes. EIR is a database that includes numbers of all registered mobile units.

The OSS main unit is the *Operational and Maintenance Center* OMC. OMC includes the equipment through which the service provider can monitor, troubleshoot, and control all aspects of the GSM system.

The Mobile Station MS is the equipment used by the subscriber to access the service offered by the system. It is traditionally considered to be part of the BSS although it is one end of the conversational path. The primary function of the MS is to transmit and receive voice and data over the air interface of the GSM system.

3.3 Channel and Frame Structure

The GSM system operates in two paired bands each of 25 MHz : 890-915 MHz for uplink or reverse link where the mobile transmits to the base station, and 935-960 MHz for the downlink or forward link where the transmission is from the base station to the mobile. These two bands are divided into 200 KHz wide channels. Taking out 200 KHz as a guard band at the lower end of each band will leave 124 paired duplex channels with 45 MHz spacing. Each of the 124 channels operates at different carrier frequency.

Figure 3.2 shows the composition of a GSM time slot. Most of the transmission time of the slot is reserved for two bursts of 58 bits each. 57 of these bits are presenting the user

information and the other bit is a flag which indicates whether speech or any other kind of information is being transmitted. There is a 26 bit sequence for equalizer training in the middle of the slot. The slot begins and ends with 3 tail bits, and finally, 8.25 bits are needed as a guard time. The total of 156.25 bits are transmitted in each time slot of 577 μ s duration, therefore, the aggregate transmission rate is 270.833 kbps [Goo91].

Fig 3.2 GSM time slot structure.

Figure 3.3 shows a GSM multiframe of 120 ms duration divided into 26 frames, each containing eight time slots. In the multiframe, 24 frames out of the 26 are for user information carried into traffic channels, while the other two are reserved for control information carried through control channels.

Fig 3.3 GSM multiframe and frame structure

51 multiframes are grouped into a *superframe*. A *hyperframe* contains 2048 superframes and has a duration of 3 hours, 28 minutes and 54 seconds.

A combination of a time slot number and a carrier frequency number form a physical channel number that will be assigned to a user during the call. There are two types of GSM logical channels, *Traffic Channels* TCH which carry user speech or data, and *Control Channels* CCH which carry signaling commands between the mobile and the base station [De'c93].

TCH can be either *full rate* TCH/F where user data is sent in one slot per frame, or *half rate* TCH/H when the same time slot is shared by two users sending alternately. TCH/F allows for a data transmission rate of 9.6, 4.8, or 2.4 kbps while TCH/H provides rate of 4.8 or 2.4 kbps.

CCHs consist of three main control channels : *Broadcast Channel* BCH, *Common Control Channel* CCCH, and *Dedicated Control Channel* DCCH. The BCH is used to send cell identity, cell service available, synchronization of frequency and time. These functions are done through the use of three types of the BCH : *Broadcast Control Channel* BCCH, *Synchronization Channel* SCH, and *Frequency Correction Channel* FCCH. BCH is a forward link channel that occupies only the first time slot of certain GSM frames.

There are three CCCH types: *Paging Channel* PCH which is a forward link channel used to notify the MS of incoming calls, *Random Access Channel* RACH is a reverse link channel used by the MS to originate a call or acknowledge a page. The third type of CCCH is the *Access Grant Channel* AGCH which is a forward link channel used by the base station to provide a link communication to the MS. Similar to the BCH, CCCH occupies the first time slot, however, CCCH uses those frames not used by BCH.

Finally, DCCH are bi-directional and can exist in any time slot except those of the BCH. The three types of the DCCH are the *Stand Alone Dedicated Control Channel* SDCCH which is

needed to provide the signaling services required by the user, *Slow and Fast Associated Control Channels* SACCH and FACCH are used to provide supervisory signaling during the call.

3.4 Power Control and Handoff

Power control is required to maintain the quality of link between the base station and the mobile station during the call by decreasing the interference level through reducing transmitted power to the smallest level required to provide the desired link quality. When power control fails to improve the link quality to an acceptable level, the network operator will work to handoff the call to an adjacent cell chosen following a number of measurements done by either the BSS or the MS or both. If neither power control nor handoff succeed to improve link quality, the call will be lost and we will have what is known as a *Call Drop*.

Power control can be implemented in both the forward and the reverse links independently of each other. Although power control can improve the performance and increase the spectral efficiency of the GSM system, it is up to the network operator to decide whether to adopt it or not. Furthermore, the exact strategy of power control is decided by the network operator to achieve the desired quality of service.

Handoff can be performed between channels of the same base stations (i.e. change sectors in the same cell), or between channels of different base stations (i.e. different cells). As mentioned before, a handoff decision is usually the result of an unacceptable quality level in most cases, however, the decision to hand a call off might be based on the distance from the MS to the neighboring base stations. In other word, the MS will be assigned a channel belongs to the nearest base station.

In the case of a quality based handoff, some measured parameters are compared with their associated threshold values and if any of these parameters are found to be below the

corresponding threshold, power control will be attempted to meet the desired level, otherwise, handoff will be performed, see figure 3.4 [Ste92]. The parameters used in the power control and handoff are the received signal RXLEV and the received quality RXQUAL in both forward link (measured by MS) and reverse link (measured by BS). In addition, the MS keeps track of RXLEV and RXQUAL from all neighboring cells. One value of each of these parameters is measured in every SACCH multiframe (i.e. 104 frames = 0.48 sec). To avoid premature use of power control or handoff, averaging of the above parameters is carried out for 32 measured samples over a 15 sec duration.

Fig 3.4 Power Control and Handoff Algorithm in GSM

3.5 Frequency Hopping

Frequency hopping is the process of changing the frequency of the transmitted signal at regular intervals. Hopping can occur over a maximum of 64 frequencies, however, and due to limited allocated spectrum for each service provider, much less frequencies are available for hopping. These frequencies should be sufficiently separated to provide enough decorrelation.

Although frequency hopping is optional for each cell separately, it is very important to counteract the selective frequency fading characteristic of the radio channel where one channel may experience a deep fade while another channel is enjoying a good communication quality. This feature of frequency hopping is called *frequency diversity*, which can increase the efficiency of the error correction process if it is combined with interleaving. Frequency hopping can also provide *interference diversity* because interference levels differ from one channel to another, therefore, changing the carrier frequency in every burst will allow any receiver to spend only a fraction of the time on a high interference channel.

Frequency hopping can be classified in different ways. For example, it can be *Slow SFH* if a different frequency is used in each burst (TDMA frame), or *Fast FFH* where more than one change of frequency occurs per symbol duration. Another way to classify frequency hopping is by the algorithm or the fashion with which the carrier frequency is changed from one burst to another. *Cyclic hopping* involves hopping through the assigned frequency set one by one from first to last in an ordered fashion. Whereas the *Non Cyclic hopping* involves random selection of a frequency from the set, thus, it is also called *Pseudo Random hopping*. It is worth noting that no hopping is allowed to take place in the first time slot of the BCCH frequency.

Frequency hopping can be effective for stationary or slow moving mobiles with a high correlation within a radio channel from frame to frame. However, fast moving mobiles suffer a

little or even no correlation between frames which reduces the effect of FH. Effect of FH on system performance can be seen in figure 3.5 for cyclic FH, and in figure 3.6 for random FH where different frequency set sizes were tested. Set size has a noticeable impact on the link performance. It was shown in [Joy97] that increasing the frequency set size beyond 8 will cause

Fig 3.5 Link performance for cyclic FH[Mog95]

Fig 3.6 Link performance for random FH [Mog95].

negligible gain to the performance. This might be due to the fact that a bigger set size reduces the frequencies separation which in turn reduces the decorrelation which the FH aimed to make.

The parameters of FH are the following [Meh97] :

- N : number of available hopping frequencies.
- MAIO : $0 \leq \text{MAIO} \leq N-1$, used to distinguish between different frequencies in a cell.
- HSN : $0 \leq \text{HSN} \leq 63$, all mobiles in a cell have the same HSN but each with different MAIO.
- FN : TDMA frame number.
- MA : set of available hopping frequencies.
- NBIN : number of bits required to represent N .
- RNTABLE : look up table of 114 integers.
- MAI : index of the frequency to be used in the next burst, MAI is to be returned by the hopping algorithm.

GSM FH algorithm is shown in figure 3.7. It can be seen that if HSN equals zero, cyclic hopping is implemented, otherwise, random hopping is implemented. Synchronization parameters T1, T2, and T3 are needed to determine the superframe number in the hyperframe,

Fig 3.7 FH algorithm in GSM.

the multiframe number in the superframe, and the frame number in the multiframe respectively.

3.6 GSM Air Interface Technical Specifications

Air interface specifications of GSM are summarized in table 3.1 below [Cox92].

Access Technology	TDMA/FDD	Speech Coding	RPE-LTP
Frequency Band (Eur) Uplink	890-915	Speech Rate (kbps)	13
Downlink	935-960	Users Per Frame	8
RF Channel Spacing (KHz)	200	Channel Bit Rate (kbps)	270.833
Frequency Assignment	Dynamic	Frame Duration (ms)	4.615
Modulation	0.3 GMSK	Channel Coding	$\frac{1}{2}$ rate Convolutional
Power Control	Yes	Interleaving (max Delay in ms)	40

Table 3.1 Air Interface Specifications of GSM.

Chapter 4

DATA IN THE GSM SYSTEM

The first generation of cellular systems were mainly concerned with providing the service of speech signal transmission, however, they were able to transmit data at low bit rates. All these systems were analogue since speech signals were not digitally encoded. Second generation systems are all digital with telephony as the primary service and also provide some data applications. For example, a GSM system provides a very good facsimile service, a two – way paging function or short message service SMS which allows up to 160 text characters to be sent, and Cell Broadcast service CBS which can broadcast messages in a given geographical area. Generally, GSM offers data services with a throughput up to 9.6 kbps. These days, much more data applications are required in wireless systems especially after the rapid growth in the number of internet users. As a result, third generation of GSM and other cellular systems are targeting to provide high quality data services with a higher bit rate for the maximum number of users at low cost. In this chapter, data services and applications in the GSM system will be introduced. Future evolution of data services in GSM will be discussed in the second section of the chapter.

4.1 GSM Data Applications and Services

After the success of the telephony service of cellular systems, data communication services have become popular. However, the success of data services depends mainly on the availability of end user mobile applications. Mobile data applications can be military or civilian

such as industry, trade, transportation, and distribution (ITTD). Other applications might be in the health care field. Using mobile data applications can reduce the paper flow, save time, and improve the accessibility and accuracy of information. A research was carried out in [Lut98] to study the factors effecting the success of mobile data applications, these factors were found to be as follows:

- Processes supported by data applications should be dynamic, otherwise, fixed network or paper communication can offer a simpler and cheaper alternative.
- Information related to supported processes should be electronically available.
- Mobile handset should be small, flat, light, easy to handle, and sufficiently sturdy.
- Mobile data applications should have error-free functioning.

GSM data services can be either *transparent* or *nontransparent*. Transparent service offers a consistent delay, 9.6 kbps throughput with variable error rate. Nontransparent service offers a consistent low error rate and variable delay at the same throughput with a built in error correction technique. Some of the data services provided by the GSM system are:

1- Internet

By dialing the internet service provider number, a user can connect with a transmission rate of 9.6 kbp and up. GSM offers a robust wireless data service because it is a digital system.

2- Mobile Fax

At any place where the GSM service is available, even away from office, a GSM user can receive and send faxes just by connecting his/her laptop computer to the GSM phone handset.

3- Secure Corporate LAN Access

The digital feature of GSM facilitates in encrypting the air link. This provides security for confidential emails, files, or faxes. This way, GSM can provide secure access to its user's

corporate LAN.

4- Cell Broadcast

Cell broadcast is a short text service to broadcast messages that can be received by all users within a particular area. This service might be suitable for information type messages like road traffic and weather conditions.

5- Short Messages

160 characters limited messages can be transferred between mobiles and the short message service center SMSC. Messages can be bi-directional where messages sent to the mobile are called *Mobile Terminated*, and messages sent from the mobile are called *Mobile Originated*. Mobile terminated messages may be displayed either by mobile itself or on a connected terminal device. On the other hand, Mobile originated messages can be sent from mobile's keypad or from a terminal device.

4.2 Future GSM Data

As was mentioned earlier, the current GSM system is capable of providing data service with a throughput up to 9.6 kbps. Several evolutionary steps are already in the final stages of standardization to provide services with a much higher bit rate. A third generation system UMTS is being standardized in a parallel process to provide the top of line multimedia services with higher capacity.

Different techniques have been proposed for the evolved GSM systems (GSM generation 2+) to achieve the goal of increasing the bit rate. The first technique is to use *Multi-Modulation* where Higher Level Modulation (HLM) schemes are used. This technique is adopted in the *Enhanced Data rates for Global Evolution system* EDGE. *High Speed Circuit-Switched Data system* HSCSD is using the *Multi-Slot* technique in which more than one time slot per frame can

be assigned to a user. EDGE and HSCSD are designed based on the current GSM slot structure. On the other hand, General Packet Radio Services system GPRS has proposed to adopt the concept of sharing radio resources for bursty data transmission.

With a little modification to network and base stations, these evolutionary proposals will fit in with the current GSM system structure. Furthermore, they can also provide a bridge toward the third generation system UMTS, as shown in figure 4.1.

Fig 4.1 GSM Evolution towards 3rd generation.

4.2.1 Enhanced Data Rates for Global Evolution EDGE

EDGE was first proposed early in 1997. Less than a year later, a feasibility study was approved by ETSI to give the green light for a standardization process which is still on going. Although EDGE uses the GSM frequency band and time slot structure, nevertheless, EDGE was found to be a generic air interface that can be used with other cellular systems in a way to facilitate an evolution towards third generation systems [Fur98].

The concept used in EDGE is to utilize a higher level modulation. This involves replacing GMSK by one of the OQPSK or O16QAM modulation schemes combined with four different coding schemes. Eight different combinations result, ECS-1 to ECS-4 with O16QAM,

and ECS-5 to ECS-8 with OQPSK. These combinations have different code rates as shown in table 4.1. High level modulation implies a change in the content of the EDGE normal burst compared to that of GSM as shown in figure 4.2.

EDGE packet is divided into a number of Radio Link Control (RLC) blocks. An RLC block has 20 ms duration with an information content of 224 to 1304 bits depending on the applied modulation and coding scheme. EDGE has an error detection capability by means of a Cyclic Redundancy Check (CRC) within the RLC block. Based on that, the receiver decides whether or not to ask for a retransmission of RLC block.

Scheme Name	Modulation Used	Gross Bit Rate (kbps)	Code Rate	Radio Interface Rate (R) kbps
ECS-5	OQPSK	32.4	0.35	11.2
ECS-6	OQPSK	32.4	0.45	14.5
ECS-7	OQPSK	32.4	0.52	16.7
ECS-8	OQPSK	32.4	0.70	22.8
ECS-1	O16QAM	65.2	0.51	33.0
ECS-2	O16QAM	65.2	0.63	41.0
ECS-3	O16QAM	65.2	0.74	48.0
ECS-4	O16QAM	65.2	1.0	65.2

Table 4.1 Radio interface rates for EDGE modulation and coding schemes

GSM

3	58	26	58	3	8.25
T	Traffic	Training	Traffic	T	G

EDGE

2	82	28	82	2	12.3
T	Traffic	Training	Traffic	T	G

Fig 4.2 Normal burst for GSM and EDGE (in symbols)

Link Adaptation is the key feature of the EDGE system. It makes it possible for the multi-modulation concept to provide a significant increase in performance over the use of only one mode. Link adaptation ensures that the most efficient modulation and coding scheme is used at any moment. This adaptively selected scheme achieves the highest throughput among the available eight schemes ECS-1 to ECS-8 according to a variable (over time) link quality C/I .

The throughput S_n of each scheme n is given by

$$S_n = R_n [1 - \text{BLER}_n(C/I)] \quad (4.1)$$

Where R_n is the radio interface rate in table 4.1, and BLER_n is the block error rates found in [Sch98] and shown in figure 4.3. Figure 4.4 illustrates the link adaptation concept. Link adaptation should be as accurate and as fast as possible.

Fig 4.3 BLER vs C/I for different schemes.

Fig 4.4 Ideal link adaptation in EDGE.

4.2.2 High Speed Circuit-Switched Data HSCSD

The increasing use of the internet has led to a growing demand for data services with a higher bit rate. The ETSI responded to this demand by standardizing a new GSM data system HSCSD, that can provide a service with a transmission rate comparable to that of ISDN. The HSCSD system was proposed and specified in 1997. It was first implemented in a real GSM network under laboratory environment to offer multimedia services to mobile users as part of the MOBILE Media ENTertainment Services (MOMENTS) project.

The main idea behind HSCSD is that multiple TDMA traffic time slots TCH/F are allowed to be assigned to one data user during radio connection instead of a single slot as in GSM. This approach increases the bit rate linearly with the number of assigned TCH slots. For example, allocating all the eight time slots of the TDMA frame to a data user increases the bit rate up to 76.8 kbps instead of the 9.6 kbps rate offered by allocating one slot. New channel coding uses punctured coding with reduced overhead to improve the time slot bit rate up to 14.4 kbps. This implies that HSCSD can provide a data speed of up to 115.2 kbps[Cal98].

HSCSD is an attractive option for both end users and service provider. The end user can enjoy a higher transmission rate, while the service provider can offer many services with different user rates without the need to change the physical infrastructure of the current GSM network. Nevertheless, new functionality is required in the network and the mobile set to combine and split data into n different streams, each of which will be transmitted via one time slot out of the allocated n slots.

There are two HSCSD configurations, *Symmetric* and *Asymmetric*. Symmetric configuration involves an equal number of time slots for uplink and downlink. The asymmetric configuration provides more downlink slots than uplink. This type is suitable for internet

applications which is mainly a data download process.

Different traffic policies were proposed for HSCSD, three of which were considered in [Cal98] as follows:

1- The Call is blocked and cleared if the required number of time slots is not available. This policy allocates Maximum Capacity (MC).

2- Allocate maximum available time slots. This number can only be changed (increased or decreased) if the call is handed over to another base station. This policy is called No Rate Adaptation (NRA).

3- Same as the second proposal with the minor exception of the ability to increase the number of slots during a call which was not originally allocated all of its required slots. This is the reason this policy is called Rate Adaptation (RA).

All the slots assigned to a call need to be on the same carrier frequency because of some physical constraint. However, these slots need not be consecutive on their carrier.

4.2.3 General Packet Radio Services GPRS

In oppose to current primarily circuit-switched technologies, GPRS is a packet-switched system which has many important features over the current GSM system. The primary advantage of GPRS is the low charge to be paid by the end user. This is because the user only needs to pay for what he transfers instead of paying for the call duration which is much longer than the actual time used to transfer user information since many applications have idle periods during the call.

GPRS user can enjoy more advantages such as the high transmission rate in the 100 kbps range which can be achieved through the adoption of the multi slot concept. This can provide multimedia services including voice, data, graphics, and video services. Furthermore, GPRS supports the widely used internet communication protocols, Internet Protocols IP and X.25. This

feature helps GPRS to be seen as a wireless extension of the existing internet networks.

As a packet switched system, GPRS has an important feature from the perspective of the service provider. This is because GPRS uses the available radio link only for the duration in which data is being transmitted or received. This implies that multiple users can share the available radio bandwidth. This is in contrast with the circuit- switched system in which each user is allocated a dedicated channel for the entire call, whether data is being sent or not. Therefore GPRS service providers can serve more users with their own spectrum[Rys98].

The standardization process of GPRS system is carried out by the ETSI committee GSM SMG4. Field trials are expected in 1999 and deployment is planned to begin in the year 2000. Most GSM vendors such as Alcatel, Ericsson, Motorola, Nokia, and others have been active in developing required infrastructure elements.

The GPRS network is shown in figure 4.5. Unlike the circuit-switched data calls which are forwarded from and to the MSC, GPRS data packets are forwarded from the base station to the *Serving GPRS Super Node* SGSN. Every network can use multiple SGSN's. The SGSN node communicates with the *Gateway GPRS Super Node* GGSN, which maintains the connection with other networks. Only one GGSN is required for each external network. Packets sent from an external network (e.g. internet) to the mobile station are received by the GGSN and then forwarded to the SGSN which in turn sends them to the required mobile station. A specialized protocol called the GPRS Tunnel Protocol GTP, operates on top of TCP/IP to help forward the packets between the SGSN and the GGSN.

Fig 4.5 The GPRS network architecture.

Chapter 5

DESCRIPTION OF THE GSM SYSTEM SIMULATION MODEL

In this chapter, the GSM system simulation model to be implemented is thoroughly described and all related assumptions are clearly stated and justified. The implemented model needs to be as realistic as possible in order for the results to be reliable. However, real systems are so complicated in a manner that necessitates having some assumptions to simplify that system, which enable research people to carry out the task of simulating these systems. The simulation is used in the thesis to analyze the downlink performance of a data mobile users in a multi-user hybrid GSM network that provides voice and data services.

5.1 Cells & Base Stations Grid

In order for the model to be implemented in a simulation program, the geographical area under consideration should be limited. That was the reason to have only 31 cells, where each cell has a 2 km radius, as shown in figure 5.1. Therefore, the geographical area of interest is only 249.415 km^2 (18 km x 13.856 km). Cells are assumed to be hexagonal and the base station of each cell is located at the center of the hexagon. A wrap around grid has not been considered and as a result, mobiles leaving the area of the 31 cells are assumed to be out of the coverage area of the simulated network and might be handed over to another MSC. However, in order to eliminate the border effect, simulation results have been obtained only from the seven central cells of the 31-cell network, namely, cells number 11, 12, 15, 16, 17, 20, and 21. These cells are exposed to

the highest interference level among all cells, therefore, the worst case has been assumed. 120° sectorization is also assumed, i.e. each cell consists of three sectors no matter what frequency reuse pattern is being used.

Fig 5.1 Cells grid of the GSM model.

5.2 Frequencies used & Frequency Reuse Plan

Although 124 carrier frequencies are allocated for GSM communication, it can be assumed that service provider has much less than all the 124 frequencies. It is common to have more than one service provider operating in the same geographical area, where the 124 frequencies are divided into smaller groups, one for each service provider. A subset is reserved for signaling and control purposes.

In the model under consideration, 15 carrier frequencies are allocated to each cell regardless of the used frequency reuse plan. Therefore, each sector has 5 frequencies, one as a control or broadcast channel BCCH, and the remaining four are for traffic channels TCH. As in GSM specifications, each carrier frequency has 8 time slots per frame. Frequencies

are divided between cells and sectors in a way that decreases the correlation and reduces adjacent channel interference.

Table 5.1 shows the frequency allocation scheme applied through out this thesis. This scheme implies that fixed allocation is considered where each cell can only use its allocated frequencies, as opposed to the dynamic allocation strategy in which frequencies are not allocated to cells permanently. The GSM operator uses only one or two time slots of the BCCH carrier frame for control while the other slots can be used for TCH. In the model in hand, two time slots are dedicated to BCCH. There are 38 TCH channels ($(5 \times 8) - 2$) per sector, this large number of channels has been chosen to observe the system behavior in high traffic conditions.

Sector No	Allocated frequencies				
1	1	4	7	10	13
2	2	5	8	11	14
3	3	6	9	12	15

Table 5.1 Frequency allocation scheme for 15 carriers.

Two frequency reuse plans have been investigated in this thesis, 1/3 reuse plan (i.e. cluster size of 1 with each cell having 3 sectors) was first considered because it was found in [Joh95] that the 1/3 plan has a maximum capacity when compared with some other reuse plans. An almost similar conclusion was reached in [Car94]. The 3/9 plan was then assumed and the improvement in system performance over the 1/3 plan was observed.

5.3 Propagation Considerations.

As mentioned in section 2.1, a transmitted signal in a radio channel is subjected to a

propagation path loss directly proportional to the propagation exponent α , shadowing, and Rayleigh fast fading. The Rayleigh fading runs about a local mean level determined by shadowing which has a log normal distribution about an area mean level that is set by the path loss [Jak74]. A simple model for the propagation path loss is used and given by equation 5.1.

$$PL = 35 \text{ Log}_{10}d \quad (5.1)$$

where antenna gain of both the receiver and the transmitter are assumed to be unity, propagation exponent α was chosen to be 3.5 throughout the whole area so as to not assume an optimistic environment [Joh95], and d is the T-R separation distance.

Large scale fading or shadowing is considered in this model as another effect of the environment on the transmitted signal. Log normal distribution with a chosen standard deviation value of 6 dB is used in the simulation. Two consecutive samples of log normal shadowing have some kind of correlation because of mobile's motion, shadowing was found to be correlated over a distance of 110 meter [Car94]. In this model, however, no correlation is considered which corresponds to the worst case situation [Che96]. Small scale fading (Rayleigh) is also included in the simulation.

Although it is assumed that directional antennas with an ideal 120° radiation pattern are used in all base stations, signals will not be attenuating according to that ideal pattern. The reason for this is that the urban environment causes transmitted signals to reflect in the opposite direction from high buildings and other obstacles. This phenomena is called *Back Scattering* and can be avoided if Omni directional antennas are used (which is not the Case considered here). This back scattering increases interference levels and consequently degrades system capacity.

The ratio of signal strength in the main direction to that in the opposite direction is called *Front to Back Ratio*. Higher values of the front to back ratio cause negligible

degradation in system performance. Approximately 5.0 dB reduction in CIR was found for low ratio values [Joh95]. This is the main reason for assuming a 20 dBm front-to-back ratio is in the model under consideration. The link budget calculations in [Joh95] found that 34 dBm transmitted power is sufficient to have a received power median at the cell edge of -81 dBm. However, link budget calculations were based on a voice user with a minimum required CIR of 9 dB, such CIR value corresponds to high bit error rate, which is below the desired quality for data user. Therefore, the impact of different transmitted power values for data user is studied while keeping the voice user transmitted power at 34 dBm.

5.4 Mobiles Location, Direction, and Speed

Mobiles are uniformly distributed over the area of all 31 cells. Each generated mobile (i.e. call) is assigned an initial (x , y) location coordinates (assuming at ground level) where x can take any value between 0.0 and 18.0 km, and y takes any value between 0.0 and 13.856 km with equal probability. Mobiles are assumed to move in a random direction that is uniformly distributed between 0 and 360°. Direction of mobile movement is maintained throughout the call duration.

Since two types of mobile are considered, voice mobile and data mobile, each should have a different speed, where data mobile users are mostly stationary or moving with low speed. In the most part of this thesis, data users are assumed to be moving with a random speed that is uniformly distributed between 0.0 and 3.0 m/s (~11 km/hr). However , to investigate the effect of non mobility on the system capacity, stationary data users have been considered. On the other hand, voice mobile users are assumed to have uniformly distributed speed between 0.0 and 14 m/s (~ 50 km/hr). The speeds of both mobile types do not change during the call.

5.5 Traffic Analysis

Traffic characteristics of the data mobile user are different than those of the voice mobile user because data users tend to make less number of calls λ with a longer duration $1/\mu$ at a higher transmission rate n (i.e. more time slots). The voice user is assumed to make 3.6 calls per busy hour ($\lambda_v = 3.6$). Each call has an exponentially distributed duration with an average $1/\mu_v$ of 200 seconds. Since only one time slot per TDMA frame is assigned to each voice user $n_v = 1$, the traffic generated by a voice mobile user was found as follows

$$E_v = \frac{\lambda_v n_v}{\mu_v} = \frac{(3.6)(1)(200)}{3600} = 0.2 \text{ Erlang} \quad (5.2)$$

The data user is assumed to make only one call per hour ($\lambda_v = 1$). A call is assigned a random duration that is exponentially distributed with an average $1/\mu_v$ of 300 seconds. As mentioned in chapter 4, the multi slot approach in which more than one time slot is assigned (up to eight) is used to provide the data user with higher transmission rates upon request. The probability that a user asks for n (where $n = 1:8$) time slots is shown in table 5.2. These probabilities were chosen to model the situation where fewer users need to transmit with a higher transmission rate. However, the case where the data user can ask for any number of slots with equal probability is also studied where the average number of required slots per user is 4.5.

Number of slots k	1	2	3	4	5	6	7	8
Probability that $n = k$	0.28	0.18	0.14	0.12	0.10	0.08	0.06	0.04

Table 5.2 Probability distribution of required number of time slots.

Using figures from table 5.2, the average number of time slots required by a data user can be calculated as shown below

$$\bar{n}_d = \sum_{i=1}^8 n_i p_i \quad (5.3)$$

$$= (1)(.28) + (2)(.18) + (3)(.14) + (4)(.12) + (5)(.10) + (6)(.08) + (7)(.06) + (8)(.04)$$

$$= 3.26 \text{ slots.}$$

therefore , the average traffic generated by a data user is

$$E_d = \frac{\lambda_d \bar{n}_d}{\mu_d} = \frac{(1)(3.26)(300)}{3600} = 0.2717 \text{ Erlang} \quad (5.4)$$

The first available channels are assigned to the user, where the data user is allocated the maximum available slots even if it is less than the required number. This number cannot be increased during the call. However, this number can be decreased when handoff is necessary if it is not available in the new cell or sector. Although a higher transmission rate means shorter transmission time, the call duration is set randomly for each user and never changes even if the transmission rate (i.e. number of slots) has been decreased.

5.6 Frequency Hopping

GSM specifications describe two frequency hopping schemes: Cyclic and Random frequency hopping. These two schemes are described in section 2.2.3. System performance for both schemes have been observed through simulation. Hopping is allowed only over four carrier frequencies out of the five available because no hopping is allowed on the BCCH frequency as shown in figure 5.3. It was found in [Joh95] that a minimum of 3 to 4 hopping frequencies are required to achieve the desired frequency diversity.

Fig 5.2 Hopping over 4 frequencies.

5.7 Handoff and Power Control

GSM specifications do not require a specific handoff scheme, however, there are two schemes which are widely used in most of the studies concerned with GSM. *Distance based* handoff connects the mobile to the nearest base station. The second scheme is the *quality-level based* handoff which connects the mobile to the best base station, i.e. the base station with the strongest signal and least interference level (described in section 3.4). These two schemes are applied for both handing over a call that is already in progress, and a call that has just been originated.

The distance based scheme gives a pessimistic result because it tends to underestimate the CIR by as much as 5 dB compared to the best base station scheme. This might be due to the fact that the nearest cell does not guarantee the strongest signal level especially when one or more of the neighboring cells does not suffer heavy shadowing while the nearest one does [Jov97].

For the sake of simplicity, the distance based handoff scheme has been adopted in this thesis. Handoff can take place from one cell to another, or within the same cell when the mobile crosses the virtual borders between sectors. Latter handoff is initiated based on the angle measurement that determines whether the mobile has changed sector or not.

No power control mechanism has been implemented in this study. Two reasons were behind this assumption, first, GSM is a TDMA system which is a bandwidth limited rather than interference limited system like the case of CDMA systems [Jov97][Rap96]. Second, power control is required more in the uplink than downlink, where the main objective in uplink is to prolong the MS battery life. The assumption of no power control does not violate the GSM specifications which leaves it open as an option for the network operators. Nevertheless, it is highly recommended to consider that case in future studies.

5.8 Summary of Model Parameters

The parameters of the GSM network model needed to be simulated which are described in previous sections are summarized in table 5.3

Path Loss model	$PL = 35 \text{ Log } d$
Standard deviation of log-normal shadowing	$\sigma = 6 \text{ dB}$
Rayleigh Fading	Included
Cell Radius	2 km
BS Tx power to voice users	34 dBm
BS Tx power to data users	Variable (34, 40, or 44 dBm)
Carrier Frequencies used	5 / sector
Time slots used	8 / Carrier frequency
Antennas	120° sectorized
Voice user Call mean duration	200 seconds
Data user Call mean duration	300 seconds
Voice user speed	Uniform[0 , 14] m/s
Data user speed	Variable (stationary, or Uniform[0 , 3] m/s)
Mobility Direction	Uniform[0 , 2π] (constant over call duration)
Frequency hopping scheme	Variable (Random or Cyclic)
Power control scheme	No power control is assumed
Handoff scheme	Based on distance

Table 5.3 Summary of GSM model parameters.

Chapter 6

SIMULATIONS TO BE PERFORMED

In the previous chapter, the GSM network model to be simulated was described. In this chapter, the simulation tool as well as a detailed flow diagram are presented. The final part of the chapter focuses on the different simulation runs associated with the different parameters settings used to study the capacity and performance of the GSM network model under consideration.

6.1 Description of Simulation Tool

MODSIM III has been used to write the computer program designed to simulate the GSM model. MODSIM III is a general-purpose, modular, block-structured high level programming language which provides direct support for object-oriented programming and discrete event simulation [CACI97]. The decision to use MODSIM III was mainly because of two features, first of which is the existence of a library module that contains objects and support procedures. The second feature is that all objects can perform actions that elapse simulation time.

Every entity in the model (e.g. mobiles, cells, sectors, carrier frequencies, and channels) was presented by an object and each contains a number of fields (variables) used to describe the status of its associated entity. For example, a mobile user object is generated whenever a call request is initiated from/to a mobile station. This mobile user object carries all the information of the mobile station such as geographical location, cell, sector, type, distance to other cells, measured CIR and BER value etc.

6.2 Flow Diagram

Figure 6.1 shows the flow diagram of the computer program used to simulate the GSM network model described in chapter 5. Small loops are executed over all active mobiles one by one before jumping to the next loop. The active mobile is the one with call in progress, while the inactive mobile is the mobile that is generated and initialized but has not yet started the call. Each frame loop takes 4.615 ms which equals the TDMA frame duration as per GSM specifications. A large loop of 104 frames has a duration of 0.48 second and corresponds to one SACCH multi frame.

Although the “Generate Mobiles” block appears on the top of the flow diagram, it works in parallel with all of the other blocks, i.e. the function “generate mobiles” works independently and does not wait for other functions to complete execution before starting. The work of the generate mobiles block can be summarized in two repeatedly executed steps: first generate a mobile, then wait an inter-arrival time before generating another mobile and so on, where inter-arrival time can be determined by the traffic and loading calculations.

Generated mobiles then need to be initialized with many parameters such as random location, type, speed, direction, call duration, transmission rate for data users, and the parameters required for frequency hopping. The distances from the mobile to all cells are measured and the nearest cell is chosen as a home cell which in turn finds the proper sector and the available channels. If no channels are available the call will be blocked and cleared, and the mobile will be left inactive, otherwise the mobile is considered active and call will start. This implies that calls (mobiles) generated within an SACCH multi frame need to wait until the beginning of the next multi frame before starting , which means that the average delay is only 0.24 seconds.

Fig 6.1 Flow diagram of simulation program.

The path loss as well as the log-normal shadowing are calculated for each active mobile as described in section 5.3. Since the library of random variables in MODSIM III does not include Rayleigh distribution, the technique described in chapter 9 of [Ban96] is used to simulate the effect of fast Rayleigh fading by generating a new value from the Rayleigh distribution in each TDMA frame. Using the above mentioned propagation parameters, the received signal level and the received interference level can be calculated. The interference received from all co-channel cells is included (not only the first tier) which leads to the worst case of the CIR level. For data users allocated more than one time slot, the same calculations are carried out in each time slot since interference level varies from one slot to another depending on whether the corresponding slots on the same frequency in other co-channel cells are being used or not.

While the CIR level can be used as a quality measure for voice transmission, the *Bit Error Rate* BER is the quality measure in the case of data transmission. Therefore, the variable of interest in this simulation is the BER of data users. All 104 values of BER -each calculated in one TDMA frame - are averaged to have one value per SACCH multi frame for every data user sent and saved in an output file. A lookup table was included in the simulation program to convert CIR values to BER values based on the curves given in figures 3.5 and 3.6.

The mobiles are assumed to be stationary within the 4.615 ms small loops where displacement is negligible, however, the mobiles locations, (x,y)'s, are updated after each 0.48 second main loop according to the assigned velocities. The distances to all base stations are also updated based on new locations. Consequently, the decision to perform a call handoff can be made if the mobile gets closer to a new cell (or sector) than its old one. A mobile will be made inactive if it goes out of the designated coverage area or if its call duration becomes expired.

6.3 Simulation Time

In order to obtain reliable simulation results presented by CDF curves, a large number of data should be collected by the simulation tool. In this thesis, simulating the network for 20 minutes of real time was found to be sufficient to produce data that is large enough to have results with a high level of accuracy. However, since the system is assumed to be initially empty, a warm up or initialization period is required to fill up the system and remove the bias caused by that assumption. Using the moving average method described in [Ban96], a 4 minutes warm up period is approximately sufficient to reach the steady state. Therefore, each simulation is for a total of 24 minutes with no data to collect in the first 4 minutes. This way, the impact of the initial conditions is almost removed. Using Windows NT on PII machine, maximum simulation running time was about 6 hours.

6.4 Collecting Data and Displaying Results

As mentioned previously, the 104 BER values which correspond to the 104 TDMA frames per SACCH multi frame are averaged to have one BER value sent to the output file for each mobile every 0.48 second. In this case, the performance of the whole system rather than individual mobiles is of interest.

BER rather than CIR was chosen to be the measure of the system performance since it reflects the quality of transmission as experienced by the end user, whereas CIR is the quality measure experienced by the receiver. System performance is a two-fold term which contains two measures with a trade off. These two measures are the transmission quality mentioned above, and the capacity which is the maximum number of users served by the system while maintaining a required level of quality. The trade off between quality and capacity is due to the

fact that any increase in number of users causes an increase in the interference level which in turn reduces the system quality, or in other words, increases the BER.

Both CIR and BER are stochastic processes, therefore, the system performance is presented and displayed by the Cumulative Distribution Functions CDF of the BER values collected by the simulation program. The CDF gives the probability that BER goes below some certain threshold ber. This threshold can take any value such as 10^{-2} , 10^{-3} , or 10^{-4} depending on the required quality level

$$\text{CDF} = \text{Pr} (\text{BER} < \text{ber}) \quad (6.1)$$

6.5 Simulations to be performed

In previous chapters, many parameters that can affect the performance of a GSM network model were described. Simulation is needed in order to investigate any influence caused by the different settings of these parameters. In this section, all of the performed simulations aimed at coming up with solid conclusions regarding the above mentioned influences will be described.

1- The capacity of a GSM system is limited by two factors; hard blocking and soft blocking. Hard blocking occurs when no channels are available to accommodate a new call, while soft blocking occurs when the link quality cannot be preserved at some required level due to high interference from other cells. In most cases, 2% hard blocking and 90% CIR of 9 dB or better soft blocking is required. One way to achieve these goals is by adopting *Fractional Loading* through the use of some kind of call *Admission Control*. 10%, 20%, 30%, 40% and 70% traffic loads are investigated in this thesis.

2- The second factor to be tested is the network resources *Occupancy Ratio* of voice users to data users (OR). The occupancy ratio can be found by

$$OR = \frac{p \frac{\lambda_v n_v}{\mu_v}}{(1-p) \frac{\lambda_d n_d}{\mu_d}} \quad (6.2)$$

where p is the probability that the mobile is a voice user, λ_v , n_v , μ_v , λ_d , n_d , and μ_d are the traffic characteristics of the voice user and data user respectively as described in chapter 5. 0.5/0.5, 0.7/0.3, 0.8/0.2, and 0.9/0.1 ratios have been simulated.

3- Another factor that needs to be simulated is the base station power transmitted to the data users. 34 dBm, 40 dBm, and 44 dBm powers were simulated while the transmitted power to voice users was kept at 34 dBm in all cases.

4- The effect of different frequency reuse plans has also been studied through the focus on 1/3 and 3/9 reuse plans.

5- The random frequency hopping effect is compared with that of the cyclic hopping.

6- The mobility of data users can have a noticeable impact on system performance. As shown in previous chapters, mobiles moving with high speed will not benefit from frequency hopping since it will completely uncorrelate the fading from one burst to another. In this study, two cases are considered, stationary data users, and data users moving with a maximum speed of 3 m/s, i.e. (~ 10 km/hr).

7- The last factor is the probability distribution of the data users required transmission rate. Two distributions have been studied, the one shown in table 5.2, and a uniform distribution in which it is equally probable to have any of the available transmission rates (i.e. number of time slots).

A summary of all the different simulations that were performed is shown in table 6.1.

<i>Variable Name</i>	<i>Description or values</i>
Traffic loading factor	10%, 20%, 30%, 40%, or 70%
Occupancy ratio	0.5/0.5, 0.7/0.3, 0.8/0.2, or 0.9/0.1
Transmitted power to data users	34 dBm, 40 dBm, or 44 dBm
Frequency reuse plan	1/3 or 3/9
Frequency hopping scheme	Random hopping or cyclic hopping
Mobility of data users	Stationary or with 3 m/s speed
Probability distribution of data users transmission rates	Table 5.2 or Uniform distribution

Table 6.1 Simulations to be performed.

Chapter 7

RESULTS AND DISCUSSION

The purpose of this chapter is to present and discuss all of the results that were obtained by the computer simulation program outlined in chapter 6 for the GSM network model described in chapter 5. The results will be given at the end of the chapter .

7.1 Traffic load

The results for 10%, 20%, 30%, and 40% traffic loads at different voice/data occupancy ratios 0.7/0.3, 0.8/0.2, and 0.9/0.1 are shown in figures 7.1-7.3. As intuitively expected, the lower traffic load factors led to better system performance in terms of the BER's CDF at all occupancy ratios. This is because more interference was generated at higher traffic loads. A higher interference level reduces the CIR level which in turn increases BER (i.e. makes it worse). Another observation of the results is that a BER of 10^{-2} is one at which changing the traffic load from 40% to 10% causes the highest level of improvement with all tested ratios.

7.2 Voice/Data Occupancy Ratio

In figures 7.4 - 7.6, the results of the simulation at 0.7/0.3, 0.8/0.2, and 0.9/0.1 ratios are shown for 30%, 40%, and 70% traffic load factors respectively. In all three cases, different voice/data ratios almost have the same performance. The Partial Loading might be the reason behind the above results, since for any load factor including high loads like 70%, changing the voice-to-data ratio does not change the total load of the network, rather, changes the distribution

of network resources between voice and data while the total load is still the same. Since the performance of the system is measured by BER which is a function of the CIR affected by the traffic generated regardless of its type (voice or data), the performances of different ratios were almost the same.

7.3 Transmitted Power to Data Users

The effect of base station power transmitted to data users is shown in figure 7.7 for a 30% load factor and in figure 7.8 for a 40% load factor. For both factors, base station power transmitted to voice users was kept at 34 dBm, while 34, 40, and 44 dBm of transmitted power to data users was studied by simulation. At a 30% load, increasing the transmitted power to data users slightly improves the performance. For example, at BER of 10^{-3} , CDF rises from 0.15 to 0.18 then to 0.22 as the power increases from 34 dBm to 40 dBm, and then to 44 dBm. On the other hand, at a 40% load, performance was improved by increasing power from 34 dBm to 40 dBm since a higher level of transmitted power means a higher received signal level. However, a further increase of power to 44 dBm causes a degradation in performance due to the increase in the received interference level from other base stations. The latter case is observed clearly at a 40% load where more interference is generated comparing with the case of a 30% load.

7.4 Frequency Reuse Plan

Although a 1/3 frequency reuse plan was found in [Joh95] to give the best performance, the results found in previous sections show a pessimistic performance at the 1/3 reuse plan. Therefore, a 3/9 reuse plan was investigated by simulation and the results of both plans at a 30% load and 0.8/0.2 ratio are shown in figure 7.9. A significant improvement was caused by changing the frequency reuse plan to 3/9. For example, a jump from 0.22 to 0.53 in the CDF can

be seen at a BER of 10^{-3} . The reason behind this improvement is a reduction in the number of co-channel cells to one third of their number at a 1/3 plan and increasing the distances between those cells.

The influence of the voice/data ratio is studied once again at a 30% loading and a 3/9 reuse plan. This time a 0.5/0.5 ratio is included beside 0.7/0.3, 0.8/0.2, and 0.9/0.1. The simulation produced almost similar performances for all four ratios as shown in figure 7.10 which confirms the results found in section 7.2 of this thesis.

7.5 Mobility of Data Users

Figure 7.11 shows simulation results which compares the network performance when data users are stationary with that when data users are moving with a uniformly distributed speed between zero and 3 m/s. These two simulations were run for a 30% loading and a 3/9 reuse plan and gave the same performance regardless of the data users speed. This result is due to the assumption of uncorrelated shadowing samples, which almost eliminates the effect of speed (or motion) on performance because high speed users benefit from the uncorrelation caused by their speed in reality. This should improve the performance of the moving data users over that of the stationary users. However, since no correlation was assumed, both scenarios ended up having the same result.

7.6 Probability Distribution of Data Users Transmission Rate

The impact of two data user Tx rate probability distributions on system performance are shown in figure 7.12. The case of a uniformly distributed Tx rate (or time slots from 1 to 8) has a performance that is better than that of the distribution in table 5.2 by 0.04 on the BER CDF curve. The reason for this is that with uniform distribution, the average number of used time slots

per data user is 4.5, while the average is only 3.26 in the other case. Consequently, with more used time slots, it is probable to have more interferers which increases the received interference level.

7.7 Frequency Hopping Scheme

The objective of this section is to compare the system performance of each of the two different frequency hopping schemes: random hopping, and Cyclic hopping. The results given in figure 7.13 show that random hopping has a better performance than cyclic hopping. For example, at 10^{-3} , random hopping is 0.10 higher on the BER CDF curve. This result confirms those obtained in [Joh95] for a system with voice service only. The above results are due to the interference diversity utilized by random hopping which leads to the averaging of the received interfering signals. No interference diversity can be obtained with cyclic hopping in which the hopping sequence on used carrier frequencies is identical for all interferers.

7.8 General Remarks

In all of the cases discussed above, hard blocking was found to be much below the required 2% level and negligible in most of the cases considered due to partial loading. However, the network performance was found to be unsatisfactory for data applications that require a bit error rate of 10^{-4} or less. The best performance observed indicates that only 40% of the users can be provided with a bit error rate of less than 10^{-4} . Many factors may have contributed to this poor performance:

- 1) The interference received from the co-channel cells in all tiers was included in the calculations of CIR, while only one or two tiers have been considered in most of previous studies.
- 2) The BER of data users that reside in the seven central cells were measured and displayed in CDF curves. Those users are exposed to the highest level of interference

from all other cells that are almost at a short distance, compared to those users in boundary cells which are sufficiently distant relative to the majority of their interferers.

- 3) No coding or interleaving was adopted in this simulation. In practice, the bit error rate can be significantly improved through coding and interleaving which work to combat the disturbances caused by the radio channel such as noise, fading, and interference and provide a robust transmission.
- 4) Power control can further enhance the network performance by ensuring that each mobile or base station transmits the minimum power necessary to maintain the required quality of transmission. This way, interference level can be reduced, and as a consequence, the BER will improve.

7.9 Analysis of Statistical Confidence of Results

Every simulation run collects a different number of BER data which is presented by the CDF curve. Therefore, the results of different simulations have different statistical confidence margins. Using the approach described in appendix E of [Joh95], the relative error ΔE , with respect to the number of errors can be found by the following equation:

$$(\Delta E)^2 = \frac{1-P_e}{P_e N} \quad (7.1)$$

Where N is the number of collected data, and P_e is the probability of error which is usually chosen to be 10% since a coverage of 90% is often used as a measure of the required quality level. From equation 7.1, simulations that use more data sets to plot the CDF curve have lower relative error and hence higher confidence margin. Knowing that 18000 data was the smallest number of collected data in all the simulations performed in this thesis, the highest statistical relative error in the presented results is 2.2%.

Fig. 7.1 Simulation for different loads @ 0.7/0.3 ratio and 1/3 reuse plan.

Fig 7.2 Simulation for different loads @ 0.8/0.2 ratio and 1/3 reuse plan.

Fig 7.3 Simulation for different loads @ 0.9/0.1 ratio and 1/3 reuse plan.

Fig 7.4 Simulation for different ratios @ 30% load and 1/3 reuse plan.

Fig 7.5 Simulation for different ratios @ 40% load and 1/3 reuse plan.

Fig 7.6 Simulation for different ratios @ 70% load and 1/3 reuse plan.

Fig 7.7 Simulation for different Tx power to data users @ 30% load, 0.8/0.2 ratio, and 1/3 reuse plan.

Fig 7.8 simulation for different Tx power to data users @ 40% load, 0.8/0.2 ratio, and 1/3 reuse plan.

Fig 7.9 Simulation for different frequency reuse plans @ 30% load and 0.8/0.2 ratio.

Fig 7.10 Simulation for different ratios @ 30% load and 3/9 reuse plan.

Fig 7.11 Simulation for moving and stationary data users @ 30% load, 0.8/0.2 ratio, and 3/9 reuse plan.

Fig 7.12 Simulation for different probability distributions of data users Tx rate @ 30% load, 0.8/0.2 ratio, and 3/9 reuse plan.

Fig 7.13 Simulation for different frequency hopping schemes @ 30% load, 0.8/0.2 ratio, and 3/9 reuse plan.

Chapter 8

CONCLUSION

This chapter summarizes the work done in this thesis and presents some conclusions based on the results described earlier in chapter 7. In addition, this chapter also presents some recommendations for future research work.

8.1 Thesis Summary

As stated in chapter 1, the objectives of this thesis are:

- 1) Studying the effect of voice/data users traffic ratio on the quality of transmission.
- 2) Observing the influence of network operational parameters such as load factor, frequency reuse plan, and frequency hopping on network performance.
- 3) Investigating the impact on the network performance of changing the data users radio transmission parameters, such as transmitted power from the BS, mobility, and the transmission rate.

During the course of this study, a computer simulation program was developed to determine the performance of a GSM network with mixed voice/data traffic based on the network model described in chapter 5. Using this simulation tool, an extensive study on the effect which the various parameters have on the performance of the network was conducted and presented in chapter 6.

8.2 Conclusions

The following conclusions can be made based on the results of this study:

1) Changing the network resources occupancy ratio of voice users to data users has little impact on the system performance presented by the CDF curve of the BER.

2) Partial loading where only a fraction of the network resources is allowed to be used at any moment through the utilization of some kind of admission control algorithm is needed to achieve both hard and soft blocking requirements. However, increasing the load causes performance degradation.

3) The 3/9 frequency reuse plan was found to have a superior performance compared to the 1/3 recommended by [Joh95].

4) Random frequency hopping improves the performance significantly when compared to cyclic frequency hopping.

5) Increasing the base station transmitted power to data users can enhance the performance. However, there is an upper limit to the transmitted power beyond which performance starts to degrade, particularly at higher load factors.

6) When no correlation is assumed in shadowing, the mobility of data users was found to have no effect on the performance.

7) A higher average transmission rate for data users implies the use of more time slots per TDMA frame and has a negative influence on the network performance noticed in the higher BER values.

8.3 Recommendations for Further Studies

Based on the analysis and the results obtained in this study, the following areas are recommended for further research:

1) Effect of power control on network performance as well as finding the proper power control algorithm and parameters are two important areas that need to be investigated.

2) A uniform frequency reuse plan for both voice users and data users was adopted throughout this thesis. However, since some data applications require very low BER values which correspond to low CIR, different reuse plans for data users and voice users may have a potentially significant effect on performance that is worth investigating. The practical considerations for adopting different frequency reuse plans within the same GSM network should also be investigated.

3) A hierarchical cell structure which includes cells with different sizes such as macro, micro, and pico cells may improve the network performance. The simulation model presented in this thesis needs to be modified for this purpose.

Appendix A

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