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Title:	Performance Analysis of a Multi Carrier-TDMA/SFH-Dual Signal Receiver System using BPSK modulation in a Macrocellular Environment
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Abstract:

The development of a third generation mobile communication network is in progress. A European project is working towards a European standard. The investigation of multiple access schemes is part of this project, as was development of the Multi Carrier TDMA/SFH Dual Signal Receiver protocol. A mobile communication system is designed based on this protocol, incorporating issues as multiple cell and integration of services. A proposal for an appropriate capacity allocation protocol for integration of speech-, data- and packet service is given. Simulation results are given for the performance in terms of capacity, spectral efficiency and link budget evaluation. The results are compared with results for the Hybrid Direct Sequence/Slow Frequency Hopping protocol which was also investigated by the Telecommunication and Traffic Control Systems Group of the Department of Electrical Engineering of Delft University of Technology.

Indexing terms:

Multi Carrier, Time Division Multiple Access, Slow Frequency Hopping, Dual Signal Receiver

Summary

The transmission medium in outdoor wireless communications requires the use of special communication protocols. In this report we investigate a combination of Time Division Multiple Access (TDMA) and Slow Frequency Hopping (SFH) using a Multi Carrier (MC) technique. A Dual Signal Receiver (DSR) is used to obtain a high capacity, compared with a conventional receiver. The DSR is unique because it can receive two narrowband BPSK modulated signals without the use of spreading codes like in CDMA.

The investigated system is proposed as a candidate for the Universal Mobile Telecommunications System (UMTS). The capacity allocation flexibility is a very important issue in UMTS, because this system requires the integration of speech-, data-, and packet services. The combination of TDMA, SFH and the MC technique provides a great flexibility in capacity allocation. Performance results in terms of spectral efficiency, capacity and allowed cell size are obtained by simulations. Results are given for single cell and multiple cell.

Performance results of the MC-TDMA/SFH-DSR show that the DSR doubles the uplink capacity, however, requiring a higher value of E_b/N_0 compared with a conventional receiver, to fulfil the same proposed BER.

The obtained results are compared with the results of a spread spectrum protocol. This protocol is Hybrid DS/SFH-CDMA. The performance results for this protocol were obtained, using the same system parameters as were used for the MC-TDMA/SFH-DSR protocol. The MC-TDMA/SFH-DSR protocol shows to give much better results for the uplink. For the downlink the results are a little better for the MC-TDMA/SFH-DSR protocol.

CONTENTS

List of symbols and abbreviations

symbols

α_i	amplitude gain of the i^{th} path
α^2_i	power gain at a certain delay τ_i
β	path loss law factor
γ	signal to noise ratio per bit
Δs_i	remaining of s_i
$\delta(t)$	dirac function
$\delta_{i,c}$	random timing difference between two data signals
η	capacity
ϕ_c	carrier phase
σ_τ	delay spread
σ^2	variance
τ_i	delay of path i
θ_i	phase of the i^{th} path
υ	spectral efficiency
$\psi_{i,c}$	signal to interference ratio
ξ	shadowing margin
ω_0	carrier frequency
A	peak amplitude of the dominant signal
A_i	amplitude of the large signal
A_c	amplitude of the small signal
B_D	total bandwidth occupancy of a signal set
B_{ss}	bandwidth of a spread spectrum signal
B_c	coherence bandwidth
B_D	bandwidth of the original data signal
B_T	transmission bandwidth

$B_{channel}$	channel bandwidth
C	cluster size
c(t)	time invariant impulse response
$\frac{C}{I}$	signal to interference ratio
d	reuse distance
$\hat{d}$	estimate of the data signal d
$d_{i,c}(t)$	asynchronous random data signals
$d_{i,j}$	large signal for the j^{th} antenna
$d_{c,j}$	small signal for the j^{th} antenna
$E[\bullet]$	expectation operator
$\frac{E_b}{N_0}$	signal to noise ratio per bit
$\frac{E_c}{N_0}$	signal to noise ratio per chip
G_{AR}	antenna gain of the receiver
G_{FS}	free space gain
G_{HO}	hand over gain
G_p	processing gain
G_{PC}	power control gain
I	average interfering power
$I_0[\bullet]$	modified Bessel function of the first kind and zero order
k	Boltzmann's constant
L_{FS}	path loss
NF	noise factor of the receiver
n(t)	Additive White Gaussian Noise signal
P	received signal power
$\overline{P}$	average power
P_{ei}	Bit Error Rate of the large signal
P_{ec}	Bit Error Rate of the small signal
P_{EIRP}	effective isotropic radiated power

P_{RB}	average interference power from a neighbour cell
$P_{RB,\ 1st\ tier}$	average interference power from the first tier
$Pr\{\bullet\}$	probability
$Q(z)$	Q-function
R	cell radius
R_b	bitrate
R_s	symbolrate
r	distance
r(t)	composite input signal
$s_l(t)$	large signal
$s_c(t)$	small signal
$\hat{s}_i$	estimate signal
T_c	chip duration
T_b	bit duration
T_{syst}	system noise temperature

abbreviations

AWGN	Additive White Gaussian Noise
BER	Bit Error Rate
BPSK	Binary Phase Shift Keying
BS	Base Station
CDMA	Code Division Multiple Access
DECT	Digital European Cordless Telecommunications
DS	Direct Sequence
DSI	Digital Speech Interpolation
DSR	Dual Signal Receiver
ETSI	European Telecommunications Standard Institute
FDD	Frequency Division Duplex
FDMA	Frequency Division Multiple Access
FFH	Fast Frequency Hopping

FH	Frequency Hopping
GMSK	Gaussian Minimum Shift Keying
GSM	Global System for Mobile communications
IC	Interference Cancellation
ISI	Inter Symbol Interference
LOS	Line Of Sight
MAP	Multiple Access Protocol
MC	Multi Carrier
MS	Mobile Station
OSI	Open Systems Interconnections
PN	Pseudo-random Noise
PSTN	Public Switched Telephone Network
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
rms	root mean square
SFH	Slow Frequency Hopping
SIR	Signal to Interference Ratio
TDD	Time Division Duplex
TDMA	Time Division Multiple Access
TH	Time Hopping
UMTS	Universal Mobile Telecommunications System

1. Introduction

In the last decades mobile communication has become a research subject of increasing importance. At this moment all around the world numerous mobile communication systems are active. The demand for radio communication services is growing fast. Because of the limited available bandwidth for radio communication services, the available bandwidth has to be used as efficiently as possible. The systems that are active all over the world do not all use the same standards with respect to modulation technique, coding, access protocols, frequency band and so on. Looking at the development of mobile communication systems we can consider roughly two periods in the cellular mobile communication history [1,p.24-28]. These are known as the analog cellular era (1979-1992) and the digital cellular era (1992 -now). Digital communication provides advantages in comparison with analog communication :

- large variety of services (voice, fax, video, multimedia)
- high degree of reliability
- high degree of flexibility
- high degree of privacy
- higher spectral efficiency

These aspects lead to the fact that digital mobile radio systems are more economical than analog mobile radio systems.

In the analog cellular era only voice communication was provided by the analog systems. These systems are called the first generation mobile radio communication systems. In the digital cellular era voice and data communication are provided by the digital systems using data rates of up to 10 kbit/s. The information that has to be send is translated in bits and a number of bits is assembled into a packet. These systems are known as the second generation mobile radio communication systems. In Europe, the second generation system in use is the Global System for Mobile communications (GSM) [1].

The next step in the development of mobile radio communication systems is called the Universal Mobile Telecommunications System (UMTS), which should provide a much wider range of mobile services to the users via a range of mobile terminals at a data rate of upto 2 Mbit/s. The availability of the system at any place at any time for any service depends on technical and economical issues. The system should provide several applications with different demands with respect to data rates. The UMTS project is a European project which will be standardized by the European Telecommunications Standard Institute (ETSI) and this system will be the third generation mobile radio communication service. To finally obtain this system operational in Europe a lot of work still has to be done when we look at the three phases for the standardization of UMTS [2]:

phase 1	objectives of UMTS
phase 2	requirements definitions on: radio interface, protocols and signalling, network aspects, satellite aspects and services and applications.
phase 3	detailed standards for UMTS.

At this moment the standardization with respect to the radio interface is in progress. One main aspect of the radio interface is the multiple access scheme. The multiple access scheme describes the way the radio spectrum is shared between several simultaneous communications, occurring between different mobile stations and a base station. In this graduation report a contribution will be given to development of an appropriate multiple access scheme for UMTS. The *Multi Carrier - Time Division Multiple Access/Slow Frequency Hopping - Dual Signal Receiver* (MC-TDMA/SFH-DSR) protocol will be investigated as a possibility for macrocellular environment in UMTS.

What is the MC-TDMA/SFH-DSR protocol? This protocol is based on Time Division Multiple Access (TDMA) and Slow Frequency Hopping (SFH). These are both methods to use the available bandwidth in a multiuser environment, where different users simultaneously operate in the same channel without destroying the information of other users. Both TDMA and SFH will be explained in this report (chapter 2). Multi Carrier (MC) stands for the possibility of allocating more than one carrier to one user simultaneously. The Dual Signal Receiver (DSR) [3] provides simultaneous reception of two narrowband co-channel signals and is used in the system that is investigated here to enhance system performance. The performance analysis that is made of the MC-TDMA/SFH system using the DSR is unique in the development of narrowband mobile communication systems. It will become clear that the receiver enhances the performance of the system if we compare it with a conventional receiver using the same modulation form. In mobile communication systems a distinction is made between the communication from the Mobile Station (MS) to the Base Station (BS), which is called the uplink connection, and the communication from BS to MS, called the downlink connection. The DSR enhances the uplink capacity by a factor two, because two co-channel users can be allowed simultaneously. The effect of the presence of two co-channel signals simultaneously is derived in chapter 5.

The exact principle of this receiver and the properties it provides are explained in chapter 4.

Upto now a lot of research has been done in the area of multiple access protocols. Considering numerous system configurations with respect to the cell size, (macro, micro, pico) [4] a lot of protocols and combinations of protocols (hybrid protocols) are investigated throughout the world with respect to performance considering a lot of system parameters.

The system that is investigated in this report should fulfil the requirements that are defined for UMTS in the macrocellular environment. These requirements are given as [5]:

• the system must be able to handle variable bitrates of different services in an efficient way.

• the second generation mobile digital communication system must be integrated in UMTS. There must be a compatibility between the two systems. This of course for economical reasons.

• high bitrates up to 2 Mbit/s should be possible.

Chapter 2 gives an overview of multiple access protocols. Advantages and disadvantages are given of these protocols. From this overview the TDMA/SFH structure, which is the basis for our protocol, shows to have good properties for UMTS. Of course this is not the only good protocol for UMTS. Therefore, other possibilities will also be mentioned. Some of them have already been investigated.

MC-TDMA/SFH-DSR is a narrowband system. This means that the bandwidth of the transmitted signal is about the same as the bandwidth of the information signal (this in contrary to spread spectrum systems where the bandwidth of the transmitted signal is much larger than the bandwidth of the information signal). In chapter 3, a characterisation is given of the narrowband radio channel. Propagation and interference aspects will be considered. These aspects are very important for further system design which will be treated in chapter 5.

In chapter 4 the priciples of the DSR that is used are reviewed from [3]. This receiver, which was developed in the Telecommunication and Traffic Control Systems group of Delft University of Technology, provides simultaneous reception of two narrowband Binary Phase Shift Keying (BPSK) modulated co-channel signals, e.g. two desired signals of different strength [3]. The receiver structure and the Bit Error Rate (BER) for both signals is given in this chapter.

The physical layer analysis of the proposed MC-TDMA/SFH-DSR is described in chapter 5. The physical layer is the lowest layer in the Open Systems Interconnection (OSI) communication model [6]. To transmit a packet of information, each bit is converted into an electrical signal by the physical layer. The signals are sent over the physical link (transmission of radio waves through the transmission channel: the macrocellular environment) and are received at the other end, where they are converted back into bits. Successive bits are reassembled into a packet by the receiver. The packet is then passed through to the data link layer of the receiver.

This chapter gives a complete overview of the performance of this protocol in single cell (no intercell co-channel interference) and multiple cell (with intercell co-channel interference). This analysis is done for two types of services:

- Speech service
- Data service

The packet service which is also a required service for UMTS will not be treated in this chapter.

For the MC-TDMA/SFH-DSR protocol, the basis is roughly TDMA and SFH. To allocate the capacity available to the different users who operate simultaneously in the same channel, some rules have to be created to avoid mutual destruction of information by the users that are active. The datalink layer protocol contains rules which the users have to follow up. The access protocol to the channel is not the same for all the services. For efficient capacity allocation to the different types of users, a method is developed qualitatively in chapter 6. In this chapter a start is made for the capacity allocation protocol.

As mentioned MC-TDMA/SFH-DSR is a narrowband system. In the overview of multiple access protocols, spread spectrum techniques, like Direct Sequence Code Division Multiple Access (DS-CDMA), are mentioned as a candidate for UMTS. Results for a system that uses DS-CDMA, are given in chapter 7. These results are for DS/SFH-CDMA which is also investigated in the Telecommunication and Traffic Control Group for macrocellular environment for the same system parameters as were used to investigate MC-TDMA/SFH-DSR. These results are compared with the results for MC-TDMA/SFH-DSR.

Finally, in chapter 8, conclusions and recommendations are given.

2. Multiple Access Techniques

In cabled communication networks like the Public Switched Telephone Network (PSTN) users communicate via a dedicated cabled channel. This means that there is no interference between users in the system which makes it easy to design this kind of networks. In a mobile communication system, which is investigated in this report, the communication between mobile users and the BS is wireless. To realize a network for wireless communication using radio transmission, it is evident that the available bandwidth has to be shared by the users. The main problem in sharing this bandwidth is the mutual interference between users. The role of a Multiple Access Protocol (MAP) is to coordinate the access to the common radio channel in such a way that the channel is used as efficiently as possible. A good Multiple Access Protocol should therefore possess the following basic properties:

- Access of the users to the channel must be controlled.
- Stability must be guaranteed as much as possible.
- In changing conditions the protocol should be robust. This means that even if the user does not operate in the right way, negative effect to the rest of the system should be minimized.
- Every user should be able to use the capacity is requires.

In this brief introduction on Multiple Access Protocols we want to look especially at the advantages and disadvantages of Time Division Multiple Access Protocol (TDMA), the Frequency Division Multiple Access Protocol (FDMA) and the Code Division Multiple Access Protocols (CDMA). Before this is done, an overview of the position of these protocols in the existing Multiple Access Protocols will be given.

We can divide the protocols roughly into two groups [7]: the contentionless (scheduling) protocols and the contention (random access) protocols. We can count the mentioned protocols to the group of contentionless protocols. CDMA is a contentionless protocol where a number of users are allowed to transmit simultaneously without conflict. However if the number of simultaneously transmitting users rises above a maximum, contention will occur. An overview is given in figure 2.1 [7].

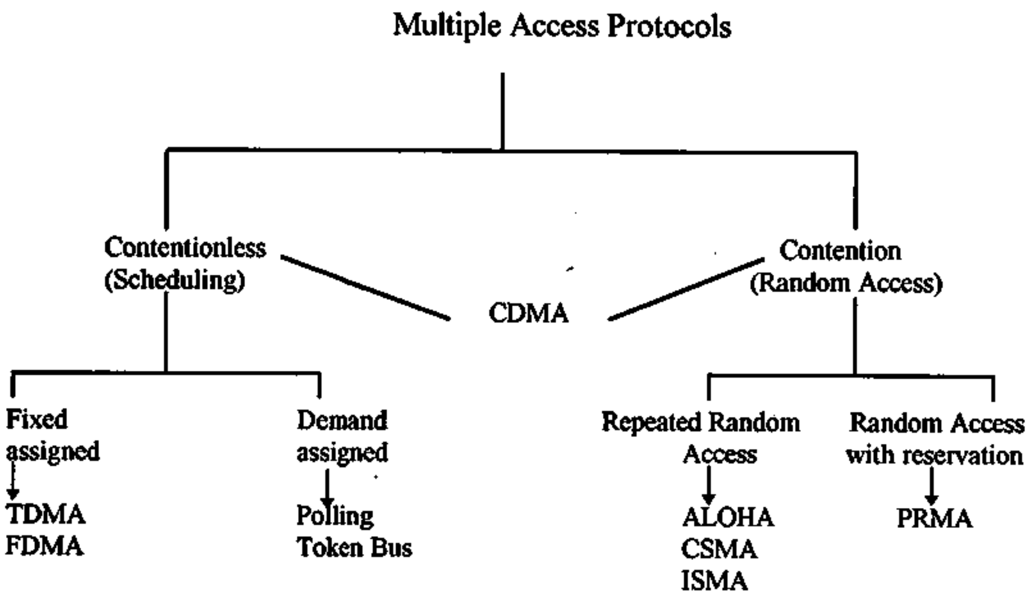

Figure 2.1 Overview of Multiple Access Protocols

2.1 A brief overview of TDMA and FDMA.

2.1.1 Introduction

TDMA and FDMA belong to the group of fixed assigned contentionless Multiple Access Protocols, which means that the available channel capacity is divided among the users in such a way that each user is allocated a fixed part of the capacity independent of its activity.

2.1.2 Time Division Multiple Access (TDMA).

In the TDMA protocol, the channel capacity is divided in time. The time axis is divided into frames. These frames all have the same duration. Each frame is divided into time slots and all the frames have the same number of time slots. All these time slots have the same duration. Each user is allocated a specific slot in a frame and the slot number stays the same for a user over the sequence of frames that a user is active (figure 2.2) [8].

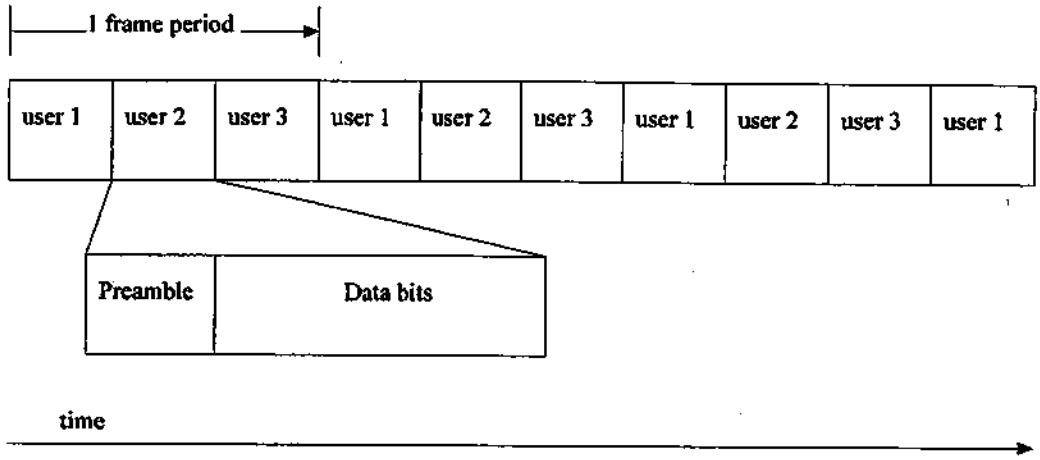

Figure 2.2 TDMA frame format

From figure 2.2 we see that synchronization is very important to avoid overlap in time with other users. Each user has during his timeslot the whole channel bandwidth to its disposal. With this way of dividing the channel capacity each user has the same amount of capacity if he uses it or not. This means that because of the fact that not every user needs the same amount of capacity, a lot of capacity is wasted. The following advantages and disadvantages can be mentioned.

Advantages of TDMA:

- GSM system is based on TDMA, so the use of TDMA provides compatibility with GSM.
- TDMA structure makes flexible capacity allocation possible (a user can be allocated more than one timeslot in a frame to reach higher bitrates).

Disadvantages of TDMA:

- capacity is wasted if a user doesn't transmit when a timeslot is allocated to that user.
- synchronisation is necessary.

2.1.3 Frequency Division Multiple Access (FDMA).

With FDMA, the bandwidth of the communication channel is divided in a number of frequency bands with guard bands between them to achieve frequency separation of adjacent bands [8]. Each user is allocated a particular frequency band for its own private use. So with FDMA a user can use part of the transmission channel all the time (figure 2.3).

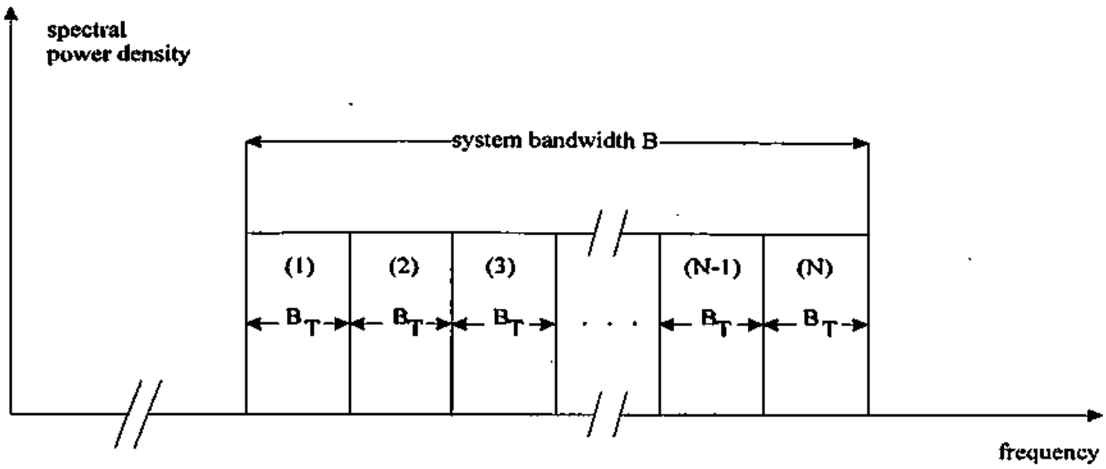

Figure 2.3 FDMA format with N users

FDMA has the same wasting properties as TDMA. If a user has nothing to transmit, its frequency band can not be used by other users. The following advantage and disadvantage can be mentioned:

Advantage of FDMA:

- no synchronisation needed

Disadvantage of FDMA:

- capacity is wasted if a user doesn't transmit when a frequency band is allocated to a user

2.2 *A brief overview of Spread spectrum techniques.*

2.2.1 Introduction.

Using the spread spectrum technique, the signal occupies a bandwidth larger than the minimum necessary to send the information; the band spread is accomplished by means of a code which is independent of the data, and a synchronized reception with the code at the receiver is used for despreading and recovery.

We can spread the spectrum for example to achieve anti-jamming and anti-interference capabilities. Some of the techniques are: "Direct Sequence (DS)" modulation in which a fast pseudorandomly generated sequence causes phase transitions in the carrier containing data, "Frequency Hopping (FH)", in which the carrier is caused to shift frequency in a pseudorandom way, and "Time Hopping (TH)", wherein bursts of signal are initiated at pseudorandom times. Combinations of these techniques are frequently used.

2.2.2 Some properties of the spread spectrum modulation techniques.

A very important aspect of spread spectrum is how the technique affords protection against interfering signals. The principle works as follows. A relatively low dimensional data signal is distributed in a high dimensional environment so that a jammer with a fixed amount of total power is obliged to spread that fixed power over all the coordinates thereby inducing just a little interference in each coordinate [9].

The signal to be sent can be completely specified by a linear combination of no more than D<M orthonormal basis functions. The signal set defined in this way is D-dimensional if the minimum number of orthonormal basis functions required to define all the signals is D. D can be shown [9] to be approximately $2B_DT$ where B_D is the total bandwidth occupancy of the signal set.

If code sequences with good correlation properties are used, after the correlation operation the unwanted signals appear as noise-like signals with a very low power density spectrum. In order to establish the spreading effect, the duration T_c of a code (chip) must be chosen much smaller than the duration T_b of a data symbol. Therefore the bandwidth B_{ss} of the transmitted spread spectrum signal is much larger than the bandwidth B_D of the original data signal (figure 2.4).

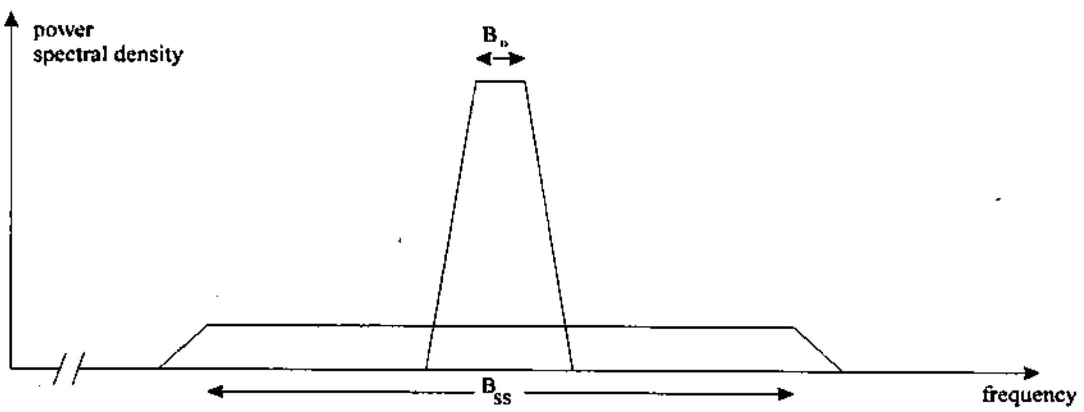

Figure 2.4 The principle of spread spectrum

The ratio between transmitted and original bandwidth is defined as the processing gain [9]:

$$G_p \triangleq \frac{B_{ss}}{B_D} \qquad (2.1)$$

We use an approximation:

$$G_p \triangleq \frac{B_{ss}}{B_D} \approx \frac{T_b}{T_c} \qquad (2.2)$$

2.2.3 Code Division Multiple Access (CDMA).

In the previous section a brief overview was given of spread spectrum techniques. CDMA is a Multiple Access Protocol that is based on the spread spectrum technique. CDMA protocols rely on coding to achieve their multiple access property. The code is used to transform a users signal in a wide band signal as was described in the previous section. The codes we use in this technique are generated by pseudorandom sequence generators. It is not in the scope of this report to work out the methods of generating these codes.

There are several CDMA methods. We can divide the CDMA protocols based on the modulation method used to obtain the wideband signal. This division leads to four protocol types:

1. Direct Sequence (DS) CDMA
2. Frequency Hopping (FH) CDMA
3. Time Hopping (TH) CDMA
4. Hybrid CDMA (combination of the types 1 t/m 3)

1. DS CDMA

In a DS CDMA system each user is given its own code, which is approximately orthogonal (i.e., has low crosscorrelation) with the codes of the other users.

The data baseband signal of each user is multiplied by its unique code. This results in the spread baseband signal. This principle is shown in figure 2.5.

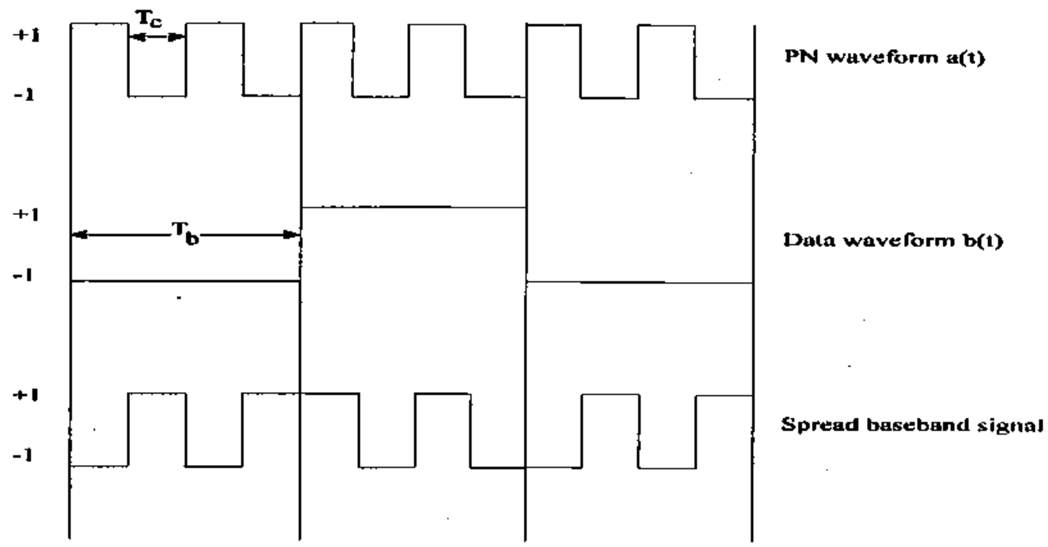

Figure 2.5 The principle of Direct Sequence CDMA

In figure 2.5 the earlier mentioned chip duration time T_c and the bit duration time T_b are shown. Since two code sequences with a relative delay of more than two chip times usually have a low correlation factor compared to the fully synchronized situation, DS-CDMA offers the possibility to distinguish between paths with a relative delay of more than two chip times [9]. This is called the inherent diversity of DS-CDMA, implying that it is possible to resolve a number of paths separately using only one receiver. This implies that resistance to multipath fading is obtained. Direct sequence systems are relatively simple, they do not need specific timing or frequency coordination between the transmitters due to the coding. The technique is being used in communication and military applications, as well as in navigation and positioning systems. DS-CDMA offers a lot of advantages but also has some disadvantages. Some of the advantages and disadvantages are listed below:

Advantages:

- resistance to multipath fading.
- low peak-to-average power ratio.
- low probability of intercepting.
- anti-jam resistance.
- security.

Disadvantages:

- near-far effect problem.
- fast code generator needed.
- long acquisition time.

In the case of the near-far effect problem, the N users are typically geographically separated so that a receiver trying to detect the k^{th} signal might be much closer physically to, say, the i^{th} transmitter rather than the k^{th} transmitter. Therefore, if each user transmits with equal power, the signal from the i^{th} transmitter will arrive at the receiver in question with a larger power than that of the k^{th} signal. This problem is often so severe that DS-CDMA can not be used. To combat the "near-far effect problem" several solutions are possible. One of them is strict power control.

2. FH-CDMA

In FH-CDMA, the wide band channel is divided in equal frequency bands. The carrier frequency is changed periodically during the transmission. This results in a periodically change of the frequency band occupied by the user. The code assigned to the user determines the pattern of frequency changes. Because of the fact that each user has his own hopping pattern, all users can transmit simultaneously. This aspect reduces the chance that a user close to the base station and a user far away from the base station transmit at the same moment which results in a certain resistance to the near-far effect problem which is present in DS-CDMA. At the receiver the demodulator follows the changes of the carrier frequency when demodulating the signal.

We can divide the frequency hopping CDMA systems into two categories: Fast Frequency Hopping (FFH), in which multiple frequency hops take place during the transmission of one data bit, and Slow Frequency Hopping in which multiple data bits are transmitted during one frequency hop. Figure 2.6 gives the principle of Frequency Hopping CDMA.

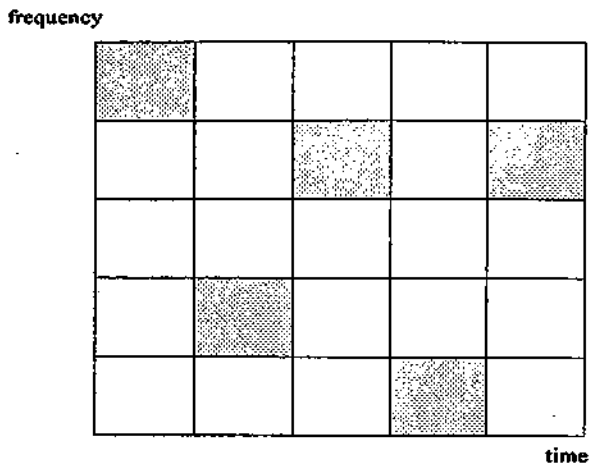

Figure 2.6 Frequency Hopping pattern with 5 hops

In mobile communication systems SFH is more easy to implement than FFH. Two types of hopping patterns can be distinguished [4]:

• Random hopping patterns; the carrier frequency hops randomly from one frequency to another under control of a Pseudo-random Noise (PN) process. Each user has its own hopping pattern, and all the users can transmit their information simultaneously. The number of hopping patterns in use can be greater than the number of frequency bands. This results in the possibility of a "hit". A "hit" occurs if two users send simultaneously in one frequency band.

• Deterministic (orthogonal) hopping patterns; the carrier hops from one frequency to another according to a hopping pattern. The hopping patterns that are available are all orthogonal which means that the number of available hopping patterns is equal to the number of frequency bands.

When orthogonal hopping patterns are used, there is theoretically no mutual interference (no "hits" occur). Frequency Hopping is used to avoid deep fades for a long time when a frequency selective channel is considered.

Frequency Hopping has the following advantages and disadvantages:

Advantages:

• GSM system uses SFH, this provides compatibility with GSM

• if there is a poor frequency band in the spectrum, every user uses it once in a while instead of one user all the time.

Disadvantage:

- complex frequency synthesizer.

3. TH CDMA

In the Time Hopping CDMA protocol a user signal is not transmitted continuously. It is transmitted in short intervals (bursts). The start of each burst is decided by the code assigned to the user. Because the signal is compressed in time, it will need a larger bandwidth during its transmission times.

4. Hybrid CDMA protocols

At this moment a lot of research is going on with respect to hybrid protocols. A hybrid protocol is a combination of two or more protocols. This can be any combination of contention or contentionless protocols. The goal is to combine the advantages of the separate protocols to suppress negative effects in certain circumstances.

An example of a hybrid protocol on which a lot of research has been done is the DS/SFH-CDMA protocol.

For example, for single pico cell environment it is shown [10] that the hybrid protocol has a better performance with respect to delay and throughput compared with the DS protocol and the SFH protocol seperate.

2.3 Performance of protocols

To measure the performance of a protocol we have to consider the circumstances in which the protocol operates, for example, the type and number of users, type of channel, size of the system are only a few parameters describing the environment.

In this report, the performance of the MC-TDMA/SFH-DSR is investigated. The basis of this protocol is TDMA and SFH. The combination of these protocols was chosen because this combination is currently in use in the GSM mobile communication system. The TDMA structure further provides an excellent capacity allocation flexibility and SFH avoids users to be in a deep fade for a long time. MC will show to be very useful to keep the channel bitrate low, even for high bitrate data services. It will be shown that this important to keep Inter Symbol Interference at a low level (chapter3).

In the Telecommunications and Traffic Control Systems Group of the Faculty of Electrical Engineering in Delft, a special narrowband receiver was developed [3]. This receiver is used to enhance the system capacity as will be shown in chapter 5. The principle of this receiver is reviewed in chapter 4.

Chapter 5 describes the performance of the system in a macrocellular environment. The performance of hybrid TDMA-DS/SFH-CDMA is also investigated by our Telecommunications and Traffic Control Systems Group, using the same system parameters as where used for the performance analysis for MC-TDMA/SFH-DSR. These results will also be presented in this report (chapter 7) and a comparison is made.

3. Characterization of the narrowband radio channel

3.1 Introduction

Mobile radio communication in a cellular structure takes place between a fixed BS and a number of MS. The geographical area in which these communications occur is called a cell, and we may consider that the cell boundary marks the maximum distance that a MS can move from the BS before the quality of communication becomes unacceptably poor. The cells in mobile communications vary substantially in size. An often used distinction in cell sizes is the following [4]:

1. Macrocell 2 < cell radius < 20 km
2. Microcell 0.2 < cell radius < 2 km
3. Picocell cell radius < 200m

The MC-TDMA/SFH-DSR system is investigated for a macrocellular environment (outdoor radio propagation).

In this section, an overview is given of the aspects of narrowband radio propagation that are considered in the investigation of the MC-TDMA/SFH-DSR system.

3.2 Narrowband multipath fading

In the investigation of the MC-TDMA/SFH-DSR system, a narrowband channel is considered. It is important to make a clear distinction between narrowband and wideband channels because of the different fading characteristics. These characteristics are described in section 3.3.

In mobile radio communication systems, a signal that is sent will propagate via a multipath channel which consists of a number of paths. This occurs because the signal is sent in all directions and is reflected by the earth surface, buildings etc. The delay times τ_i of the n paths, are not the same for the n different paths and the signal suffers from attenuation α_i which is not the same in every path. The phases θ_i of the signal are assumed to be time variant due to the time variations in the structure of the medium and varying antenna positions. The values of θ_i are assumed to vary randomly between 0 and 2π in time. Throughout the investigation the Codit COST 207 macrocell channel model is used [11]. This model is given in table 3.1.

table 3.1 :Codit COST 207 Macrocell channel model.

i	Relative delay τ_i (ns)	Relative power α^2_i (dB)
1	100	-3.2
2	200	-5.0
3	500	-4.5
4	600	-3.6
5	850	-3.9
6	900	0.0
7	1050	-3.0
8	1350	-1.2
9	1450	-5.0
10	1500	-3.5

Throughout the investigation of the MC-TDMA/SFH-DSR protocol only narrowband channels are considered and all reflections are assumed to add non-coherently. The non-coherent addition corresponds to a phasor addition of a large number of reflected signals.

The equivalent low-pass channel that contains discrete multipath components is described by the time variant impulse response of the channel, derived from [12]:

$$c(t) = \sum_{i=1}^{n} \alpha_i \cdot \delta(t - \tau_i) \cdot e^{j\theta_i(t)} \qquad (3.1)$$

where α_i is the amplitude gain of the i[th] path, τ_i is the propagation delay of the i[th] path and θ_i(t) is the phase of the i[th] path. In the channel that is used to investigate the protocol, has ten paths so n = 10.

3.3 Root mean square delay spread and coherence bandwidth

Root mean square (rms) delay spread and coherence bandwidth are parameters that quantify the multipath channel and can be determined from a power delay profile. In table 3.1 the power delay profile that is used is given. The root mean square delay spread is defined as the square root of the second central moment of the power delay profile, and is given by [13,p.160]:

$$\sigma_\tau = \sqrt{E[\tau^2] - E^2[\tau]} \tag{3.2}$$

with

$$E[\tau] = \frac{\sum_n \tau_n \alpha_n^2}{\sum_n \alpha_n^2} \tag{3.3}$$

and

$$E[\tau^2] = \frac{\sum_n \tau_n^2 \alpha_n^2}{\sum_n \alpha_n^2} \tag{3.4}$$

where α_n^2 represents the relative power gain at a certain delay τ_n.

A measure for the coherence bandwidth (B_c) is now found from [14]

$$B_c = \frac{1}{2\pi\sigma_\tau} \tag{3.5}$$

The coherence bandwidth is defined as the frequency seperation beyond which two signals of different frequency are not correlated. Two signals with a frequency separation greater than the coherence bandwidth are affected differently by the channel. The ratio of the bandwidth of the signal to be transmitted, which is denoted by B_T and the coherence bandwidth is used to characterize the channel:

$B_T << B_C$	Narrowband communication; frequency non-selective channel
$B_T >> B_C$	Wideband communication; frequency selective channel

For a digital signal, Inter Symbol Interference (ISI) can occur. No serious ISI is likely to occur if the symbol rate $R_s << (2 \cdot \sigma_\tau)^{-1}$ [15].

It will become clear that one of the advantages of MC is the possibility to choose such a value for the channel bitrate that no serious ISI is likely to occur.

3.4 Path loss

In radio communications the received signal power $\overline{P}$ statistically decreases as a function of distance r according to a general attenuation law:

$$\overline{P} = r^{-\beta} \tag{3.6}$$

where β is the path loss law exponent that is two in the case of free space loss and lies between three and five for mobile systems. In the investigation on the MC-TDMA/SFH-DSR protocol, $\beta = 4$ is assumed.

3.5 Statistical description of fading

The Ricean and Rayleigh distribution are commonly used to describe the statistical time varying nature of the received envelope of a fading signal. When there is a stationary (non-fading) signal component present, such as a Line Of Sight (LOS) component, Rician distribution is used. When there is no LOS component, Rayleigh distribution is used to describe the envelope of the received fading signal. The Ricean distribution is given by [13, p.176]

$$p(r) = \begin{cases} \dfrac{r}{\sigma^2}\exp\left(-\dfrac{(r^2+A^2)}{2\sigma^2}\right) I_0\left(\dfrac{Ar}{\sigma^2}\right) & for\ A \geq 0, r \geq 0 \\ 0 & for\ \ r < 0 \end{cases} \tag{3.7}$$

The peak amplitude of the dominant signal is denoted by A, r is the value of the envelope of the received fading signal, σ^2 is the average power that is received over specular paths, and $I_0(\bullet)$ is the modified Bessel function of the first kind and zero order. The Ricean distribution is often described in terms of a parameter K which is defined as the ratio between the deterministic signal power and the scattered power. It is given by K = $A^2/2\sigma^2$ or, in terms of dB:

$$K(dB)=10\log\left(\frac{A^2}{2\sigma^2}\right) \tag{3.8}$$

The parameter K is known as the Ricean factor and specifies the Ricean distribution completely.

If there is no LOS signal present, A = 0, and the Ricean distribution given in formula 3.7 simplifies to a Rayleigh distribution [13, p.173].

$$p(r)=\begin{cases} \frac{r}{\sigma^2}\exp\left(-\frac{r^2}{2\sigma^2}\right) & \text{for } r \geq 0 \\ 0 & \text{for } r<0 \end{cases} \tag{3.9}$$

This is the analytical model for the fading. The performance of MC-TDMA/SFH-DSR is investigated using the measured model given in table 3.1. This measured power delay profile will cause a Rayleigh fading envelope of the received signal.

In the outdoor environment there is not always a line of sight path. The Rayleigh fading model is used to derive results, so these results are derived for the worst case situation because sometimes there is a line of sight path between BS and MS.

4. The Dual Signal Receiver

4.1 Introduction

In the MC-TDMA/SFH-DSR protocol, the Dual Signal Receiver (DSR) is used to enhance the performance. In chapter 5 the protocol is investigated and results in terms of capacity, spectral efficiency and coverage analysis will be given for the DSR and a conventional receiver. In this chapter the principle and configuration of the DSR is reviewed [3]. The DSR is investigated for two narrowband BPSK modulated co-channel signals.

When a conventional BPSK receiver is considered, a strong interfering signal s_i in the presence of a weak signal s_c will capture the receiver and the weak signal will be lost. In this chapter the concept of the DSR is described for BPSK modulated signals. In the DSR the large signal s_i is first demodulated in the presence of the small signal s_c and noise, by a coherent BPSK demodulator. This demodulation results in a recovered data signal $\hat{d}_i$ of the large signal with Bit Error Probability P_{ei}. The estimate signal $\hat{s}_i$ is reconstructed from the estimated data and is subtracted from the total input signal. The signal that remains consists of s_c, noise and the remainder of s_i due to bit errors and nonperfect carrier and bit timing estimation. The remaining signal s_c is then demodulated by the second demodulator.

In this way both the large signal and the small signal are received simultaneously. The DSR is used to allow two co-channel users simultaneously in the uplink connection and to suppress negative effects from a co-channel interferer in the downlink. In chapter 5, a qualitative investigation is made on the performance of the system.

4.2 Signal Model

Let the input signal r(t) of the receiver consist of two BPSK modulated signals of different strength and Additive White Gaussian Noise (AWGN) $n(t)$.

$$r(t) = A_i d_i(t) cos(\omega_0 t + \phi_i) + A_c d_c(t) cos(\omega_0 t + \phi_c(t)) + n(t) \qquad (4.1)$$

where A is the signal amplitude, $\omega_0 = 2\pi f_0$ is the carrier frequency and ϕ is the carrier phase. The subscripts i and c indicate for the large and the small signal. $d_{i,c}(t)$ are asynchronous random data signals.

$$d_{i,c}(t) = \sum_{k=0}^{\infty} a_{k_{i,c}} \prod (t - kT + \delta_{i,c} T) \tag{4.2}$$

Here, $\prod(t - kT)$ indicates a rectangular pulse with duration T and the value of $a_k \in \{-1,+1\}$ with equal probability of occurrence. $\delta_{i,c}$ accounts for a random timing difference between both data signals. The frequency difference between s_i and s_c is assumed to be very small when compared to the bitrate and is therefore accounted for in the time dependency of ϕ_c. In the following, s_i is assumed larger than s_c so $A_i > A_c$.

4.3 Receiver concept

The DSR consists of two demodulators. This is shown a block diagram of the DSR, given in figure 4.1. The first demodulator is a coherent BPSK modulator, because of its superior performance in an environment with co-channel interference. This demodulator is captured by the large signal s_i, and produces an estimate of the data signal $\hat{d}_i$ with BER P_{ei}. These errors are caused by the presence of the small signal s_c and noise.

The estimated data signal $\hat{d}_i \in \{-1,+1\}$, is used to remodulate the composite signal $r(t)$, and thus decorrelation of signal s_i is achieved.

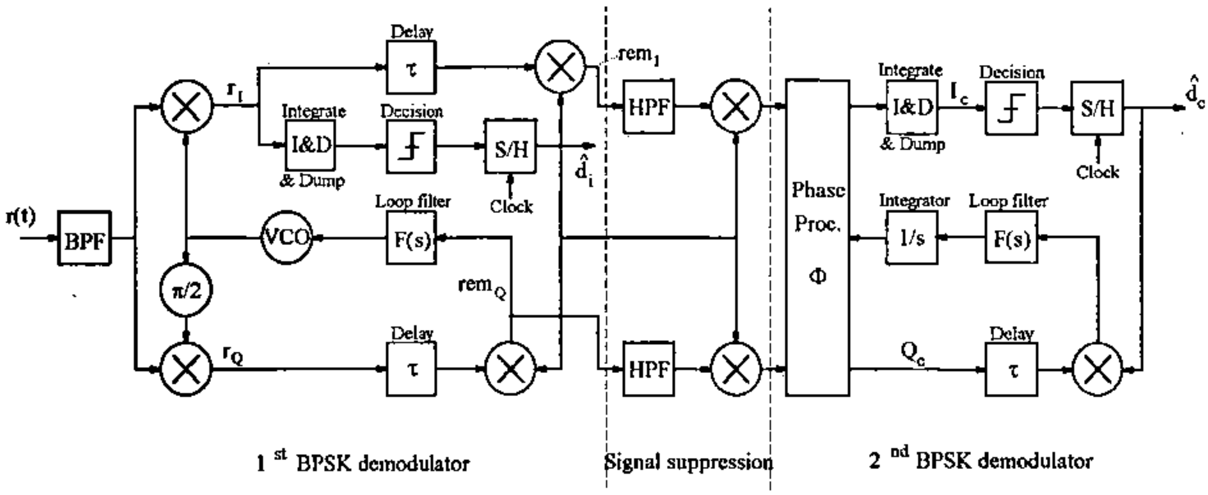

figure 4.1 block diagram of the Dual Signal Receiver

This remodulation results in concentration of the signal power of s_i into a carrier at frequency f_0. Due to data estimation errors the large signal is not completely concentrated at carrier frequency f_0. Now the signals s_i and s_c have become separated in the frequency domain, because nearly total power of the large signal s_i is concentrated in a small bandwidth carrier signal, whereas the bandwidth of the remodulated signal is still 1/T. By means of high pass filtering the carrier signal is extracted, and the power of the large signal is removed up to a large extend. Before demodulation of the remaining signal s_c is possible, a second remodulation with $\hat{d}_i$ is required to remove this modulation from $\hat{d}_i s_c$, since $\hat{d}_i^{\,2} = +1$.

The resulting signal after cancellation from which s_c is demodulated, consists of s_c, noise and Δs_i, where Δs_i indicates the remaining of s_i due to bit errors, timing errors and imperfect cancellation.

In figure 4.2, the time domain baseband signals in different stages of the signal processing of s_i are shown for the case that no timing errors occur. In practice bit errors and timing errors will occur and therefore the signal power is partly concentrated in the carrier component and partly in a residual signal spectrum.

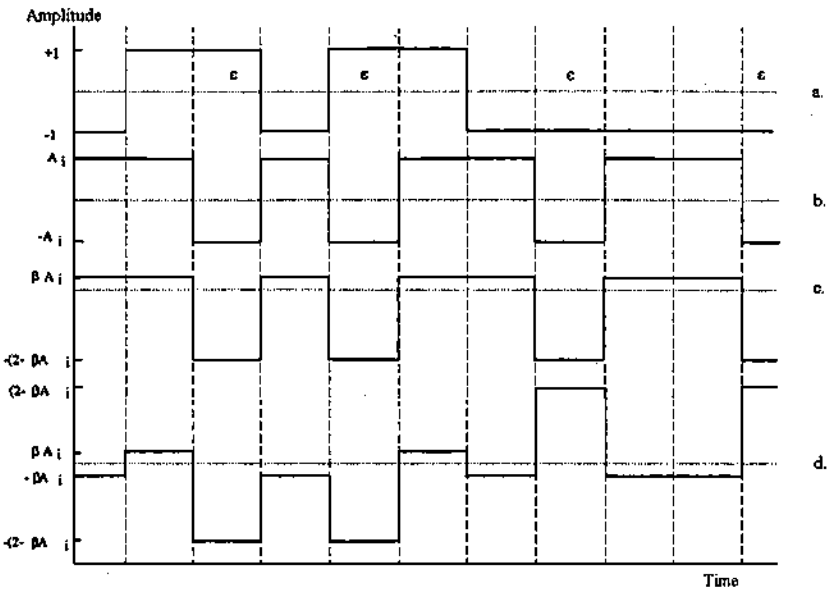

figure 4.2 Time domain sample of the baseband signal of s_i without timing error: (a) estimated data d_i, ε indicates an error; (b) signal after the 1st remodulation; (c) remodulated siganl after carrier suppression; (d) signal Δs_i after the 2nd remodulation.

The magnitude of $\alpha_j A_i$ of the symbols is randomly of two values $\alpha_j = \beta$ in case no error occurred in $\hat{d}_i$, and $\alpha_j = 2 - \beta$, when $\hat{d}_i$ was in error. β is a measure for suppression performance of s_i, $\beta = 1 - (1 - 2P_{ei})(1 - \eta)$.

4.4 BER expressions for the non fading AWGN channel

BER for the large signal

The BER P_{ei} of the large signal s_i, which is coherently demodulated by the first demodulator in the presence of the small signal s_c and noise, is derived using the conditional error probability:

$$P(\varepsilon|\phi',\Delta') = \frac{1}{4}\sum_{All\alpha_j,d_j} Q\left(\sqrt{2\gamma_i}\left[1+\frac{\alpha_0\Delta' d_0+\alpha_1(1-\Delta')d_1}{\sqrt{\psi_i}}\cos\phi'\right]\right) \qquad (4.3)$$

Here $\gamma_i = E_{bi}/N_0$ is the signal to noise ratio of s_i, with $E_{bi} = A_i^2T/2$. $\psi_i = A_i^2/A_c^2$ is the signal to interference ratio, $\alpha_0 = \alpha_1 = 1$, $\phi' = \phi_c - \phi_i$ is the carrier-phase difference between s_i and s_c, $\Delta' = |\Delta| \leq 0.5$ with ΔT is the timing difference between the data signals d_i and d_c.

Q(z) is the Q-function which is defined as

$$Q(z) \triangleq \frac{1}{\sqrt{2\pi}} \int_{z}^{\infty} e^{-\lambda^2/2} d\lambda \qquad (4.4)$$

The average BER P_{ei} is calculated by averaging (4.4) over ϕ' and Δ'.

BER for the small signal

The total conditional BER $P_{ec}(\varepsilon | \phi', \Delta')$ is expressed as [3]

$$P_{ec}(\varepsilon / \phi', \Delta') = P_0 P_{c0}(\varepsilon / \phi', \Delta') + P_1 P_{c1}(\varepsilon / \phi', \Delta') + P_2 P_{c2}(\varepsilon / \phi', \Delta') \qquad (4.5)$$

The BER P_{ec} is now calculated by averaging (4.5) over ϕ' and Δ'.

5. Performance analysis of the MC-TDMA/SFH-DSR system.

5.1 Introduction

In this chapter a mobile communication system with the multiple access scheme MC-TDMA/SFH-DSR is investigated. This communication system is a digital land mobile system and consists of a number of Base Stations. Two types of services are investigated, speech service with a bitrate of 12 kbit/s and data service with a bitrate of 128 kbit/s. In this chapter the performance of the mobile communication system is derived. Performance results are obtained in terms of capacity, spectral efficiency and coverage analysis. The DSR [3] which was reviewed in chapter 4, is used in the Mobile Station and in the Base Station. In the Base Station, the DSR is implemented with the goal of enhancing the system capacity by allowing two BPSK modulated co-channel users simultaneously. At this moment the demand for radio communication services urges the system designers to use the scarcely available bandwidth as efficiently as possible. Protocols, modulation types and coding schemes are developed to achieve optimum use of the available frequencies in a multiuser environment. The way of enhancing the spectral efficiency by using the DSR to allow two co-channel users simultaneously, is unique. The performance of the DSR in this way has not been investigated before. In the Mobile Station the DSR is also implemented but now to suppress negative effects of an unwanted co-channel interferer. By investigating the performance of these two applications of the DSR in terms of capacity, spectral efficiency and coverage analysis an overview of the possible uses of the DSR in a mobile communication system is made. Results are first obtained for a system with one Base Station and a number of Mobile Stations (single cell situation). After that, results are given for a system with a number of Base Stations and Mobile Stations (multiple cell situation). The performance of the DSR with two co-channel users is compared with the performance of a conventional BPSK receiver without co-channel interference for the given channel model in a multiple cell system. Using the DSR in the uplink, the uplink capacity is twice as large as the capacity for a conventional BPSK receiver. However a higher value of E_b/N_0 is required to obtain the same performance results as the conventional receiver. From the performance results that are derived in this chapter it is found that the implementation of the DSR in the uplink is a very good method to enhance the capacity of the system. This means that the performance of the second generation mobile communication

system GSM using the DSR is very interesting to investigate. The comparison between the DSR and a conventional receiver is also made for an AWGN channel without fading.
Finally conclusions are drawn from the results that are obtained and proposals are made for applications of the DSR and further aspects that are interesting to investigate.

5.2 TDMA/SFH-DSR

The radio interface of the hybrid TDMA/SFH-DSR system uses a combination of Time Division Multiple Access and Slow Frequency Hopping. The DSR can receive two co-channel signals simultaneously as was explained in chapter 4 [3]. The DSR is applied in the uplink, to investigate the capacity enhancement by allowing two mobile users in one frequency band.
In the basic concept of TDMA/SFH-DSR, the unit of transmission is a series of modulated bits which is called a burst. Bursts have a finite duration and occupy a finite part of the radio spectrum. They are sent in time and frequency windows called slots. Every timeslot is shared by a number of users. The maximum number of users per timeslot is equal to twice the number of hopping patterns if two co-channel users are allowed simultaneously in the uplink. In the downlink the number of users per timeslot is equal to the number of hopping patterns. The organisation of the time axis is cyclic. The positioning of the cycles in time is achieved through system synchronisation. A user is allocated one hopping pattern and can be allocated one or more timeslots per frame to send its information at various bitrates. For the data service, a bitrate of 128 kbit/s is considered.
Uplink and downlink share the total available bandwidth (this means that there is no necessary separate uplink and downlink band) which makes the system more flexible. The capacity allocation for the different services is given in chapter 6. The Dual Signal Receiver is implemented to:

- Allocate one hopping pattern to two users simultaneously for Uplink communication.
- Combat co-channel interference in Downlink communication.

In this analysis orthogonal hopping patterns are assumed. The system performance using random hopping patterns is not investigated. Overhead information that is required for capacity management, synchronization management and power control management is not taken into account.

5.2.1 Uplink

At the Base Station a Dual Signal Receiver (DSR) is used which applies co-channel Interference Cancellation (IC) [3]. The DSR can receive two narrowband Binary Phase Shift Keying (BPSK) signals of slightly different strength. The concept of the receiver was mentioned in the introduction of this chapter and given in chapter 4. This property enlarges the capacity of the system because for the uplink connection two users in a cell can share one hopping pattern. However, because of the fact that two co-channel users are admitted simultaneously, the required E_b/N_o to achieve a required Bit Error Rate (BER) is higher than in the case of a single user without co-channel interferer. Therefore a comparison is made of the actual gain in performance of the DSR compared with a conventional receiver for the application described here. In the design of the system it is very important to make a distinction between the different types of traffic because of the difference in allocating capacity. Antenna diversity using 3 antennas with majority voting and error correcting coding will be implemented to enhance the performance of the system in the uplink. This antenna diversity technique is explained in section 5.3.1.

5.2.2 Downlink

The downlink connection differs from the uplink connection in a few ways:

• No Random Access scheme has to be used for capacity allocation.

• The Dual Signal Receiver can not be used in the same way as in uplink connection.

In the uplink connection, the DSR is used to let two users share one hopping pattern. This is not possible for the downlink connection. However, the Dual Signal Receiver is useful in the Mobile Stations to suppress negative effects of co-channel interference.

• No antenna diversity is used in the downlink connection. Error correcting coding is used to enhance the performance.

5.3 Performance simulation results

5.3.1 Basic performance figures

In this section BER results are given for the noise limited case and the interference limited case. The properties of the DSR are investigated. As a first step the DSR can be used to combat co-channel interference. The desired signal can be recovered even in the case that the interfering signal is stronger than the desired signal. This is not possible with a conventional BPSK receiver where the desired signal will be lost in that situation. After that, the performance of the DSR in the uplink is treated. The BER results as a function of E_b/N_0 are given to determine the value of E_b/N_0 that is required for the uplink and the downlink. The Quality of Service (QoS) requirement that has to be fulfilled for speech- and for dataservice is that the average BER must be smaller than 1e-3. This QoS requirement will be used throughout performance analysis of this system and it was also used in the performance analysis of the TDMA-DS/SFH-CDMA system (results are presented in chapter 7 and compared with results obtained in this chapter). The values for the required E_b/N_0 are used in section 5.3.2 for the coverage analysis. In the coverage analysis, the maximum allowable cell size is derived for uplink and downlink for different values of the transmitting power.

Channel Model

The channel model that is used is a measured macro cell channel model. The model is the CODIT COST 207 channel [11]. The transmitted signal is reflected by the earth surface and obstacles and ten paths arrive at the receiver with different delays and relative power gains. The CODIT COST 207 model that is used was given in table 3.1.

In chapter 3, aspects of narrowband radio propagation were given. We use the formulas for delay spread and coherence bandwidth from this chapter to find the following values for the delay spread and coherence bandwidth:

$\sigma_\tau = 0{,}44\ \mu s$ and $B_c = 360$ kHz

To keep ISI low, $B_{channel}$ must be smaller than the coherence bandwidth. In the system design which is made in chapter 6 and in the simulations in this chapter, $B_{channel}$ is chosen much smaller than the calculated 360 kHz. In the simulation programs that are made in MATLAB, the BER performance is calculated with the path gains given in table 1 and randomly generated phases for the large and the small signal. From table 5.1 we find that the transmitted signal arrives at the receiver via 10 paths because of reflection that occurs. The ten reflected paths of one transmitted signal that arrive at the receiver add non-coherently. This non-coherent addition corresponds to a phasor addition of the ten reflected paths with their randomly generated phases. In the simulations for the uplink connection the two signals are transmitted in such a way that the average received SIR is 3 dB between the large and the small signal. This value of SIR was chosen. To determine the value of SIR which gives the best results, several aspects have to be taken into account:

• The value of E_c/N_0 that is going to be used in the uplink is about 6 dB for the large signal (this will become clear from the simulation results). For this reason, the SIR must be chosen not too large because otherwise the small signal will disappear below the noise level (on average). So SIR was chosen smaller than 6 dB.

• Simulations were done to calculate the outage probability (this is the probability that the BER is larger than a specified value) of the receiver as a function of SIR with an E_b/N_0 of 6 dB. It was found that this outage probability has its maximum for SIR = 0 dB. The outage probability gets better when SIR is larger.

A value between 0 dB and 6 dB, SIR = 3 dB, was chosen throughout the performance analysis that is described in this chapter. In section 5.6, the effect of the value for SIR will be considered and it will be investigated whether SIR = 3 dB is the optimum value.

Because of the multipath fading channel, the signal that was sent as the largest signal is not necessarily received as the largest signal. This effect is also taken into account in the computer simulations.

Slow power control is used in the system. Slow power control controls the shadowing. The fast fading is not controlled. Power control that controls the fast fading is called fast power control. Slow power control requires less information overhead and less complex controlling mechanisms. Because of the slow power control, the received power is on average equal to the desired value but because of the multipath fading, it is not exactly the desired value for the received power at every moment.

Noise limited case

The noise is assumed to be AWGN. The performance of the DSR is measured for the case that we allow only one desired user in a frequency band (for downlink connection). One very small interferer is present, with SIR = 30 dB. In figure 5.1 the average BER is given as a function of E_b/N_0 for speech and data services. It can be found from figure 5.1, that an E_b/N_0 of almost 25 dB is required to reach a BER of 10e-3. This is due to the multipath fading, in a non-fading channel the BER performance is much better. Results for a non-fading channel are given in section 5.6.

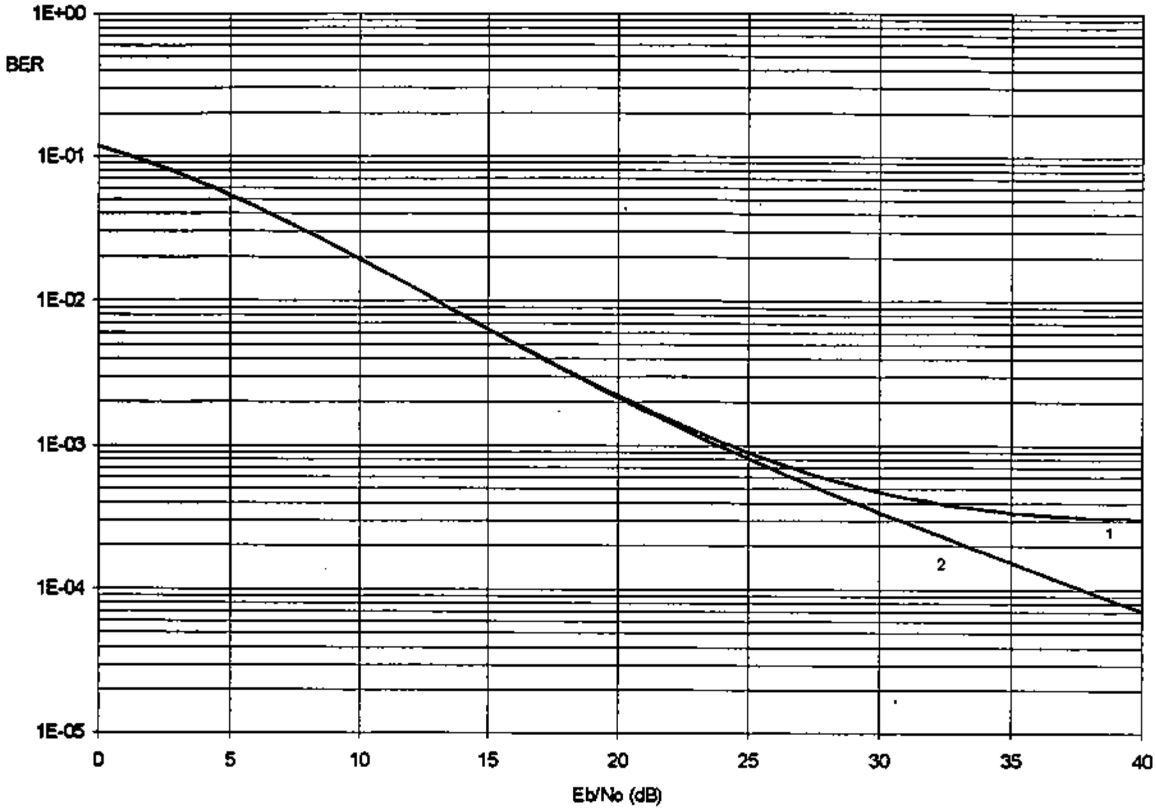

figure 5.1: Average BER as a function of E_b/N_0 for the COST 207 channel for SIR=30 dB, for downlink, with:
1. A conventional receiver
2. The Dual- Signal Receiver

The results in figure 5.1 are for the downlink connection. From figure 5.1 we find that the DSR performs better than a conventional receiver for high values of E_b/N_0. This is because when the desired signal, which is supposed to be received as the largest, is received as the smallest signal, the DSR only performs better than a conventional receiver if the signal level of the desired signal is above the noise level. For SIR = 30 dB, for small values of E_b/N_0, the signal level of the desired signal is most of the times under noise level when it is received as the small signal. For small average values of SIR, the difference between he DSR and a conventional receiver is reached for lower values of E_b/N_0. The results in figure 5.1 are generated without error correcting coding. If we implement Convolutional Coding (code rate R = 1/2, constraint length 8), which is a very often used coding technique in mobile communication, it is found from [16] that a BER of 3e-2 before decoding is required to reach BER = 1e-3 after decoding. The required E_c/N_0 to reach BER = 3e-2 before decoding is 7.8 dB. E_c is the energy per coding bit. We assume that errors that occur are random and uncorrelated after interleaving (no simulations were made to get the results with coding). The performance of the DSR was first simulated for downlink. For the uplink, the performance simulations are done for two co-channel signals, both sent at such a power level that the average received SIR between the two signals is 3 dB. The average BER for both the large and the small signal is given in figure 5.2 as a function of E_b/N_0 of the large signal. With the large signal is meant the signal from the user that is on average received as the largest of the two received signals. The small signal is the signal from the user that is on average received as the small signal.

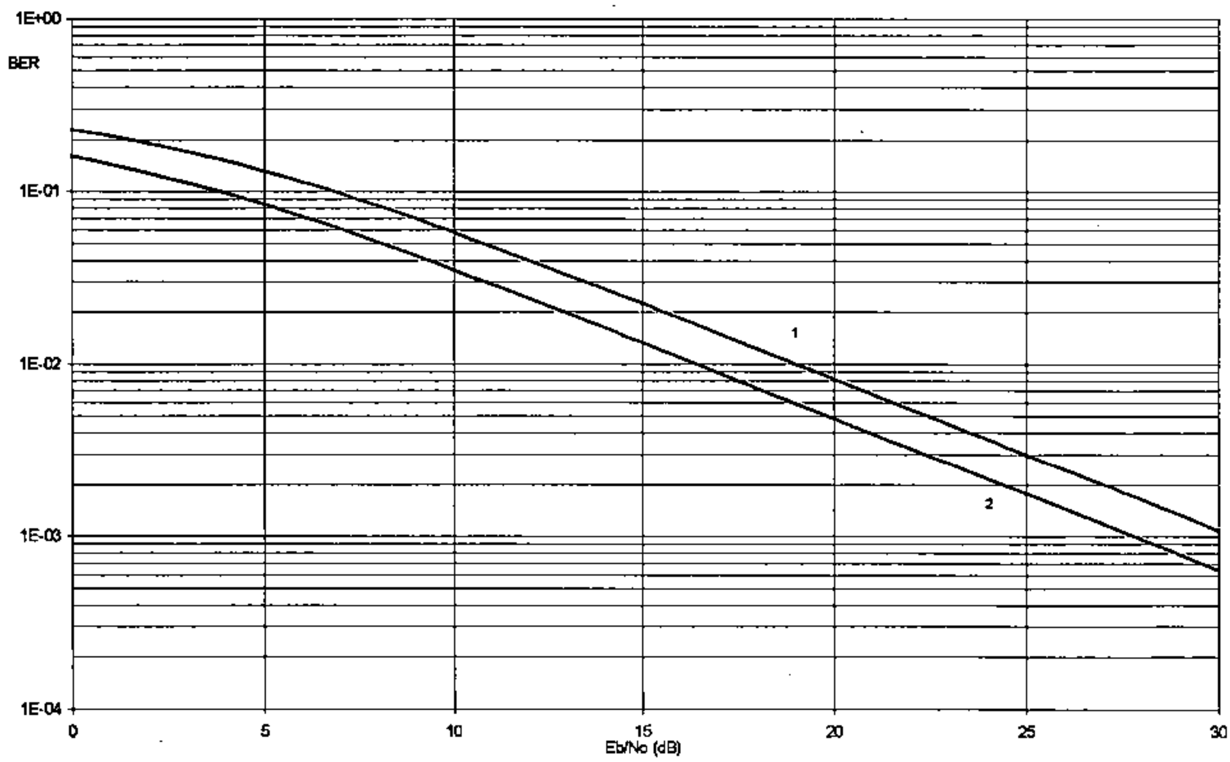

figure 5.2: Average BER for the COST 207 channel as a function of E_b/N_0 for SIR=3 dB, without antenna diversity, with:
1. BER for the small signal
2. BER for the large signal

In figure 5.2 no antenna diversity is implemented. The required E_b/N_0 for the large signal is almost 28 dB to reach a BER of 1e-3. From figure 5.2 it can be seen when error correcting coding is used, BER = 1e-3 after decoding is reached at E_c/N_0 = 11 dB. This is 3.2 dB higher than the value that was found for the downlink to reach BER = 1e-3 after decoding. The performance for the uplink connection can be improved by using antenna diversity at the Base Station. There are several methods to enhance the performance of the uplink using more than one antenna. The antenna diversity technique that was used is antenna diversity with 3 antennas using majority voting. This antenna diversity technique was chosen because this technique is relatively easy to implement. The principle of this technique is shown in figure 5.3.

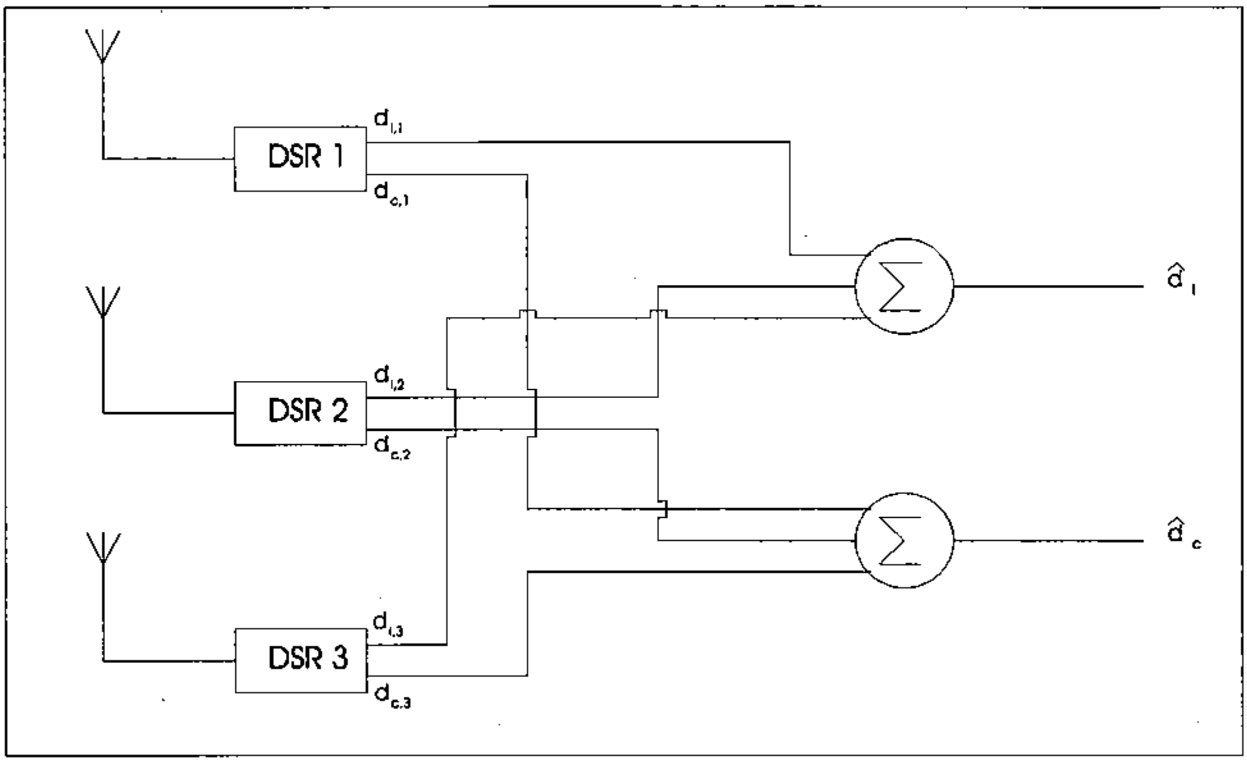

figure 5.3 Antenna diversity concept with 3 receiving antennas using majority voting

The three antennas are assumed to receive fully uncorrelated signals from a mobile station. This can be assumed if the distance between the receiving antennas is large enough. A distance between the receiving antennas of half a wavelength is enough. At 2 Ghz, which is the frequency where UMTS is going to be active, the wavelength is $\lambda = 15$ cm so a distance between the recieving antennas of 8 cm is enough. It is possible to place the receiving antennas 8 cm removed from eachother so the antenna diversity is practically implementable. Every antenna receives 2 signals both having a certain P_{ei} and P_{ec} after demodulation. P_{ei} and P_{ec} are respectively the probability of a bit error for the large and for the small signal. This results in the data signals

first antenna	$d_{i,1}$ *and* $d_{c,1}$
second antenna	$d_{i,2}$ *and* $d_{c,2}$
third antenna	$d_{i,3}$ *and* $d_{c,3}$

The received bits from the three antennas are compared for the large and the small signal. To define the resulting estimated bit for the large and for the small signal, majority voting is used. This means that the summation of the three bits determines the value of the estimated data bit. If P_1, P_2, and P_3, are respectively the BER for a bit from antenna 1, 2 and 3, the probability that the data bit that was sent by a mobile user is received incorrectly is:

$$P_{biterror} = P_1 \cdot P_2 \cdot P_3 + (1 - P_1) \cdot P_2 \cdot P_3 + P_1 \cdot (1 - P_2) \cdot P_3 + P_1 \cdot P_2 \cdot (1 - P_3) \qquad (5.1)$$

This formula is used for the large and for the small signal. Figure 5.3 gives the results for the uplink when antenna diversity is implemented.

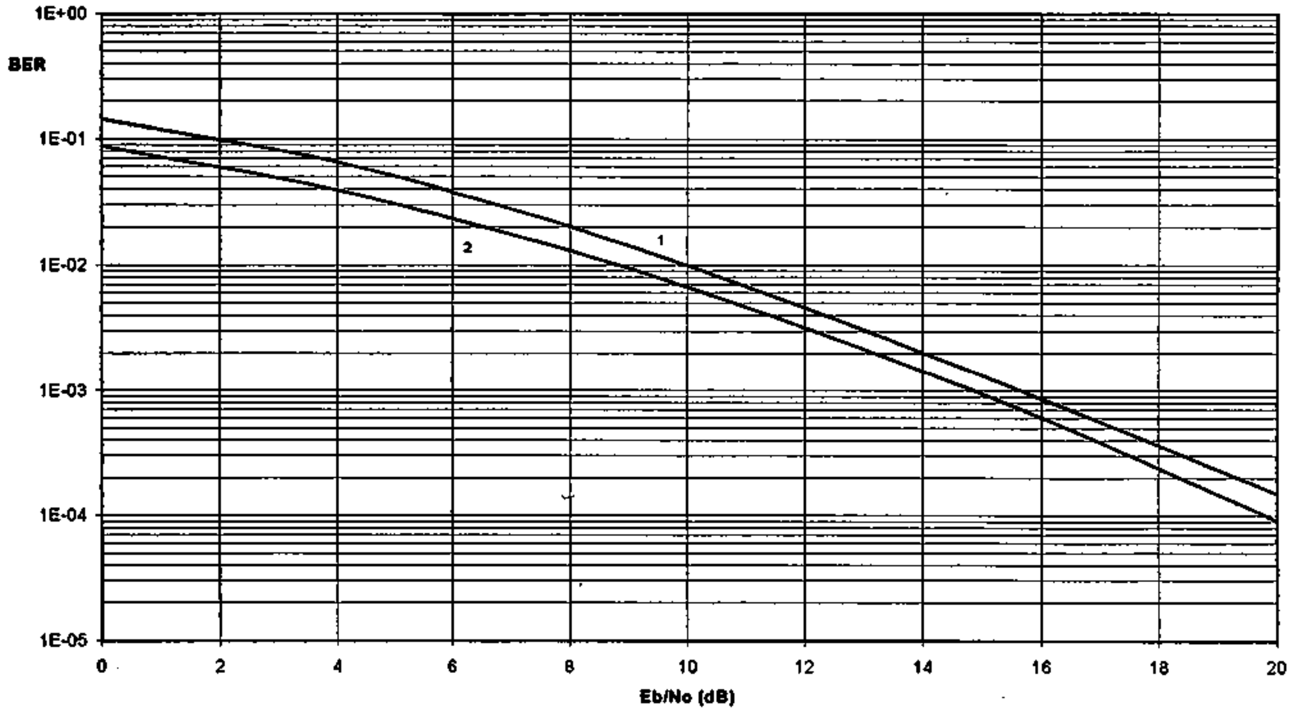

figure 5.4: Average BER for the COST 207 channel as a function of E_b/N_0 for SIR = 3 dB with antenna diversity, with:
1. BER for the small signal
2. BER for the large signal

From figure 5.4 a required E_b/N_0 of about 15 dB is found to reach the required BER of 1e-3. If Convolutional Coding is used (code rate R = 1/2, constraint length 8) we find a required E_c/N_0 of 4.9 dB to reach BER = 3e-2 before decoding and thus an BER = 1e-3 after decoding.

Notice that the performance for the small signal in figure 5.2 and 5.4 is worse than the performance of the large signal for a given value of E_b/N_0 for the large signal. This is partly caused by the fact that errors which occur in the large signal, cause errors in the small signal.

If coding is implemented and the large signal is decoded before we estimate the large signal and after that we execute the procedure to retrieve the small signal, the performance of the small signal is expected to be the same as the performance of the large signal. Therefore we take the performance of the large signal from figures 5.2 and 5.4 also for the small signal. For exact results the effect must be investigated.

Interference limited case

To derive the receiver sensitivity, we investigate the performance in the presence of interference. We consider the performance of the DSR for two received signals. A desired signal with power C, and one interfering signal with power I. The Outage Probability (= Pr{BER} >10e-3) of the desired signal is plotted as a function of C/I in figure 5.5. For the desired signal E_b/N_0 of 20 dB is used. The Outage Probability is considered instead of the average BER to have more insight in the distribution of the values of the BER. The two signals are independent and suffer from uncorrelated Rayleigh fading (modelled by the given Codit COST 207 macro cell channel model).

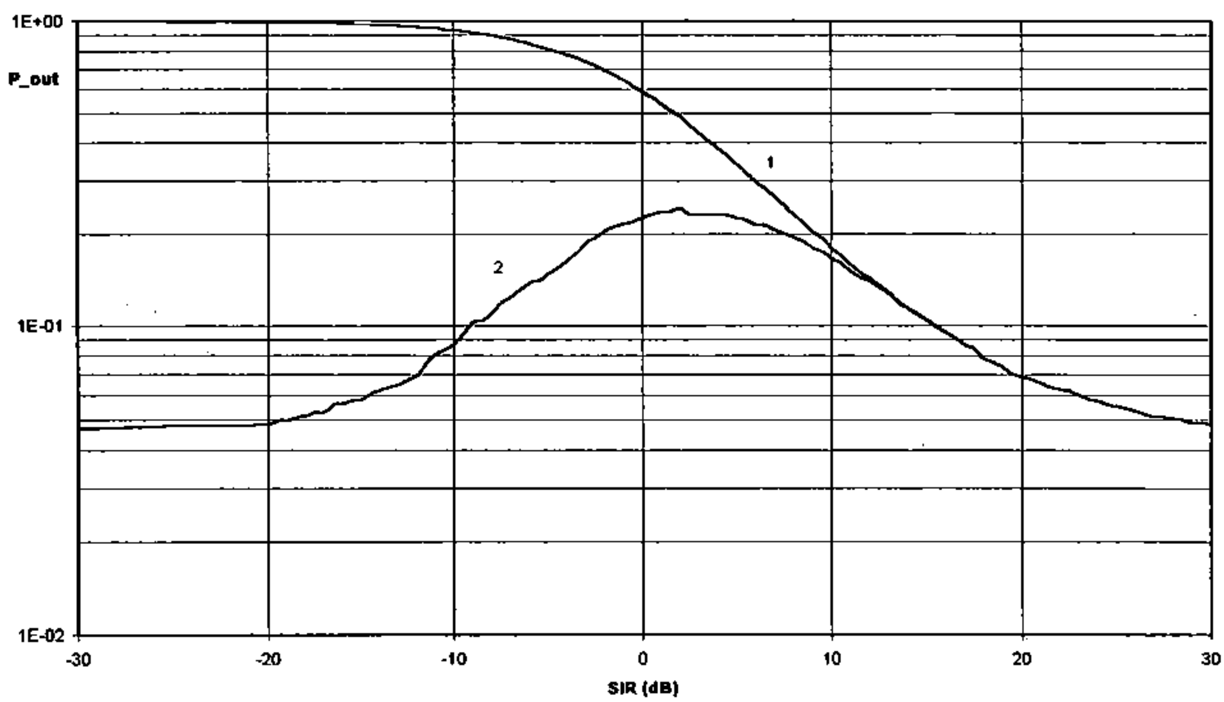

figure 5.5: Outage Probability for the COST 207 channel as a function of SIR *for E_b/N_0 =20 dB of the desired signal*
1. A conventional receiver
2. The Dual-Signal Receiver

Figure 5.5 shows that the desired signal can be received when C/I is negative. In this case a normal BPSK receiver would be captured by the interfering signal and the wanted signal would be lost. In further system performance investigation, the performance of the DSR is only investigated for the uplink. For the downlink, the DSR is also implemented but with the method that is used to model the intercell co-channel interference, the DSR has the same performance as a conventional BPSK receiver. This is described in section 5.4.2.

5.3.2 Spectrum efficiency and system capacity

In the introduction it was mentioned that a very important performance measure is the spectral efficiency and thus the capacity of a mobile communication system. In this section the spectral efficiency and capacity are given with the required E_c/N_0 to fulfil the QoS requirements. For BPSK modulation which is used in the research upto now, 1 bit per Herz can be transmitted. Convolutional Coding with coding rate R = 1/2 and constraint length 8 is used, which is the only coding technique that is used throughout this work. As mentioned, BER = 3e-2 before decoding is required to get BER = 1e-3 (QoS requirement) after decoding. To fulfil the QoS requirements for the downlink, E_c/N_0 = 7.8 dB is required and for the uplink with antenna diversity E_c/N_0 = 4.9 dB is required as we found from figure 5.1 and figure 5.4. For these values of E_c/N_0 we get the values for capacity and spectral efficiency from table 5.2. For the uplink connection, the given average value of E_c/N_0 is for the large signal. Because SIR = 3 dB, the average value of E_c/N_0 for the small signal is 3 dB lower. Capacity and spectral efficiency are defined as follows:

$$\text{Capacity } (\eta) \; = \; \frac{\text{Traffic load } (\lambda)}{\text{Bandwidth } (W)} \left[\frac{\text{users}}{\text{cel} \cdot \text{MHz}}\right]$$

$$\text{Spectral efficiency } (\upsilon) = \frac{\text{Traffic load } (\lambda) \cdot \text{Average user bitrate } (R)}{\text{Bandwidth } (W)} \left[\frac{\text{kbit / s}}{\text{cell} \cdot \text{MHz}}\right]$$

Table 5.1 gives the spectral efficiency and capacity for uplink and for downlink with the required values for E_c/N_0 to fullfill the QoS requirement that BER < 1e-3.

Table 5.1: Capacity and spectral efficiency, with coding

Service	E_c/N_0 [dB]	Bitrate [kbit/s]	Modulation	Capacity(η) [users/MHz/cell]	Spectral efficiency [kbit/s/MHz/cell]
Data (Uplink)	4.9	128	BPSK	7.8	1000
Data (Downlink)	7.8	128	BPSK	3.9	500
Speech (Uplink)	4.9	12	BPSK	83.3	1000
Speech (Downlink)	7.8	12	BPSK	41.6	500

It must be mentioned that no guard band times are taken into account. So these results are for the theoretical case that we have no guard bands in the TDMA structure and no spectrum inefficiencies because of the frequency hopping stucture.

5.3.3 Macro cell coverage analysis.

In this section the maximum allowable cell radius is calculated for the system. To do this the required values for E_b/N_0 from section 5.3.2 are used. Further analysis is based on the link budget evaluation. For the link budget evaluation we use the following formula [19, p.440]:

$$\frac{C}{N} = \frac{P_{EIRP} G_{FS} G_{AR}}{k T_{syst} B} \tag{5.2}$$

In decibel units, E_b/N_0 received at the detector input in the receiver is related to the link parameters by:

$$\left(\frac{E_b}{N_0}\right)_{dB} = (P_{EIRP})_{dBW} - (L_{FS})_{dB} + (G_{AR})_{dB} + 204 - R_{dB} - NF_{dB} + G_{HO} - G_{PC} \tag{5.3}$$

with

P_{eirp} effective isotropic radiated power

NF the noise factor of the receiver

L_{FS} the path loss

G_{AR} the antenna gain of the receiver

G_{HO} the handover gain

G_{PC} the power control margin

For the power control margin, a value of 3 dB is taken. This value is given by [17] and is taken into account because of the imperfectness of the power control.

The distance dependent part of the static macro cell propagation model is the Hata model [18] converted to a frequency band around 2 Ghz (BS antenna height equals 30m, Suburban model). The attenuation of the signal power as a function of the distance d is given by [11, p.31]:

$$L_{FS} = 123 + 36\log(d) + \zeta \qquad (5.4)$$

where d is the distance measured in km and ζ is a Gaussian stochastic variable which models the shadow fading. ζ = 6 dB is used for the shadowing margin ζ in L_{FS}. The values of E_c/N_0 that are given in the tables 5.2, 5.3, 5.4 and 5.5 are for the system with coding. Therefore, E_c/N_0 is given in the tables. For the downlink and for the uplink we used the following parameters:

Downlink :

Table 5.2: Parameters for downlink connection (convolutional coding, R = 1/2, constraint length 8)

E_c/N_0 = 7.8	P_{EIRP} = 17 dBw
G_{AT} = 10 dBi	N = 7 dB
G_{AR} = 0 dBi	L_{FS} = 123 + 36log(d) + 6

Uplink:

Table 5.3:Parameters for uplink connection (antenna diversity and coding, R = 1/2, constraint length 8)

E_c/N_0 = 4.9 dB	P_{EIRP} = 0 dBw (1W)
G_{AT} = 0 dBi G_{AR} = 10 dBi	P_{EIRP} = -9 dBW (125 mW)
N = 7 dB	L_{FS} = 123 + 36log(d) + 6

The E_c/N_0 given in table 5.2 and 5.3 is the signal to noise ratio per transmitted symbol. The results are given in the table 5.4 and 5.5 for data and speech service:

Table 5.4: Coverage analysis results for data service (channel user bitrate is 128 kbit/s)

	Required E_c/N_o (dB)	Transmitted Power (W)	ξ (dB)	Cell Radius (km)
Downlink	7.8	5	6	3.62
Uplink	4.9	1	6	2.79
Uplink	4.9	0.125	6	1.57

Table 5.5: Coverage analysis results for speech service (user bitrate is 12 kbit/s)

	Required E_c/N_o (dB)	Transmitted Power (W)	ξ (dB)	Cell Radius (km)
Downlink	7.8	5	6	6.98
Uplink	4.9	1	6	5.38
Uplink	4.9	0.125	6	3.02

The tables show the allowed cell radius for given user bitrates of 12 kbit/s and 128 kbit/s. These values result with the used coding (R=1/2) in symbol rates of respectively 24 kbit/s and 256 kbit/s.

5.4 Performance simulation results for Multiple Cell.

5.4.1 Introduction

Upto now a single cell system was considered. In this section, performance simulation results are given for the Multiple Cell system. The DSR is used again to allow two co-channel users simultaneously in the uplink. In the downlink the DSR combats co-channel interference, which gives a better performance than a conventional BPSK receiver under certain circumstances as was shown in section 5.3.1. This will not be expressed in the results that are given in this section, because for the given circumstances the performance of the DSR is the same as the performance of a conventional BPSK receiver (because the interference is modelled as noise as will be shown in section 5.4.2).

The multiple cell performance results differ from the single cell performance results because of the inter-cell interference that is present now. The model that was used to describe the co-channel intercell interference is explained. After that the same performance simulation results are given for multiple cell as they where given for single cell. At the end, the performance of the DSR in the uplink connection is compared with the performance of a conventional receiver. Performance results are also given for the non-fading AWGN channel for the DSR and a conventional receiver.

5.4.2 Macro cell hexagonal cell structure

The cellular TDMA/SFH system consists of a number of BS. Each BS not only receives a desired signal from a MS in the home cell (or two signals from two mobile stations for the DSR in the uplink), but also undesired co-channel interference from terminals located in neighbouring cells. The same effect is present for the MS, they do not only receive the desired signal from the BS in the home cell but also interference from BS in neighbouring cells. This type of interference is called inter-cell interference. This inter-cell interference is present because the available frequency band is reused to cover a wide area for mobile communication. The frequency reuse concept is used in the space domain. This means that the same frequency is repeatedly used in an area. There are therefore many co-channel cells in the system. In the hexagonal cellular structure shown in figure 5.6, frequency reuse factor 4 is used. In the cellular mobile communication system, the frequency reuse distance, which is the distance between two BS using the same frequency band, is determined by the cluster size. A cluster is a group of cells who share the total available frequency band. The possible cluster sizes are given by [19, p53]:

$$C = i^2 + j^2 + i \cdot j \qquad i,j \in \{0,1,2,\ldots,N\} \tag{5.5}$$

So cluster sizes that can be used are C = 1, C = 3, C = 4, C = 7 and even larger cluster sizes are possible. If C = 1 is taken, interference problems are enormous when a mobile user is positioned at the cell boundary. Therefore C = 1 is not practical in a multiple cell system. Results will be given for cluster size C = 4 and C = 3. Cluster size C = 7 was not investigated because C = 3 gives results that are almost as good as C = 4 so it is not interesting to investigate the performance of a system with cluster size C = 7. Notice that the spectral efficiency decreases as the cluster size increases because the available frequency band has to be shared by more cells. Cluster size C = 4 means that the available frequency band is divided in 4 equal parts and is allocated to a cluster of four cells. This cluster is repeated over the area where the mobile communication system is active. The reuse distance d, which is the distance between two base stations using the same frequency band, can be determined by [19, p.52]

$$\mathrm{d} = \sqrt{3\mathrm{C}}\mathrm{R} \tag{5.6}$$

In this formula, R is the radius of the cell. So for cluster size C = 4, the reuse distance is d = 3.46R.

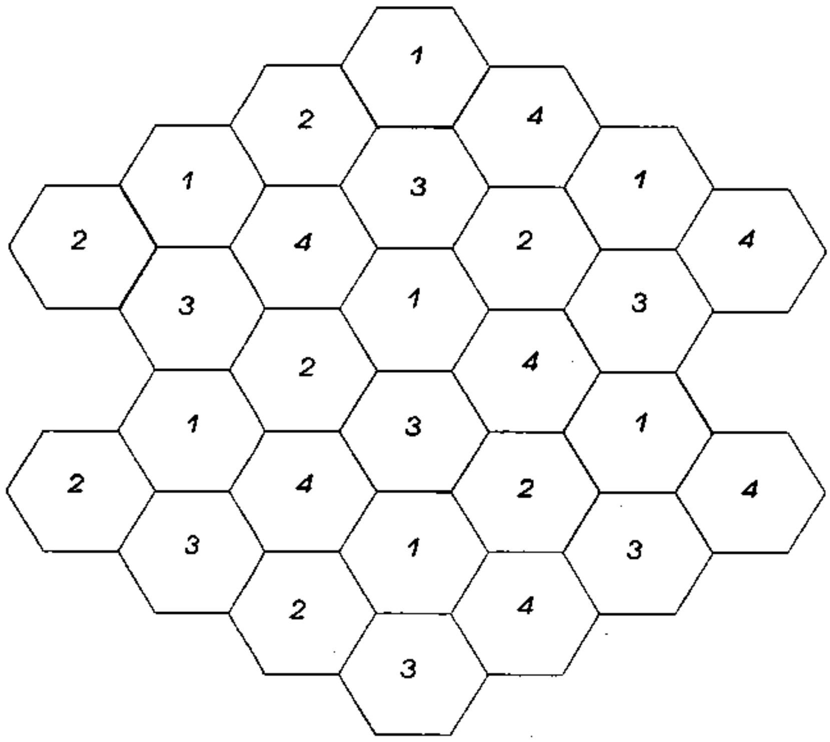

figure 5.6: frequency reuse hexagonal celllar structure cluster size C = 4

From [20] we find a model for the inter-cell Interference. This is a model for a system where power control is implemented. In the system investigated here, power control is implemented in the uplink and in the downlink. We need power control in the uplink to control the average received SIR between two co-channel signals. Power control was also taken in the downlink because the same model for co-channel interference could be used. We calculate the interference from the closest six interfering cells (see figure 5.6). This set of cells is called the first tier. This interference is the avarage interference where a user suffers from. This value is the average intereference for all the possible terminal positions, assuming that the probability of the positions in the cell is uniformly distributed. The interference from the second tier is negligible for $\beta = 4$ (β is the path loss factor and is mentioned in chapter 3). In the uplink two users share one frequency band so the total inter-cell co-channel interference from the first tier consists of the interference of 12 users. In the downlink the total co-channel interference from the first tier consists of the interference of 6 users. We first calculate the ratio of the average received power of the desired signal and the total average received power of the interfering signals from the neighbouring cells. The total average interference I from the first tier for 100 % system load is then with 6 interferers in the cells from the first tier:

Downlink:

$$I = 6 \cdot P_{RB,1sttier} \tag{5.7}$$

P_{RB} is the average interference power from one neighbouring cell. The received desired signal is S_p so the total average C/I for downlink becomes:

$$\frac{C}{I} = \frac{S_p}{6 \cdot P_{RB,1sttier}} \tag{5.8}$$

For uplink 12 interferers from the first tier are present because two users per uplink channel are present. Because of the SIR of 3 dB, the total interfering power for Uplink is 1.5 times the interfering power for downlink:

Uplink:

$$I = 9 \cdot P_{RB,1sttier} \tag{5.9}$$

so the total average C/I for uplink becomes:

$$\frac{C}{I} = \frac{S_p}{9 \cdot P_{RB,1sttier}} \tag{5.10}$$

with

$$P_{RB}(d) = 2 \cdot S_p \left[2\left(\frac{d}{R}\right)^2 \ln\left(\frac{\left(\frac{d}{R}\right)^2}{\left(\frac{d}{R}\right)^2 - 1} \right) - \frac{4\left(\frac{d}{R}\right)^4 - 6\left(\frac{d}{R}\right)^2 + 1}{2\left(\left(\frac{d}{R}\right)^2 - 1\right)^2} \right] \tag{5.11}$$

P_{RB} is the average interference power from one neighbour cell. This formula is a closed expression for β=4. The open form is given in [20]. The average interference from one cell is calculated from the formula for $P_{RB}(d)$. In the formula, d is the distance between the centres of two cells. The total interference that we calculated is assumed to be Gaussian distributed. From [15]. we find that the summation of a number of interferers which are Rayleigh distributed may be modelled by a Gaussian distributed signal if C/I is large enough and the number of interferers is large enough.

Noise limited case

The noise is assumed to be AWGN. The Gaussian distributed co-channel interference from interfering cells adds to the noise in the system. For the downlink connection figure 5.7 gives E_b/N_0 v.s. BER for single cell, multiple cell with 100 % load, 80 % load and 40 % load. The 40 % system load means that in the neighbouring co-channel cells, 40 % of the total available capacity is used. In that case the average interfering power is 40 % of the interfering power present at full system load (100 %)

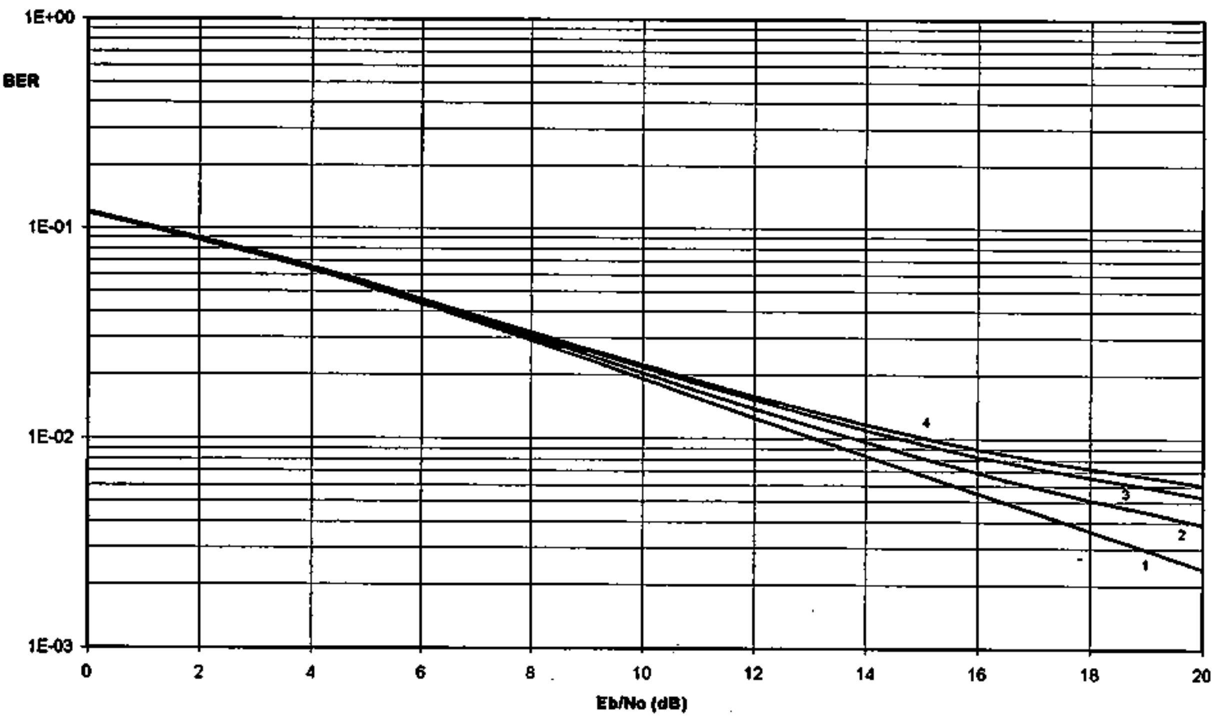

figure 5.7: Average BER for the downlink connection for the cost 207 channel for a multiple cell system with cluster size C = 4, with:
1. Single Cell
2. Multiple Cell, 40 % load
3. Multiple Cell, 80 % load
4. Multiple Cell, 100 % load

The results given in figure 5.7 are without error correcting coding. The errors that occur are assumed to be random and uncorrelated after interleaving. To reach BER 3e-2 before decoding and thus 1e-3 after decoding (with the earlier used convolutional coding), E_c/N_0 = 8.4 dB is required. Notice that since we model the inter-cell co-channel interference from neighbouring cells as AWGN, the enhancement of the DSR compared with a conventional receiver is not expressed in figure 5.7. Figure 5.7 gives the performance of a conventional BPSK receiver. In figure 5.5 the performance of the DSR is compared with the performance of a conventional BPSK receiver. This figure shows that in the case of a strong interferer the DSR performs much better than the conventional receiver. Upto now, results are given for a system with cluster size C = 4. Now results will be given for a system with cluster size C = 3. This cluster size is also investigated because the spectral efficiency is better that the cluste size C = 4. However the performance will be worse because more interference is present. Figure 5.8 gives the BER as a function of E_b/N_0 for cluster size C = 3.

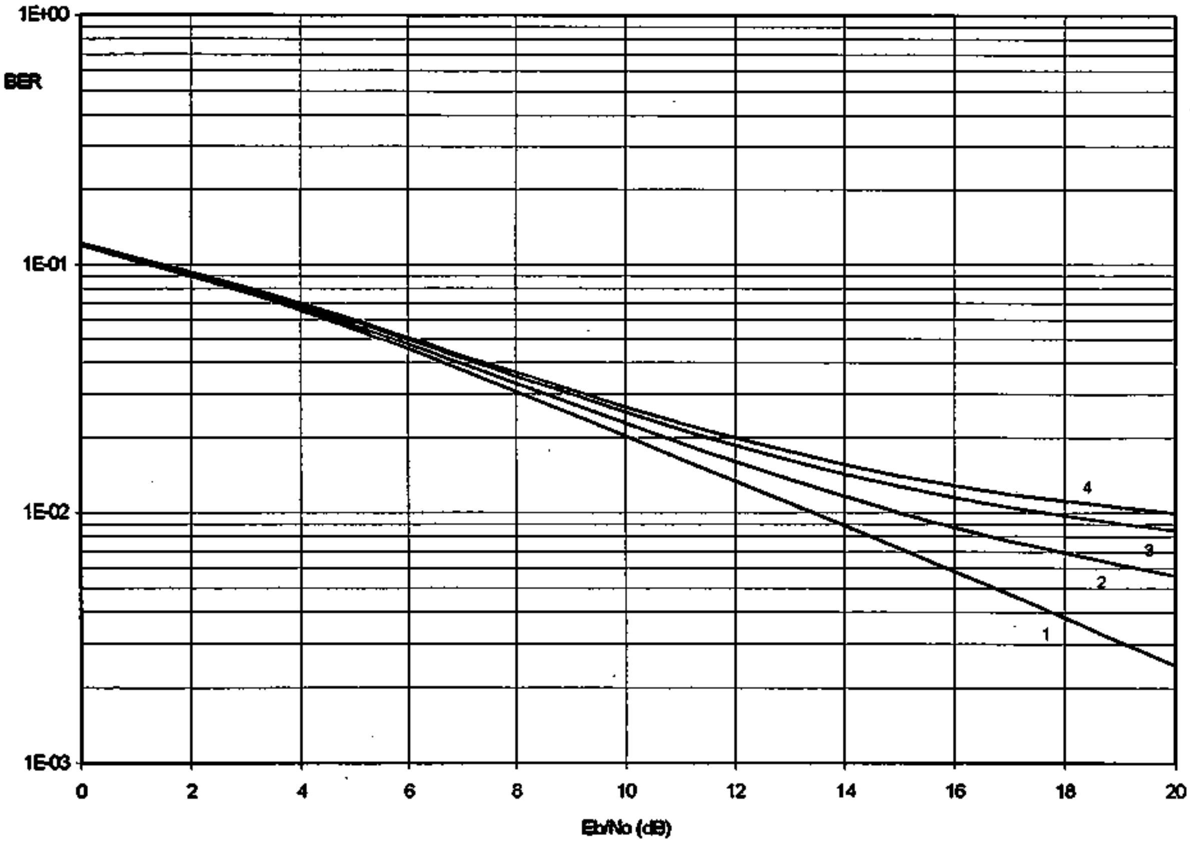

figure 5.8: Average BER for the downlink connection for the cost 207 channel for a multiple cell system with cluster size C = 3, with:
1. Single Cell
2. Multiple Cell, 40 % load
3. Multiple Cell, 80 % load
4. Multiple Cell, 100 % load

Figure 5.9 gives E_b/N_0 v.s. BER for the Uplink. In the Uplink two signals share one frequency band. Antenna diversity is used with the same technique as was described in section 5.3.1. From figure 5.7 and 5.8 we find that for small values of E_b/N_0, the influence of the interference is not so big. The difference between single cell and multiple cell 100 % load becomes larger for large E_b/N_0. This is because the interference adds to the noise and if we look for example at C/I for C = 3 with 100 % load, which results in a C/I = 14.58 dB (calculated from formula 5.10), the BER for 100 % load can not get better than BER for single cell at E_b/N_0 = 14.58 dB (BER = 8e-3). Figure 5.8 shows that line 4 goes to BER = 8e-3 for large values of E_b/N_0. Because of the error correcting coding that is used, the influence of interference is small compared with single cell because QoS requirement is fulfilled for small values of E_b/N_0. From figure 5.7 and 5.8 it can also be seen that the BER performance difference between C = 3 and C = 4 is small for small values of E_b/N_0.

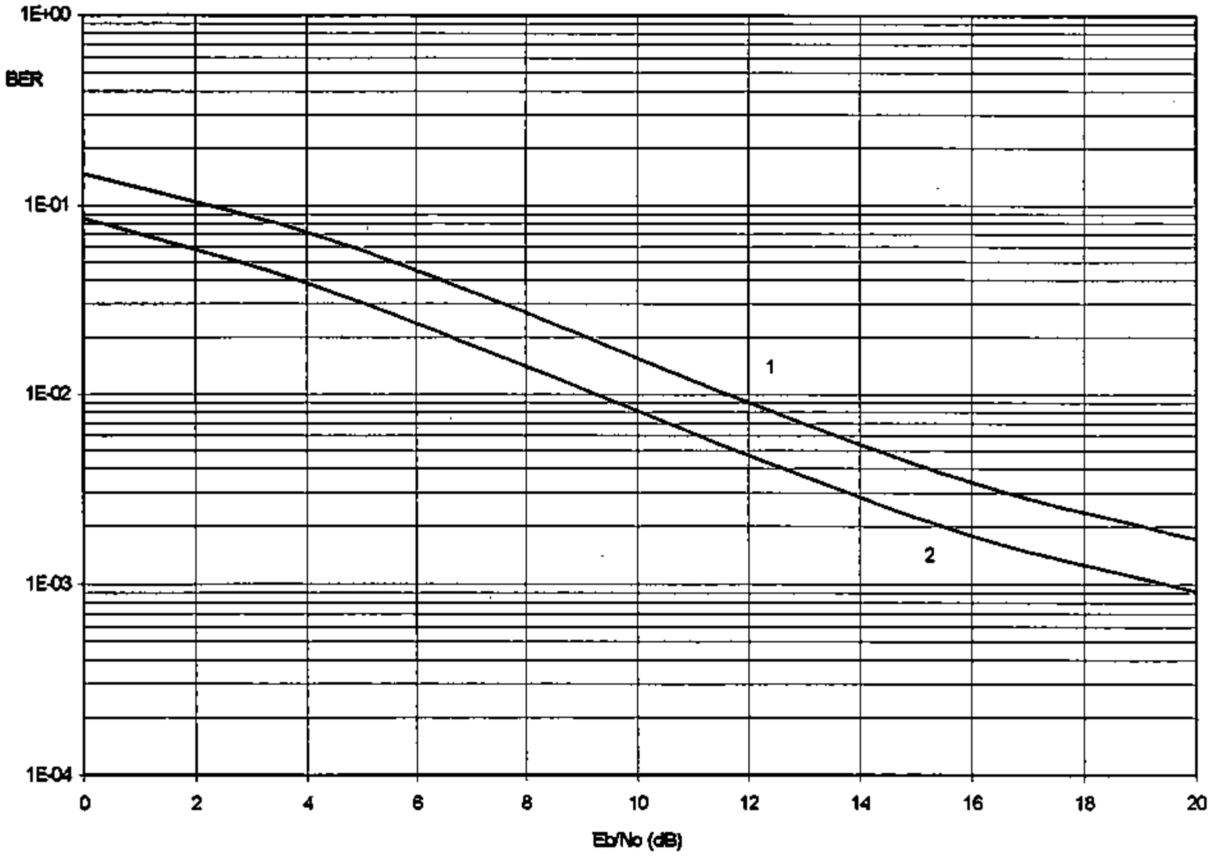

figure 5.9: Average BER as a function of E_b/N_0 (of the large signal) for Multiple Cell Uplink connection and the cost 207 channel. SIR = 3 dB, and cluster size C=4 (100 % network load), with:
1. BER for the small signal
2. BER for the large signal

The results in figure 5.9 are without error correcting coding. Using the same coding as for the Downlink we find a required E_b/N_0 = 5.0 dB for the large signal, to have a BER before decoding of 3e-2 for the large signal. Figure 5.10 gives the BER as a function of E_b/N_0 for the uplink for cluster size C = 3.

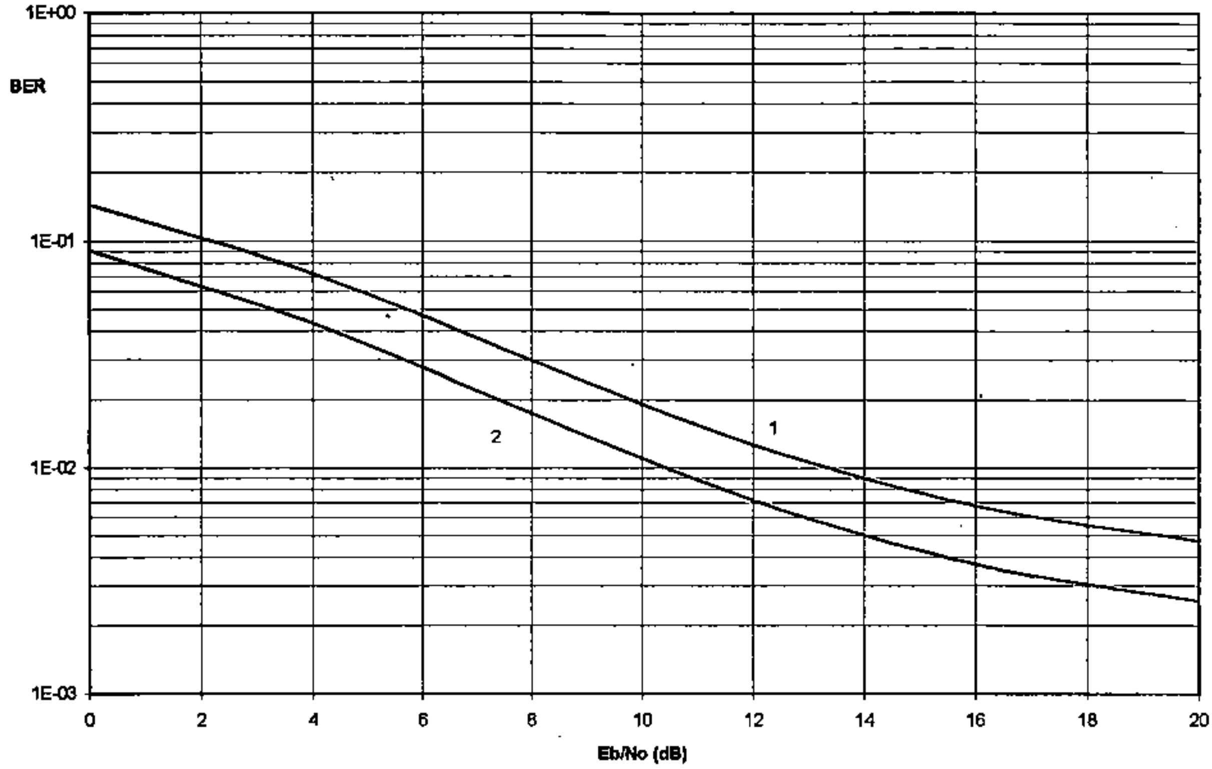

figure 5.10: Average BER as a function of E_b/N_0 (of the large signal) for Multiple Cell Uplink connection and the cost 207 channel. SIR = 3 dB, and cluster size C=3 (100 % network load), with:
1. BER for the small signal
2. BER for the large signal

From figure 5.9 and 5.10 it can be seen again that for large values of E_b/N_0 the system is limited by the co-channel interference from neighbouring cells. Notice that the BER is given as a function of E_b/N_0 of the large signal. Because SIR = 3 dB, the E_b/N_0 for the small signal is 3 dB lower (on average).

5.4.3 Spectrum efficiency and system capacity.

Introduction

In section 5.3.2, spectrum efficiency and capacity where given for single cell. The required values for E_b/N_0 were given in table 5.2. In this section results are given for a more realistic system: the multiple cell system. It was shown in section 5.4.2 that the performance of the system gets a little worse if co-channel interference from neighbouring cells is present.

Table 5.6 and 5.7 show the required E_c/N_0 to obtain the given capacity and spectral efficiency for multiple cell, for uplink and for downlink.

table 5.6 : Capacity and spectral efficiency for the Uplink, cluster size C=4 (antenna diversity is used)

Service C = 4	Required E_c/N_0 (dB) for the large signal	Bitrate [kbit/s]	Modulation	Capacity (η) [users/MHz/cell]	Spectral efficiency [kbit/s/MHz/cell]
Data	5.0	128	BPSK	1.95	250
Speech	5.0	12	BPSK	20.8	250

table 5.7: Capacity and Spectral efficiency for the Downlink, cluster size C = 4 (without antenna diversity)

Service C = 4	Required E_c/N_0 (dB)	Bitrate [kbit/s]	Modulation	Capacity (η) [users/MHz/cell]	Spectral efficiency [kbit/s/MHz/cell]
Data	8.4	128	BPSK	0.98	125
Speech	8.4	12	BPSK	10.4	125

table 5.8 : Capacity and spectral efficiency for the Uplink, cluster size C = 3 (antenna diversity is used)

Service C = 3	Required E_c/N_0 (dB) for the large signal	Bitrate [kbit/s]	Modulation	Capacity (η) [users/MHz/cell]	Spectral efficiency [kbit/s/MHz/cell]
Data	5.7	128	BPSK	2.59	333
Speech	5.7	12	BPSK	27.66	333

table 5.9: Capacity and Spectral efficiency for the Downlink, cluster size C=3 (without antenna diversity)

Service C = 3	Required E_c/N_0 (dB)	Bitrate [kbit/s]	Modulation	Capacity (η) [users/MHz/cell]	Spectral efficiency [kbit/s/MHz/cell]
Data	9.4	128	BPSK	1.30	166
Speech	9.4	12	BPSK	14.83	166

From the results presented in table 5.7 - 5.9 we see that cluster size C = 3 requires 1 dB more E_b/N_0 in the downlink and 0.7 dB in the uplink compared to C = 4, to fulfil QoS requirements.

5.4.4 Macro cell coverage analysis.

Introduction

For the macro cell system, power control for Uplink and Downlink is implemented. This power control is assumed not to be perfect, for this reason, a power control error is taken into account of 3 dB for multiple cell as was done for single cell. We assume that the terminals measure all the path loss (including the shadow fading) to Base Stations in neighbouring cells. Now the set of Base Stations that can be connected to is defined as the set of all base stations that have a path loss less than 3 dB higher than the path loss to the Base Station with the lowest path loss. The possibility of switching to another Base Station when for example heavy shadowing occurs, gives a gain compared with single cell where this is not possible. This switching property is called hand-over. This handover also occurs when a terminal moves from one cell to another. In the multiple cell system that is investigated, hand-over results in a change of frequency because of the frequency reuse structure. We call this hard hand-over. The terminal connects to a randomly chosen Base Station in the possible set of Base Stations. From the analysis definitions a value of 3 dB was given for this hard hand over gain.

Multiple Cell (with coding)

In table 5.10 - 5.13 results for maximum allowable cell size are given for cluster size C = 4 and C = 3. The same formulas are used as in section 5.3.3. Note that the hand-over gain G_{HO} was 0 dB for single cell. For the multiple cell system a hand-over gain of 3 dB was taken.

Table 5.10: Multiple Cell Coverage analysis results for data service (user bitrate is 128 kbit/s and coding rate R=1/2) with cluster size C = 4

cluster size C = 4	Required E_c/N_o (dB)	Transmitted Power (W)	ξ (dB)	Cell Radius (km)
Downlink	8.4	5	6	4.22
Uplink	5.0	1	6	3.35
Uplink	5.0	0.125	6	1.89

Table 5.11: Multiple Cell Coverage analysis results for speech service (user bitrate is 12 kbit/s and coding rate R=1/2) with cluster size C = 4

cluster size C = 4	Required E_c/N_o (dB)	Transmitted Power (W)	ξ (dB)	Cell Radius (km)
Downlink	8.4	5	6	8.15
Uplink	5.0	1	6	6.47
Uplink	5.0	0.125	6	3.64

Table 5.12: Multiple Cell Coverage analysis results for data service (user bitrate is 128 kbit/s and coding rate R=1/2) with cluster size C = 3

cluster size C = 3	Required E_c/N_o (dB)	Transmitted Power (W)	ξ (dB)	Cell Radius (km)
Downlink	9.4	5	6	3.96
Uplink	5.7	1	6	3.21
Uplink	5.7	0.125	6	1.80

Table 5.13: Multiple Cell Coverage analysis results for speech service (user bitrate is 12 kbit/s and coding rate R=1/2) with cluster size C = 3

cluster size C = 3	Required E_b/N_o (dB)	Transmitted Power (W)	ξ (dB)	Cell Radius (km)
Downlink	9.4	5	6	7.64
Uplink	5.7	1	6	6.19
Uplink	5.7	0.125	6	3.48

Notice that the allowed cell radius for C = 3 is a little smaller than for the system with cluster size C = 4. This is of course because of the extra required E_b/N_0 for C = 3 compared with C = 4. The allowed cell radius as a function of the user bitrate is given in figure 5.10, for uplink and for downlink connection for a system with cluster size C = 3, for bitrates of upto 150 kbit/s.

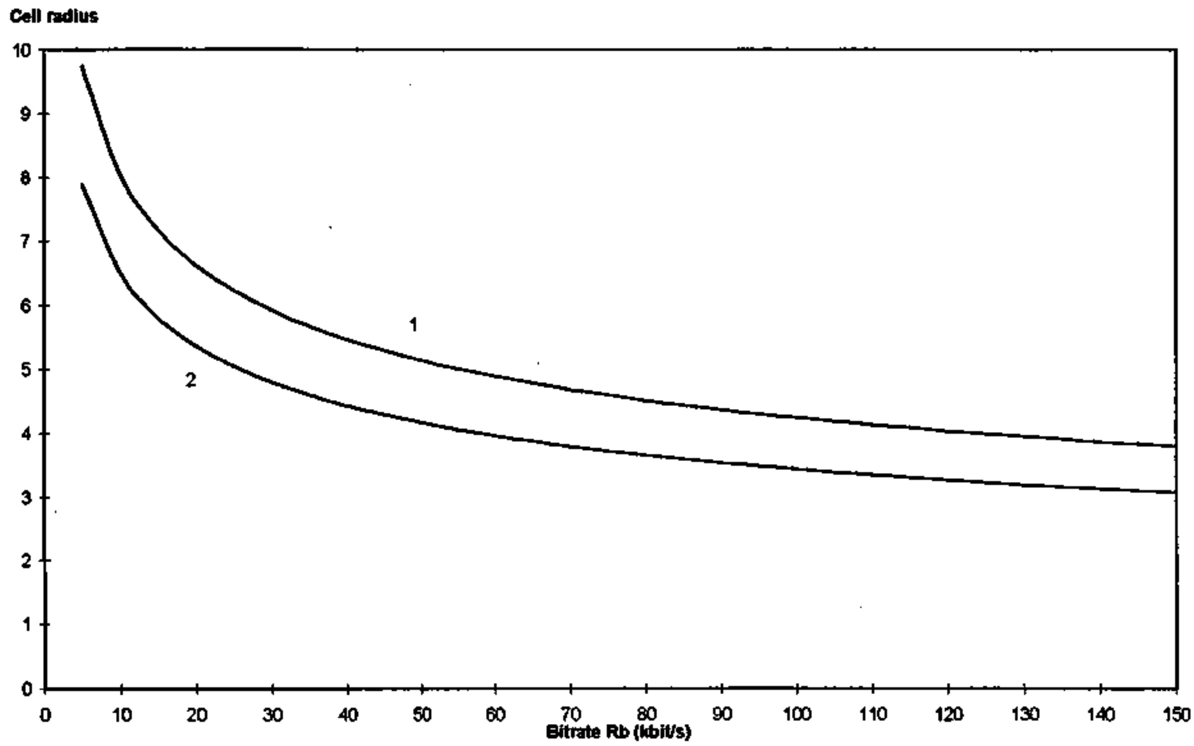

Figure 5.10: Allowed cell radius for a system with cluster size C = 3 as a function of the bitrate R_b with:
1. Downlink connection $P_{transmitted}$ = 5 Watt
2. Uplink connection $P_{transmitted}$ = 1 Watt

5.5 Implementation of Multi-Carrier technique.

The implementation of Multi Carrier allows a user to use more than one carrier simultaneously. This means that bitrates higher than the channel bitrate can be achieved. The channel bitrate is the bitrate that is achieved when a user is allocated all the timeslots in a frame. The multi carrier technique will give an enhancement in capacity allocation flexibility. This because the integration of speech service and high bitrate data service does not require to have a channel bitrate equal to the highest bitrate that must be possible to achieve. This would be the case if a user can use only one carrier simultaneously. The very high channel bitrate can give worse ISI performance than a low bitrate channel, this was given in chapter 3. Further, to implement speech service in the same system without multi carrier, a frame must be divided in a few hundred timeslots which is not the best solution to integrate speech- and dataservice in one system. This because of the fact that guard bands are needed in the TDMA structure. If the timeslots become too small, these guard bands have a negative influence on the system efficiency. How the flexibility that the multi carrier technique provides is used in the capacity allocation is explained in chapter 6.

The multi carrier technique seems to have good properties for integration of speech service of 12 kbit/s, dataservice of 128 bit/s and very high bitrate data service. However because of the restricted average transmitting power, the allowed cell size for very high bitrates is small. Figure 5.11 gives the allowed cell radius as a function of the data bitrate R_b (= information bitrate = $1/2R_c$) for a system with cluster size C = 3 for bitrates upto 2 Mbit/s. Cluster size C = 3 was chosen because is has a better spectral efficiency than a system with cluster szie C = 4 and requires only 0.7 dB more E_b/N_0 in the uplink and 1dB more E_b/N_0 in the downlink.

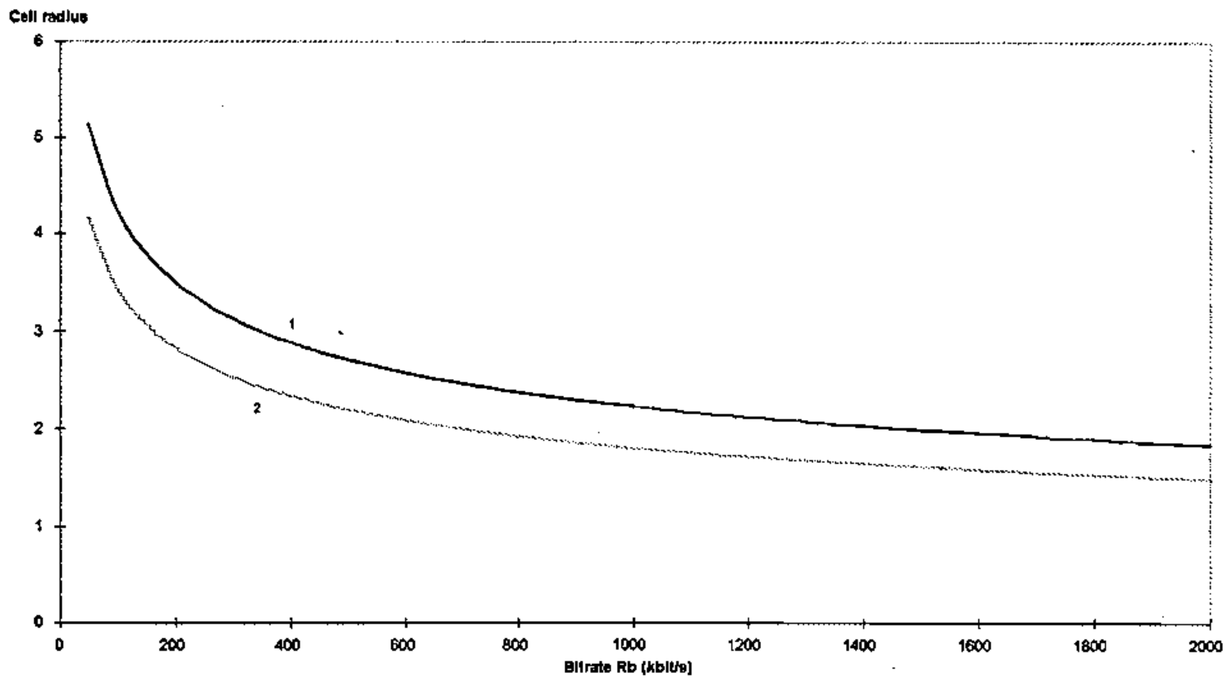

Figure 5.11: Allowed cell radius for a system with cluster size C = 3 as a function of the bitrate R_b with:
1. Downlink connection $P_{transmitted}$ = 5 Watt
2. Uplink connection $P_{transmitted}$ = 1 Watt

In the mobile communication system, terminals move with a different speed in a cell. A car for example may drive at the highway with a speed of 100 km/h. If the cell radius is too small, the terminal would have to switch often from one Base Station to another. This hand-over costs a lot of overhead information, therefore it is practical to keep the hand-over rate for a terminal as low as possible. Further, in the rural environment it may be economically very unattractive to make a cellular communication system with cell radius of for example 1 km, which is about the maximum allowable cell radius for a bitrate of 2 Mbit/s.

In future mobile communication systems, the available bandwidth will be divided in three parts: pico-, micro- and macrocell part. The multi carrier technique makes switching between the three parts at various bitrates possible because of the flexibility in allocating more than 1 carrier simultaneously to a user in case of high bitrates, so the channel bitrate can stay the same for the three cell layers.

5.6 Conclusions and recommendations

5.6.1 Introduction

In section 5.1 to 5.5 the performance of the MC-TDMA/SFH-DSR was considered. In this section the results will be given for the same system parameters using a conventional BPSK receiver instead of a DSR in the uplink. The decision criteria to find the optimum value of the average received SIR between the two co-channel users are discussed. Further, system performance results are given for the DSR and for a conventional BPSK receiver in a non-fading AWGN channel and they are compared with the results for the Rayleigh fading channel.

5.6.2 Comparison between the DSR and a conventional BPSK receiver

In this section the performance of the DSR is compared to the BER performance of a conventional BPSK receiver. This is done for the given measured Rayleigh fading channel model. Figure 5.12 gives the BER for the large signal for the DSR with SIR = 3 dB and also gives the BER for the conventional receiver when no co-channel interferer is present.

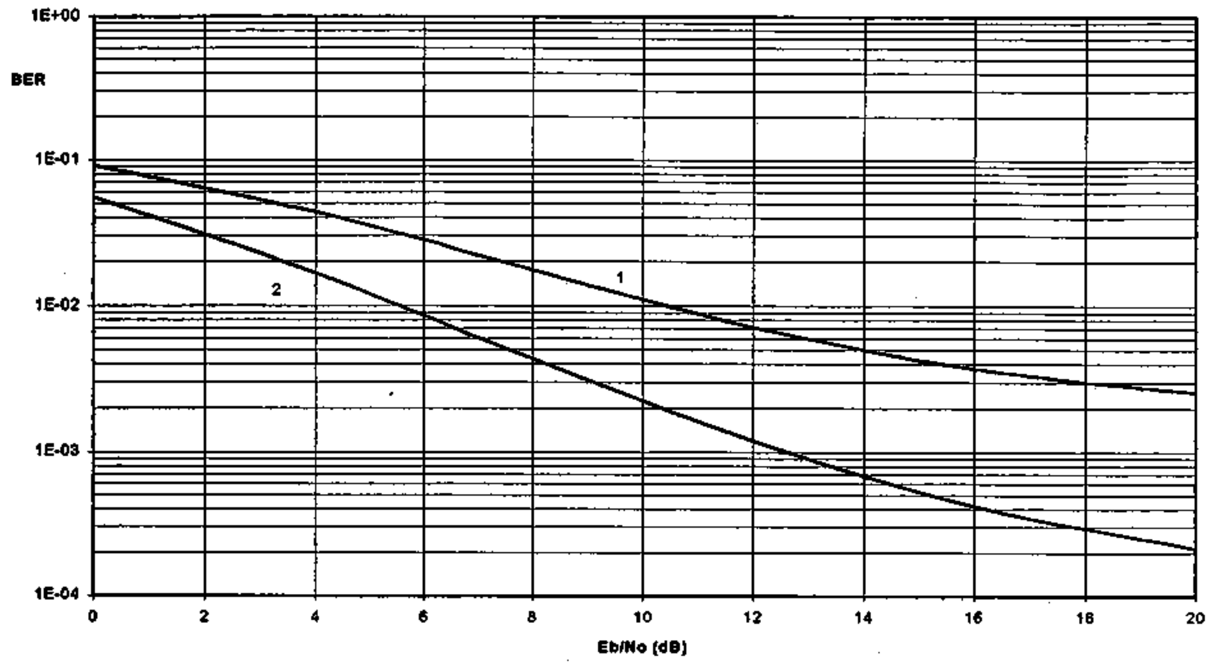

figure 5.12 BER as a function of E_b/N_0 in the COST207 channel with cluster size C = 3 for 100 % system load, with:
1. BER for the large signal for the DSR with SIR = 3 dB
2. BER for a conventional BPSK receiver

Convolutional coding (R = 1/2 and constraint length = 8) is considered again. If we compare the required E_c/N_0 of the conventional BPSK receiver with the required E_c/N_0 for the large signal using the DSR to reach BER = 3e-2 before decoding, we see a difference of 3.7 dB between the large signal of the DSR and the signal of the conventional BPSK receiver. If we assume that the performance for the small signal is the same as the performance of the large signal (when coding is used in the large signal before the small signal is retrieved) this would imply that, because SIR = 3 dB, the required E_c/N_0 of the small signal is only 3.7 - 3 = 0.7 dB higher than the required E_c/N_0 if a conventional receiver is used to reach the required BER = 3e-2 before decoding. So for the small signal the presence of the large signal costs only 0.7 dB.

Figure 5.12 shows that because of the enhance of spectral efficiency (by a factor 2) by allowing two co-channel users simultaneously, the required E_b/N_0 is higher.

What can we do with the obtained results?

At this moment GSM uses a conventional receiver (which has no DSR properties) and another modulation type than BPSK wich was investigated here. In GSM, the modulation type Gaussian

Minimum Shift Keying (GMSK) is used. The results that are obtained in this chapter are very interesting if we look at the main goal of implementing the DSR in the uplink connection : capacity enhancement. Therefore it is worth investigating the performance of the DSR for GMSK modulation. A SIR = 3 dB was chosen throughout the whole investigation of the performance of the DSR in the uplink. This is not necessarily the optimum value.

We take a look at an average SIR of 0 dB. Because of the fading characteristics of the channel, if the average received power of the two signals is the same, the average of all the instantaneous received signal to interference ratios is not 0 dB. The performance of values of 0 dB < SIR < 6 dB should be investigated. An aspect which must also be taken into account is that if the signals are sent with an average SIR = 0 dB, the two signals will alternatively be received as the large signal. This may have a negative effect on the stability of the DSR and the performance in practical situations. These are very important aspects when the performance of the DSR in GSM is considered.

In an AWGN channel there is no fading so the received SIR is constant. The difference in performance between the DSR and a conventional receiver is shown in figure 5.12.

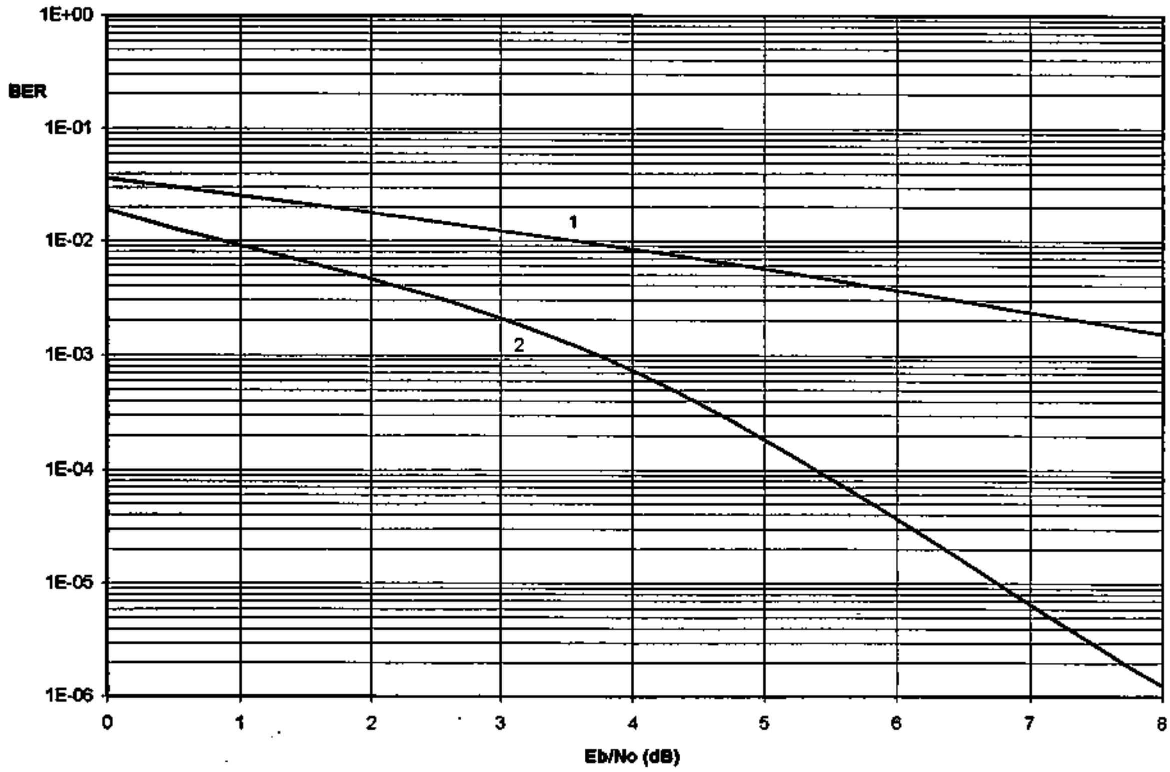

figure 5.12 BER as a function of E_b/N_0 in an AWGN channel without fading,with cluster size C = 3 for 100 % system load, with:
1. BER for the large signal for the DSR with SIR = 3 dB
2. BER for a conventional receiver with BPSK modulation

In the non fading AWGN channel the difference between the BER performance of the DSR and the conventional receiver is smaller than in the fading channel. In the fading channel, the average received SIR is 3 dB but the instantaneous values of SIR, vary a lot, values for SIR < 0 dB and SIR > 3 dB occur. This means that because of the fact that in the fading channel sometimes low values of SIR (the desired signal may even be received as the small signal) occur, higher values for P_{ec} occur which results in a higher average BER than when SIR would be constant 3 dB. In the non-fading AWGN channel the SIR is constant so the difference between the DSR and a conventional receiver is smaller than in the fading channel.

6. Capacity allocation protocol for integration of speech-, data- and packet service

6.1 Introduction

In chapter 1 the requirements for the future mobile communication system UMTS were described. It was mentioned that different types of services must be integrated in one system. The performance analysis of the proposed MC-TDMA/SFH-DSR system in terms of spectral efficiency, capacity and coverage analysis was derived in chapter 5. In this whole quantitative analysis of the physical layer, it was assumed that there is a connection between the BS and the MS. Another aspect of the mobile communication is the way how the Mobile Stations get a connection with the BS. This phenomenon is called Multiple Access. In this section, the way how the different types of traffic get a connection with a BS is described. For existing mobile communication systems which have to serve one single type of traffic, appropriate protocols have been developed. However, for UMTS different types of traffic have to be integrated in one system. Since there is no protocol that is appropriate for the three types of services, this means that besides of the protocol that is used for each individual service, a *"capacity allocation protocol"* must be designed to let the users with their different types of traffic share one frequency band in an efficient way. A qualitative proposal is given for this capacity allocation protocol that is suitable for UMTS in outdoor environment. First the three types of traffic are described. Then the structure of the TDMA/SFH format will be given. After that, the capacity allocation will be described for the three services for a conventional receiver in the uplink and for the uplink connection using the DSR to enhance the uplink capacity. At last the Multiple Access Protocols that are proposed for the three types of traffic are described.

6.2 Description of the three types of services

The following three services will be used in the system:

- Speech service with a bitrate rate of 25 kbit/s
- Data service with a bitrate of 200 kbit/s upto more than 2 Mbit/s

• Packet service

To be able to find an appropriate capacity allocation protocol, these three services are described first.

Speech service:

To obtain the characteristics of the allocation of capacity needed for this type of traffic, we describe the characteristics of voice communication. During a telephone conversation, two people talk alternatively for a period. This period can be a few seconds upto a few minutes or even more. During the telephone call, both users must be able to speak at any moment and delay constraints are very severe. A connection has to be available during the whole telephone conversation in uplink and downlink direction simultaneously. This type of connection is called a full duplex connection. Throughout this chapter, speech service will be indicated by *traffic 1*. This full duplex connection is also used in GSM. In the Digital European Cordless Telecommunications (DECT) system, Digital Speech Interpolation (DSI) is used and there is no full duplex connection for speech service. This technique provides a more efficient use of the capacity than the full duplex connection that is used in GSM and in the system proposed in this chapter. It is worth investigating the use of DSI in the system for UMTS.

Data service:

In this chapter, data traffic is defined as traffic with a bitrate of α*200 kbit/s (α is an integer). This data traffic will be indicated from now on by *traffic 2*. In *traffic 2*, uplink and downlink are not necessarily in balance like in *traffic 1* and the possibility of sending uplink and downlink traffic simultaneously (by one user) is not required. *Traffic 2* can be characterized by a continuous transmission of data in one of both directions at a time for a certain period. A half duplex connection is therefore sufficient.

Packet service:

The packet service is indicated as *traffic 3*. The information that is sent is transmitted in packets. Packets have a fixed size and one packet fits exactly in one timeslot. The packets are not necessarily received in a fixed order like in the data- and speech service. *Traffic 3* has a bursty character. This means that the peak to average ratio of the transmission speed is large. Constraints towards delay are less severe than for data service.

6.3 Time Division and Frequency Division structure

In chapter 1 the requirements for UMTS were described. These were given as:

- the system is compatible with the second generation mobile communication system GSM
- variable bitrates are possible in an efficient way
- bitrates of 25 kbit/s upto more than 2 Mbit/s are possible

Let us start with the first requirement. The Time Division structure and Frequency Division structure of the second generation mobile communication system GSM are given in table 6.1.

table 6.1 GSM Time Division and Frequency Division structure

system bandwidth	25 MHz (uplink) + 25 Mhz (downlink)
number of timeslots per frame	8
frame duration	4.615 ms
timeslot duration	0.58 ms
number of bits per timeslot	116
number of frequency bands	124 (the same for uplink and downlink)
user bit rate	25 kbit/s
maximum number of users	992

The number of frequency bands is obtained by dividing 25 Mhz by 8*25 kHz. This results in 125 frequency bands. One frequency band is used for other purpose (to be explained later on in this chapter), so 124 frequency bands are available for communication. The TDMA structure given in table 6.1 is the basis for the system. This Time Division structure will be used for the proposed system MC-TDMA/SFH-DSR for UMTS. However the available bandwidth for UMTS is 60 MHz + 60 MHz so the number of frequency bands is much higher in the MC-TDMA/SFH-DSR and the maximum number of *traffic 1* users is much higher. In the system for UMTS no distinction is made between a bandwidth for uplink and downlink. There is just an available frequency band of 120 MHz to be shared by uplink and downlink connection. The Time Division and Frequency Division structure for MC-TDMA/SFH-DSR is given in table 6.2.

table 6.2 MC-TDMA/SFH, Time Division and Frequency Division structure

system bandwidth	120 MHz (shared by uplink and downlink)
number of timeslots per frame	8
frame duration	4.615 ms
timeslot duration	0.58 ms
number of bits per timeslot	116
number of frequency bands	599 (shared by uplink and downlink)
user bit rate (*traffic 1*)	25 kbit/s
maximum number of *traffic 1* users	2396

The given maximum number of *traffic 1* users is given for the system when the DSR is not yet implemented. In section 6.5 the maximum number of *traffic 1* users will be given for the system with the DSR in the uplink.

The number of frequency bands is obtained by dividing 120 MHz by 8*25 kHz. This results in 600 frequency bands. One band is used for other purpose, which will be explained later on in this chapter, so 599 frequency bands are left for communication. the maximum number of voice users is :

(8 timeslots per frame) * (599 frequency bands)/2 (*traffic 1* is full duplex) = 2396.

The maximum number of users mentioned here is valid if the whole system bandwidth is available. If we use frequency reuse patterns with cluster size C (C was described in chapter 5), the maximum number of *traffic 1* users per cell is 2396/C.

Frequency hopping structure.

In the previous section it was mentioned that the total available frequency band for uplink and for downlink is divided in 599 frequency bands. In the proposed protocol for UMTS, frequency hopping is implemented. This means that the information is not sent at one carrier frequency but the carrier hops from one carrier frequency to another following a hopping pattern. An overview of Multiple Access Protocols was given in chapter 2 and it was mentioned that orthogonal hopping patterns are used for the MC-TDMA/SFH-DSR. This means that every user has its own hopping pattern in a timeslot and the hopping patterns are designed in such a way that two hopping patterns are never active in the same frequency band simultaneously. Using orthogonal hopping patterns, the number of hopping patterns is equal to the number of frequency bands.

6.4 Capacity allocation for the Uplink and the Downlink.

6.4.1 Introduction

In section 6.2, the three services were described as three types of traffic. This was done to be able to design a system that provides the integration of three types of services as efficiently as possible. This description of the types of traffic, together with the description of the Time Division and Frequency Division determines the way of allocating capacity to the different types of services. In this section it is described how the available capacity is divided in three parts to be able to serve the three kinds of services in an efficient way. To obtain a good overall system performance two major aspects have to be investigated:

• How to share the available bandwidth by the three services? This will be treated in this section.

• If this is clear, how to allocate the available capacity to the different types of traffic (which multiple access protocol is proposed for each type of traffic). This is described in section 6.5.

6.4.2 Capacity division for the three types of services, using a conventional receiver

The available capacity is expressed in terms of hopping patterns and time slots. The total available channel consists of 599 frequency bands and a frame is divided into 8 timeslots. In the FH protocol that is used, we can assign 599 hopping patterns to the users. An overview of this capacity is given in figure 6.1. It is described how the uplink capacity is allocated to users for the three types of traffic:

traffic 1:

There is no separate frequency band for uplink and for downlink. Suppose there are no users in the system and a user wants to set-up a call. This user is allocated a hopping pattern say hopping pattern #1 and two timeslots say timeslot #1 for the downlink connection, and timeslot #2 for the uplink connection (see figure 6.1). If a second user wants to set-up a call, this user is allocated the same hopping pattern #1 and timeslot #3 for downlink connection and timeslot #4 for uplink connection. This goes on until in all the timeslots hopping pattern #1 is occupied. So user #5 gets a new hopping pattern #2 and timeslots #1 and #2, see figure 6.1. The dark grey blocks represent the downlink connection and the light grey blocks represent the uplink connection. The management of the allocation of the hopping patterns and timeslots to the users is done by the BS. The BS sends information to the mobile stations which timeslot and hopping pattern is allocated to that user.

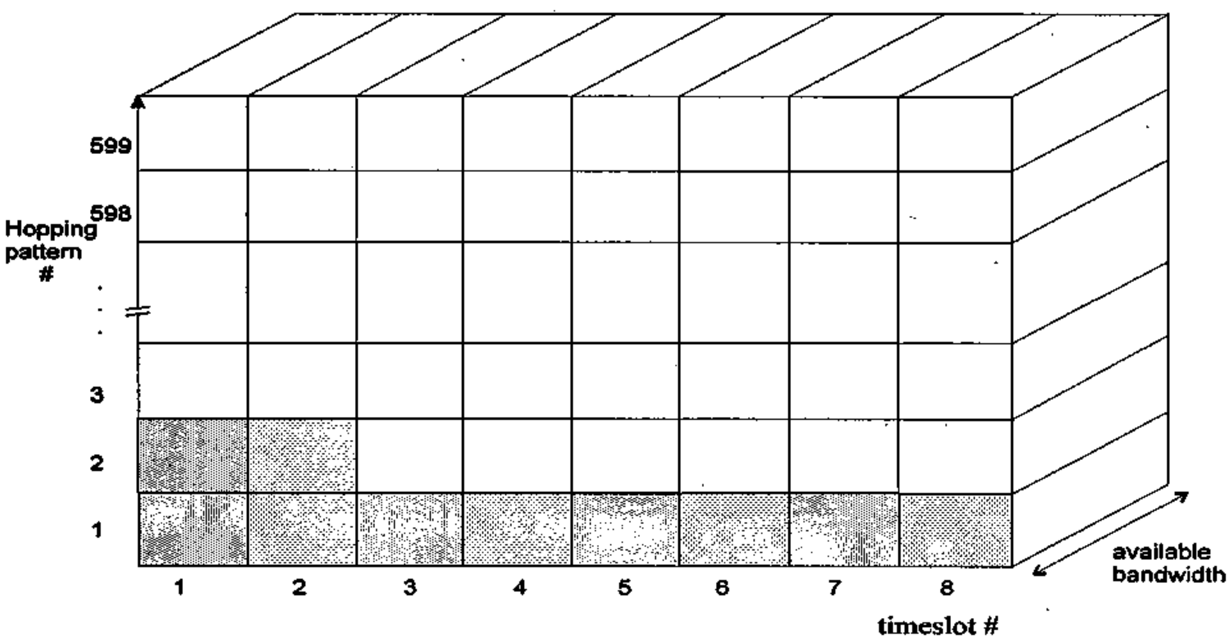

figure 6.1 capacity allocation to traffic 1 users with: the light blocks indicating the uplink connection and the dark blocks indicating the downlink connection

If a user ends his conversation, a hopping pattern is free in a certain timeslot from that moment and if a new user arrives, the system starts to check whether there are timeslots which are not occupied, starting at hopping pattern #1, after that hopping pattern #2 and so on. The reason why we do this, is to occupy for every hopping pattern the timeslots as efficient as possible to leave as many hopping patterns unused as possible. The capacity allocation for *traffic 1* is Time Division Duplex (TDD): timeslots in a frame can be used for downlink as well as for uplink. Another possible technique is Frequency Division Duplex (FDD). Using FDD, the uplink and downlink connection are realized in different frequency bands simultaneously. FDD is illustrated in section 6.4.3, in figure 6.4.

traffic 2:

In section 6.2, *traffic 2* was described. A *traffic 2* user can use bit rates of 200 kbit/s upto more than 2 Mbit/s. Variable bitrate allocation is possible in steps of 200 kbit/s by varying the number of hopping patterns that are allocated simultaneously to one user. This is the Multi Carrier technique that was mentioned earlier in chapter 5. To reach for example a bit rate of 1 Mbit/s a user is allocated 5 hopping patterns simultaneously and every timeslot in a frame is occupied. In figure 6.2 the 5 *traffic 1* users of figure 6.1 are active. Suppose a *traffic 2* user needs a 200 kbit/s uplink connection. This user is allocated one hopping pattern and the user is allocated every timeslot in a frame during the call. The method of allocating the capacity is the same as for *traffic1* but now we start at hopping pattern #599. This principle is shown in figure 6.2.

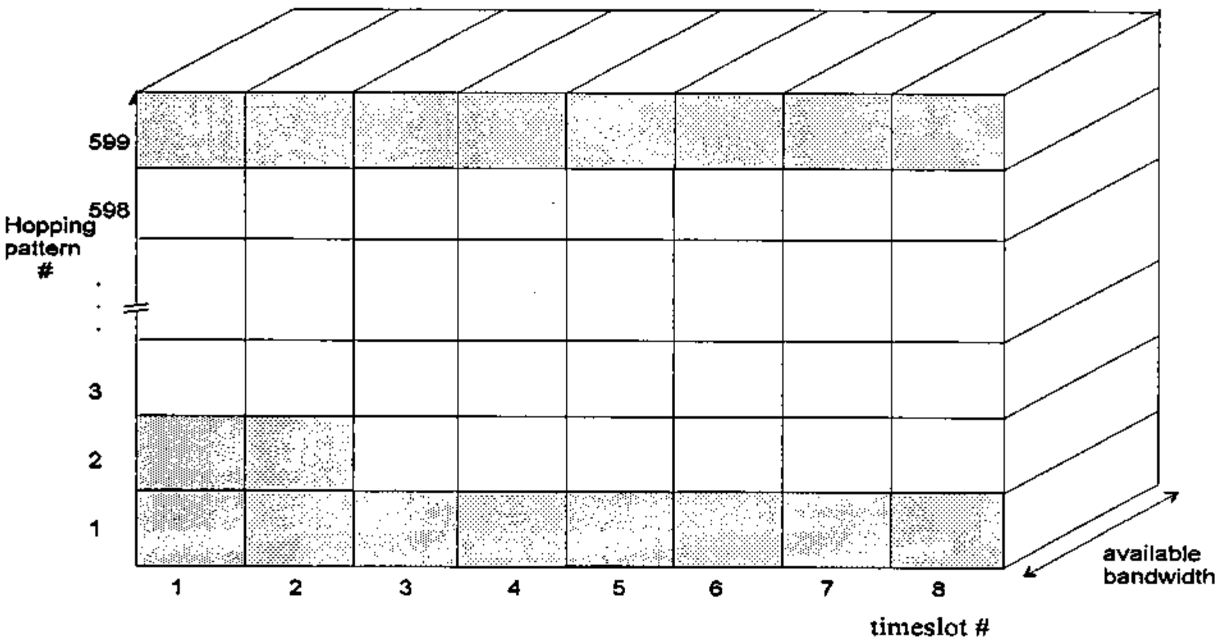

figure 6.2 uplink capacity allocation to a high bitrate traffic 2 user in a system where 5 traffic 1 users are active

Notice that this way of integrating *traffic 1* and *traffic 2* in one system leaves a number of hopping patterns unused (which means that all the timeslots available for these hopping patterns). This part of the total available capacity is called the buffer capacity and is used by *traffic 3*. How traffic 3 uses the buffer capacity is explained in section 6.5. Figure 6.3 gives an overview of the way how the total available capacity is divided.

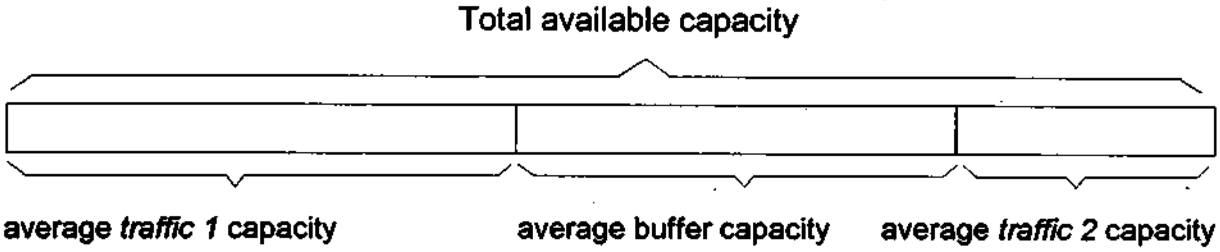

figure 6.3 overview of capacity division

Because the capacity division of figure 6.3 is based on average values, at certain moments the instantaneous values of the capacity desired by the kinds of traffic mentioned here may differ from the average value.

If we assign a fixed amount of capacity to the different types of traffic there is no flexibility. To use the capacity in an efficient way the capacity allocation must be flexible. This means that the boundaries between the *traffic 1* capacity and the buffer capacity and between the buffer capacity and the *traffic 2* capacity should be movable.

6.4.3 Capacity division for the three types of services, using the DSR

It was mentioned that the DSR technique provides a double capacity in the uplink connection compared with a conventional receiver but not in downlink connection (the effects of this technique on the performance of the system is explained in chapter 5). This means that we have to consider the three different types of traffic again to be able to propose an appropriate capacity allocation model.

If we start with *traffic 1* we see that, as was mentioned earlier in this chapter, the uplink and downlink capacity in use are in balance (full duplex connection). This means that if we double the uplink capacity without any changes, the uplink capacity for *traffic 1* is twice as big as the downlink capacity. What we have to do is getting the effective capacity for uplink and downlink in balance. We can do this by allocating twice as many hopping patterns to downlink as to uplink. Figure 6.4 gives an overview of the capacity utilisation when 4 *traffic 1* users are active in the system. Figure 6.4 shows a FDD capacity allocation for *traffic 1*. In figure 6.1 and

6.2 TDD was used. In the development of the system, timing aspects also have to be considered to decide whether TDD or FDD is more practical to use.

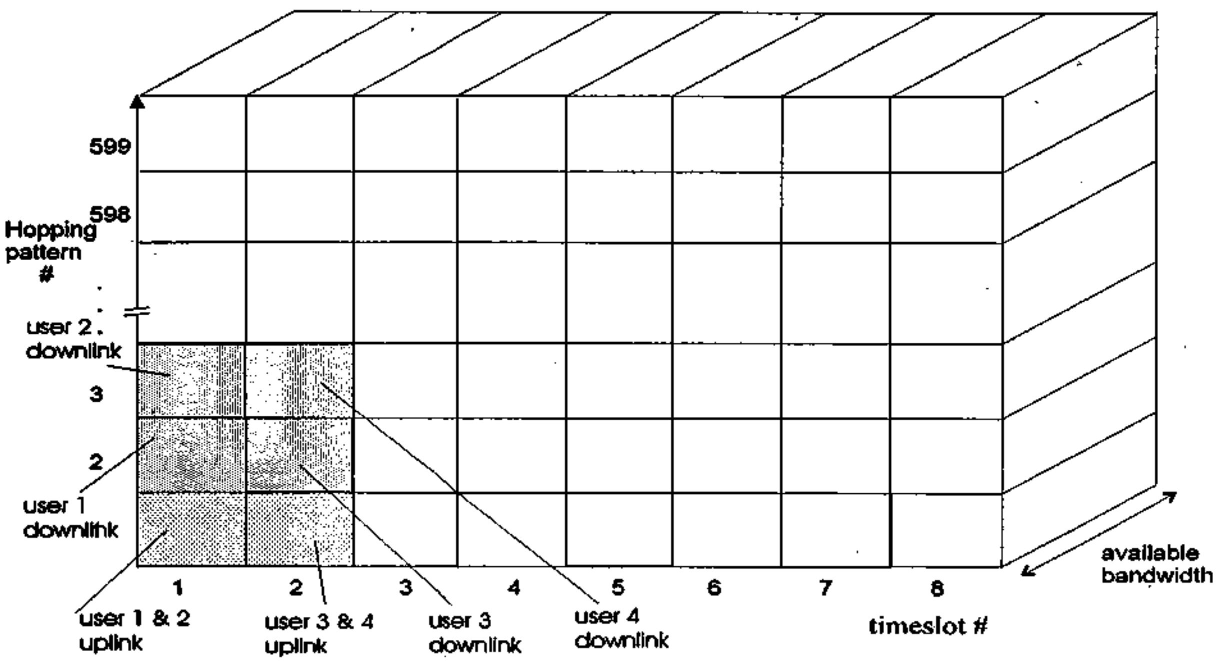

figure 6.4 capacity allocation to 4 traffic 1 users using the DSR in the uplink: the light blocks indicating the uplink connection the dark block indicating the downlink connection

For *traffic 2* users, the capacity doesn't have to be in balance, so the capacity allocation stays the same as with a conventional receiver except that two users can be allowed simultaneously in the uplink. For *traffic 3* it is very difficult to allow two users in a frequency band simultaneously in an organized way because of the slotted ALOHA protocol that is used (this is explained in section 6.5). The DSR provides for *traffic 3* the possibility to retrieve two signals if a collision occurs between two users who transmit at the same moment.

6.5 Multiple access protocols for the different types of traffic

In section 6.4 it was described how the available capacity is allocated to the different types of traffic. Since the mobile terminals have to share one communication channel (the available bandwidth), and these terminals don't know of eachother when they are going to use the communication channel, rules must be made to control the multiple access process. Numbers of protocols are designed to use the available bandwidth in an efficient way. There is not one single protocol which is suitable for all three types of services. Depending on the type of traffic and the circumstances (for example the cell size), some protocols perform better than others. In this section methods to get access to the capacity that is available will be proposed for the different types of traffic.

traffic 1 :

In section 6.2 it was mentioned that in the available bandwidth of 120 MHz, 600 frequency bands are available each having a bandwidth of 200 kHz. One of these 600 frequency bands is reserved for the capacity allocation for *traffic 1*. This channel is used as a common channel for all the *traffic 1* users. If a user wants to set-up a telephone call, it sends a message to the BS using this reservation channel. For the access to this channel, the slotted ALOHA protocol is used [9]. If a user gets through, which happens when no collision occurs (no other user sent a packet in the same timeslot on the same frequency), the BS selects a channel in the available capacity. The DSR can be used in the common channel to be able to receive two signals that were sent simultaneously. The way how the channels are selected was described in section 6.4. The use of this reservation channel avoids the loss of speech information entirely. The DSR in the uplink gives an increase of capacity of 33 % for *traffic 1*. In the proposed system given in table 2, the maximum number of *traffic 1* users is 3184.

traffic 2:

Traffic 2 requires all the timeslots in a frame as was described in section 6.4. The traffic is modelled as one way traffic, with a continuous transmission character. The user sends its (computer-) data information for a period of a few seconds upto a few minutes or even longer. The slotted ALOHA protocol is used to get access to the common channel to make a reservation. This protocol works in the same way as the protocol that was used for *traffic 1* and the same common channel is used to make the reservation. The use of the DSR in the uplink doubles the capacity. The effects of this technique on the performance of the system are described in chapter 5.

traffic 3:

The multiple access scheme that is proposed for *traffic 3* is slotted ALOHA [9]. *Traffic 3* has a bursty character. The reservation method that is used for *traffic 1* and *traffic 2* is not useful because capacity is wasted when a user doesn't transmit and a channel is reservated for some time. Slotted ALOHA is a good multiple access scheme for *traffic 3* because the delay constraints for *traffic 3* are not too severe and slotted ALOHA saves overhead information and uses the capacity in an efficient way.

6.6 Conclusions and recommendations

In this chapter an initial qualitative description of a possible capacity allocation method for the proposed MC-TDMA/SFH-DSR protocol for UMTS was given. The proposed system seems to be appropriate to integrate the three services, which is a very important requirement for UMTS. To investigate how it performs, a model has to be made for the traffic that arrives. If this model is available, an algorithm must be developed to determine the positions of the capacity boundaries in an optimal way when the traffic intensity of the three services changes according to the traffic model. It was described that only one channel is used as a reservation channel for *traffic 1* and *traffic 2*. More channels can be used of course. Further research has to determine the optimum of the number of common channels for reservation. The method of allocating capacity to the different types of services that is described in this chapter is of course not the only possible solution. In our Telecommunications and Traffic Control Systems Group, work is in progress to find solutions for the integration of the different types of services. It is very interesting to investigate the performance of the model proposed in this chapter and compare it with other proposals using the same traffic model. The protocol proposed in this chapter is relatively simple. Other protocols who provide flexible capacity allocation in a frame may be more complex. Therefore, it should also be compared how practical the obtained solutions are.

7. Comparison of MC-TDMA/SFH-DSR with TDMA-DS/SFH-CDMA

7.1 Introduction

At this moment work is in progress throughout the world to develop radio interface protocols that fulfil the requirements for UMTS. To determine which protocol is the best, the performance of the possible protocols must be compared in a fair way. The candidates can roughly be divided in two types of systems:

- Narrowband systems
- Wideband systems

The description for both narrowband and wideband are given in chapter 3.
In the Telecommunications and Traffic Control Systems Group in Delft where the MC-TDMA/SFH-DSR (which is a narrowband system) has been developed, a wideband (spread spectrum) protocol is also investigated. For the investigation of both protocols, the same system parameters where used and results were generated for a complete mobile communication system. This makes it very interesting to compare the results of the two protocols on spectrum efficiency and capacity. It will become clear that there is a difference in performance between the two systems.

7.2 Description of TDMA-DS/SFH-CDMA

Hybrid TDMA-DS/SFH-CDMA method overview

The principle of hybrid DS/SFH-CDMA scheme is that the data signal at the transmitter is first multiplied by a pseudo-random user specific code sequence. After modulation, the carrier frequency hops randomly from one frequency to another under control of a PN-process. Each user has its own hopping pattern, and the users who have the same hopping pattern have their own specific code sequence so all the users are able to transmit their information simultaneously. At the receiver the original data can be recovered by dehopping the received data signal, and after demodulation, data will be despread by correlating the received spread spectrum signal with the correct code sequence.

It is known from chapter 2 that DS-CDMA is resistant to multipath fading, but sensitive to the near-far effect. However, SFH-CDMA is sensitive to multipath fading and resistant to the near-far effect. Therefore, one of the options to minimize multipath fading and near-far effect is to combine DS with SFH (hybrid DS/SFH-CDMA). Hybrid DS/SFH-CDMA system is a candidate for UMTS because it can sustain up to 2 Mbit/s using appropriate diversity and FEC coding for uplink and downlink.

TDMA is used to be able to integrate speech and data service in one system. The method to integrate speech and data service for the MC-TDMA/SFH-DSR system was given in chapter 6.

Uplink

When considering the uplink from MS transmitter to BS receiver, the signal received by the base station from a mobile close to the base station will be much stronger than the signal received from a mobile located at the cell boundary, due to the path loss. Hence the latter user will be dominated by the close-in mobile (near-far effect). A solution for this problem is power control which attempts to achieve constant received mean power of each user.

Downlink

In a hybrid DS/SFH-CDMA system, the base station transmits signals to all active users (downlink) in its service area, using one antenna. Therefore each mobile receives a composite signal from its base station consisting of the desired signal and interfering signals.

7.3 Results for the Multiple Cell system

In chapter 5 the performance results were given for the MC-TDMA/SFH-DSR system. In this section the results are given of the TDMA-DS/SFH-CDMA multiple cell system. The performance results are given in terms of spectral efficiency and capacity. The multiple cell system that is considered has a frequency reuse structure with cluster size C = 1. This means that the total available bandwidth is available in every cell. Table 7.1 gives an overview of the parameters that were used.

table 7.1 overview of the system parameters that were used

Parameters	Value	Notes
Bandwidth per cell	15 MHz	
Shadowing error deviation	6 dB	log-normal distribution
Data modulation	spread in N chips	N is the spreading sequence (CDMA)
Chip modulation	QPSK	Modulation techniques used
Number of hopping frequencies per cell	8 Data 10 Speech	
Spreading sequence (N) per cell	31 Data 255 Speech	QPSK QPSK

The capacity and spectral efficiency results that were obtained by simulation are given in table 7.2 for the uplink connection.

table 7.2 : Capacity and spectral efficiency of DS/SFH-CDMA for the Uplink

Service $C = 1$	Required E_b/N_0 (dB)	Bitrate [kbit/s]	Modulation	Capacity (η) [users/MHz/cell]	Spectral efficiency [kbit/s/MHz/cell]
Data	10	128	QPSK	1	128
Speech	10	12	QPSK	8.5	102.4

table 7.3 : Capacity and spectral efficiency of MC-TDMA/SFH-DSR for the Uplink

Service $C = 3$	Required E_b/N_0 (dB) for the large signal	Bitrate [kbit/s]	Modulation	Capacity (η) [users/MHz/cell]	Spectral efficiency [kbit/s/MHz/cell]
Data	8.7	128	BPSK	2.59	333
Speech	8.7	12	BPSK	27.66	333

For the uplink, it can be seen that the capacity of the narrowband MC-TDMA/SFH-DSR system is about a factor 3 better than the given spread spectrum system DS/SFH-CDMA.

For the downlink connection the results for DS/SFH-CDMA and MC-TDMA/SFH-DSR are given in table 7.4 and 7.5.

table 7.4: Capacity and Spectral efficiency of DS/SFH-CDMA for the Downlink

Service C = 1	Required E_b/N_0 (dB)	Bitrate [kbit/s]	Modulation	Capacity (η) [users/MHz/cell]	Spectral efficiency [kbit/s/MHz/cell]
Data	10	128	QPSK	1.1	140.8
Speech	10	12	QPSK	11.7	140.4

table 7.5: Capacity and Spectral efficiency of MC-TDMA/SFH-DSR for the Downlink

Service C = 3	Required E_b/N_0 (dB)	Bitrate [kbit/s]	Modulation	Capacity (η) [users/MHz/cell]	Spectral efficiency [kbit/s/MHz/cell]
Data	12.4	128	BPSK	1.30	166
Speech	12.4	12	BPSK	14.83	166

From tables 7.4 and 7.5, we see that MC-TDMA/SFH-DSR system gives better results than the DS/SFH-CDMA system. However the difference is not so big as it was for the uplink. This is because the DSR doesn't enhance the spectral efficiency in the downlink, compared to a conventional receiver. Notice that for MC-TDMA/SFH-DSR a higher value of E_b/N_0 is required to fulfil the QoS requirements.

7.4 Conclusions and recommendations

In the Telecommunications and Traffic Control Systems Group a lot of research has been done to develop spread spectrum multiple access protocols. Studies where done on DS-CDMA, SFH-CDMA and Hybrid DS/SFH-CDMA, and work is still in progress to enhance the performance by implementing new advanced techniques. During the period that the MC-TDMA/SFH-DSR protocol was developed, the DS/SFH-CDMA protocol was also investigated using the techniques currently available to implement. It is interesting to compare the performance of this protocol with the MC-TDMA/SFH-DSR protocol that is investigated in this thesis to find out which protocol is a better candidate for UMTS. To be able to compare the two protocols, the same system parameters were chosen.

From the obtained results we can conclude that for the given system parameters and implemented techniques, the spectral efficiency is better for the narrowband system MC-TDMA/SFH-DSR. For the uplink, the capacity enhancement is large (a factor 3).

In the comparison between the two systems we have to be careful. The following system aspects should also be considered before the final decision can be taken which of the systems is a better candidate for UMTS:

- Compatibility with GSM
- Signalling requirements in the network (size of the overhead information must be taken into account)
- Transmitter, receiver and processing requirements
- Battery aspects: energy consumption of the mobile station
- Electro Magnetic Compatibility aspects
- Handover reliability; different types of handover are used
- Channel efficiency
- Economical efficiency

There may be even more aspects that have to be taken into account. Because it is not clear how both systems perform on the system aspects that are mentioned here it can not be decided at this moment which of the two systems is the best for UMTS.

8. Conclusions and recommendations

In this thesis work, the MC-TDMA/SFH-DSR protocol has been investigated. The protocol performance is measured in terms of Capacity and Spectral Efficiency. Furthermore, a Link Budget evaluation is made to determine the maximum allowable cell size. Results obtained from [3] were used to develop a simulation program. A solution is proposed to integrate three different services using the MC-TDMA/SFH-DSR protocol. The solution is described in a qualitative way. The following conclusions and recommendations are drawn from the previous chapters:

8.1 Conclusions

• First of all it is found from the performance analysis which is described in chapter 5 that the use of the DSR in the uplink of the MC-TDMA/SFH-DSR system enhances the spectral efficiency in the uplink connection with a factor 2, compared to a conventional receiver, requiring a higher signal to noise ratio per bit.

• This candidate for UMTS is very attractive because of its outstanding performance and the fact that the protocol is based on GSM which provides compatibility.

• The Multi Carrier technique gives the possibility to chose a universal channel bitrate for pico- micro- and macro cell and ISI can be kept at a low level.

• The combination of MC, TDMA and SFH gives a maximum flexibility in capacity allocation. The integration of speech- data- and packet service is possible with the protocol.

• Under the given circumstances the protocol that has been investigated gives much better performance results in the uplink compared with the spread spectrum protocol Hybrid TDMA-DS/SFH-CDMA. One of the advantages of the investigated protocol is the fact that the system performance in terms of spectral efficiency is independent of the total available bandwidth. This is an advantage if the protocol is compared with spread spectrum protocols where available bandwidth has much influence on the performance.

8.2 Recommendations

• The protocol was investigated for BPSK modulated signals. Because of the large enhancement of the results of the DSR in the uplink, performance results should also be obtained for the DSR in the uplink using GMSK modulation which is used in GSM.

• Throughout the work, errors were assumed to be random and uncorrelated after interleaving and a coding table was used to determine the performance of the protocol with convolutional coding using the obtained results for the average BER. Coding should be further investigated theoretically as well as with simulations.

• In the DSR, the received small signal is retrieved using the large signal. If errors are first removed from the large signal using coding, and the small signal is retrieved using this "corrected large signal" performance results of the small signal can be improved.

• Throughout the work, SIR = 3 dB is assumed. However, smaller values of SIR, for example SIR = 0 dB may also give good results because of the Rayleigh fading channel which results in instantaneous values of SIR which are not zero. An optimum value should be found for SIR in the given channel.

• Orthogonal hopping patterns are used, allowing consequently two co-channel users simultaneously in the uplink. The performance of the protocol using random hopping patterns should be investigated. The DSR is then used to be able to receive two signals when a "hit" occurs.

• The proposed capacity allocation protocol is attractive because of its simplicity. A traffic model should be made for the traffic, consisting of three different types of traffic. An algorithm should be developed to determine the position of the capacity boundaries between *traffic 1* and *traffic 3*, and between *traffic 2* and *traffic 3*. The performance results of this relatively easy to implement protocol can then be compared with other (often more complex) protocols for integration of different services.

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Appendix: Listing of computer programs

In this appendix the computer programs are given that were used for the simulations in MATLAB. Two programs are listed: For the uplink, with implementation of antenna diversity and for the downlink whithout antenna diversity. The programs calculate the BER as a function of E_b/N_0 for the single cell system and for the multiple cell system. The programs finally plot the BER as a function of E_b/N_0 for three system loads: 40 %, 80 % and 100 %.

PROGRAM 1 MATLAB PROGRAM FOR UPLINK SIMULATIONS

This MATLAB program calculates the BER as a function of E_b/N_0 for a specific SIR and Cluster size for 40 %, 80 % and 100 % network load for multiple cell. The results for single cell are also calculated. The COST 207 macrocell power delay profile is used. The results are for the uplink using antenna diversity as was described in chapter 5.

```
clear;
i = sqrt(-1);
qq = 0;
B = [0 -3.2 -5 -4.5 -3.6 -3.9 -3 -1.2 -5 -3.5];
B = 10.^(B/20);
B1 = B/sqrt(sum((B.^2)'));

Size_of_vectors = 500;

rand('seed',sum(100*clock));
phi1 = exp(2*pi*i*rand([10,Size_of_vectors ]));
phi2 = exp(2*pi*i*rand([10,Size_of_vectors ]));
phi3 = exp(2*pi*i*rand([10,Size_of_vectors ]));
phi4 = exp(2*pi*i*rand([10,Size_of_vectors ]));
phi5 = exp(2*pi*i*rand([10,Size_of_vectors ]));
phi6 = exp(2*pi*i*rand([10,Size_of_vectors ]));

u = [0 4 8 12 16 20];
% we hebben in de uplink anderhalf keer zoveel interferentie als in de downlink als SIR=3dB
% omdat er twee interfereerders zijn per cel waarvan er een de helft van het vermogen van de
% sterkste uitzendt

% cluster size C = 4, beta = 4 (path loss factor)
% 100% load : C/I = 17.46-1.76 =15.70
% 80% load : C/I = 15.7 - 10log10(0.8) =16.67
% 40% load : C/I = 15.7 - 10*log10(0.4)=19.68

% cluster size C = 3
% 100% load : C/I = 14.58 - 1.76 = 12.82
% 80% load : C/I = 12.82 - 10log10(0.8) = 13.79
% 40% load : C/I = 12.82 - 10*log10(0.4)= 16.80

MC = u - 10*log10(1+10.^((u-12.82)/10));
MC2 = u - 10*log10(1+10.^((u-13.79)/10));
MC3 = u - 10*log10(1+10.^((u-16.80)/10));
a = [u MC MC2 MC3];
for U = a
sir_i = 3;
B2 = B1/(sqrt(10^(sir_i/10)));
Peen=[];
```

```
Ptwee=[];
P1=[];
P2=[];
P3=[];
P4=[];
P5=[];
P6=[];
g=[];
e=[];
x=[];
y=[];
ggg=[];
eee=[];
xxx=[];
yyy=[];
G=[];
E=[];
X=[];
Y=[];
GGG=[];
EEE=[];
YYY=[];
XXX=[];
gg=[];
ee=[];
xx=[];
yy=[];
GG=[];
EE=[];
XX=[];
YY=[];

psir1sterk=[];
psir2sterk=[];
psir3sterk=[];

psir1zwak=[];
psir2zwak=[];
psir3zwak=[];

psnrsterk=[];
psnrsterkk=[];
psnrsterkkk=[];

psnrzwak=[];
psnrzwakk=[];
psnrzwakkk=[];

pfassterk=[];
```

```
pfassterkk=[];
pfassterkkk=[];

pfaszwak=[];
pfaszwakk=[];
pfaszwakkk=[];

sfassterk=[];
sfassterkk=[];
sfassterkkk=[];

%eerste antenne

 snr1mati = 20*log10(abs(B1*phi1))+U;
 snr1matc = 20*log10(abs(B2*phi2))+U;
 sir1mati = snr1mati-snr1matc;

%Lev1 betekent A>M en ~Lev1 betekent A<M

 Lev1 = (sir1mati>=0);

 sir1sterk =((Lev1).*sir1mati);
 sisterk =find(sir1sterk>0);
  for i = 1:1:length(sisterk)
   ssir1sterk = sir1sterk(sisterk(i));
   psir1sterk =[psir1sterk ssir1sterk];
  end

 sir1zwak =-((~Lev1).*sir1mati);
 sizwak =find(sir1zwak>0);
  for i = 1:1:length(sizwak);
   ssir1zwak = sir1zwak(sizwak(i));
   psir1zwak = [psir1zwak ssir1zwak];
  end

 snrsterk =((Lev1).*snr1mati);
 snsterk =find(snrsterk~=0);
  for i =1:1:length(snsterk)
   ssnrsterk = snrsterk(snsterk(i));
   psnrsterk = [psnrsterk ssnrsterk];
 end

 snrzwak =((~Lev1).*snr1matc);
 snzwak =find(snrzwak~=0);
  for i = 1:1:length(snzwak)
   ssnrzwak = snrzwak(snzwak(i));
   psnrzwak = [psnrzwak ssnrzwak];
  end
 fasemat1 = angle(B1*phi1);
```

```
fassterk = (Lev1.*fasemat1);
fsterk = find(fassterk~=0);
 for i =1:1:length(fsterk)
  sfassterk =fassterk(fsterk(i));
  pfassterk =[pfassterk sfassterk];
 end
fasemat2 = angle(B2*phi2);
faszwak = ((~Lev1).*fasemat2);
fzwak = find(faszwak~=0);
 for i =1:1:length(fzwak)
  sfaszwak = faszwak(fzwak(i));
  pfaszwak = [pfaszwak sfaszwak];
 end

%tweede antenne

 snr2mati = 20*log10(abs(B1*phi3))+U;
 snr2matc = 20*log10(abs(B2*phi4))+U;
 sir2mati = snr2mati-snr2matc;

%Lev1 betekent A>M en ~Lev1 betekent A<M

 Lev1 = (sir2mati>=0);

 sir2sterk =((Lev1).*sir2mati);
 sisterk =find(sir2sterk>0);
  for i = 1:1:length(sisterk)
   ssir2sterk = sir2sterk(sisterk(i));
   psir2sterk =[psir2sterk ssir2sterk];
  end

 sir2zwak =-((~Lev1).*sir2mati);
 sizwak =find(sir2zwak>0);
  for i = 1:1:length(sizwak);
   ssir2zwak = sir2zwak(sizwak(i));
   psir2zwak =[psir2zwak ssir2zwak];
  end

 snrsterkk =((Lev1).*snr2mati);
 snsterk =find(snrsterkk~=0);
  for i =1:1:length(snsterk)
   ssnrsterkk = snrsterkk(snsterk(i));
   psnrsterkk =[psnrsterkk ssnrsterkk];
  end

 snrzwakk =((~Lev1).*snr2matc);
 snzwak =find(snrzwakk~=0);
  for i = 1:1:length(snzwak)
   ssnrzwakk = snrzwakk(snzwak(i));
```

```
  psnrzwakk =[psnrzwakk ssnrzwakk];
 end
fasemat3 = angle(B1*phi3);
fassterkk = (Lev1.*fasemat3);
fsterk = find(fassterkk~=0);
 for i =1:1:length(fsterk)
  sfassterkk =fassterkk(fsterk(i));
  pfassterkk =[pfassterkk sfassterkk];
 end
fasemat4 = angle(B2*phi4);
faszwakk  = ((~Lev1).*fasemat4);
fzwak  = find(faszwakk~=0);
 for i =1:1:length(fzwak)
  sfaszwakk = faszwakk(fzwak(i));
  pfaszwakk =[pfaszwakk sfaszwakk];
end

%derde antenne

 snr3mati = 20*log10(abs(B1*phi5))+U;
 snr3matc = 20*log10(abs(B2*phi6))+U;
 sir3mati = snr3mati-snr3matc;
%Lev1 betekent A>M en ~Lev1 betekent A<M
 Lev1 = (sir3mati>=0);

 sir3sterk =((Lev1).*sir3mati);
 sisterk =find(sir3sterk>0);
  for i = 1:1:length(sisterk)
   ssir3sterk = sir3sterk(sisterk(i));
   psir3sterk =[psir3sterk ssir3sterk];
  end

 sir3zwak  =-((~Lev1).*sir3mati);
 sizwak  =find(sir3zwak>0);
  for i = 1:1:length(sizwak);
   ssir3zwak = sir3zwak(sizwak(i));
   psir3zwak = [psir3zwak ssir3zwak];
  end

 snrsterkkk =((Lev1).*snr3mati);
 snsterk =find(snrsterkkk~=0);
  for i =1:1:length(snsterk)
   ssnrsterkkk = snrsterkkk(snsterk(i));
   psnrsterkkk = [psnrsterkkk ssnrsterkkk];
 end

 snrzwakkk  =((~Lev1).*snr3matc);
 snzwak   =find(snrzwakkk~=0);
  for i = 1:1:length(snzwak)
```

```
  ssnrzwakkk = snrzwakkk(snzwak(i));
  psnrzwakkk = [psnrzwakkk ssnrzwakkk];
 end
 fasemat5 = angle(B1*phi5);
 fassterkkk = (Lev1.*fasemat5);
 fsterk = find(fassterkkk~=0);
 for i =1:1:length(fsterk)
  sfassterkkk =fassterkkk(fsterk(i));
  pfassterkkk =[pfassterkkk sfassterkkk];
 end
 fasemat6 = angle(B2*phi6);
 faszwakkk = ((~Lev1).*fasemat6);
 fzwak = find(faszwakkk~=0);
 for i =1:1:length(fzwak)
  sfaszwakkk = faszwakkk(fzwak(i));
  pfaszwakkk = [pfaszwakkk sfaszwakkk];
 end

psnrsterk = 10.^((psnrsterk/10)');
psir1sterk = 10.^((psir1sterk/10)');
pfassterk = pfassterk';
[M,N] = size (psnrsterk);
[M,N] = size (psir1sterk);
[M,N] = size (pfassterk);
I = 1;
for I=1:1:M
P = psnrsterk(I,:);
Q = psir1sterk(I,:);
T = pfassterk(I,:);
snr_i=P;
sir_i=Q;
phi=T;
delta=1;
eta = 0;
Qmat=[0:0.05:10];
Qmat=log(q_func(Qmat));
Pei_tot = 0;
Pec_tot = 0;
for k = 1:1:6
delta = (k-1)*0.1;
[Pei,Pei_mat]=pberdom(snr_i,sir_i,phi,delta,Qmat);
beta = 1-(1-2*Pei)*(1-eta);
[Pec] = pbermin(Pei_mat,snr_i/sir_i,1/sir_i,phi,delta,beta,Qmat);
Pei_tot = Pei_tot+Pei/6;
Pec_tot = Pec_tot+Pec/6;
end

% bitfoutenkans voor het sterkst verzonden signaal voor de Dual Signal Receiver
G = [G snr_i];
```

```
E = [E sir_i];
X = [X Pei_tot];
Y = [Y Pec_tot];

% voor de conventionele ontvanger is de bitfoutenkans 0.5 indien het
% sterkst verzonden signaal als zwakst wordt onvangen, dus Y_C wordt:

Y_C = [Y_C 0.5];
end

psnrzwak = 10.^((psnrzwak/10)');
psir1zwak = 10.^((psir1zwak/10)');
pfaszwak = pfaszwak';
[M,N] = size (psnrzwak);
[M,N] = size (psir1zwak);
[M,N] = size (pfaszwak);
I = 1;
for I=1:1:M
P = psnrzwak(I,:);
Q = psir1zwak(I,:);
T = pfaszwak(I,:);
snr_i=P;
sir_i=Q;
phi=T;
delta=1;
eta = 0;
Qmat=[0:0.05:10];
Qmat=log(q_func(Qmat));
Pei_tot = 0;
Pec_tot = 0;
for k = 1:1:6
delta = (k-1)*0.1;
[Pei,Pei_mat]=pberdom(snr_i,sir_i,phi,delta,Qmat);
beta = 1-(1-2*Pei)*(1-eta);
[Pec] = pbermin(Pei_mat,snr_i/sir_i,1/sir_i,phi,delta,beta,Qmat);
Pei_tot = Pei_tot+Pei/6;
Pec_tot = Pec_tot+Pec/6;
end

% bitfoutenkans voor het zwakst verzonden signaal voor de Dual Signal Receiver
GG = [GG snr_i];
EE = [EE sir_i];
XX = [XX Pei_tot];
YY = [YY Pec_tot];

% voor de conventionele ontvanger is de bitfoutenkans 0.5 indien het
% zwakst verzonden signaal als zwakst wordt onvangen, dus YY_C wordt:
YY_C = [YY_C 0.5];
end
```

```
psnrsterkk = 10.^((psnrsterkk/10)');
psir2sterk = 10.^((psir2sterk/10)');
pfassterkk = pfassterkk';
[M,N] = size (psnrsterkk);
[M,N] = size (psir2sterk);
[M,N] = size (pfassterkk);
I = 1;
for I=1:1:M
P = psnrsterkk(I,:);
Q = psir2sterk(I,:);
T = pfassterkk(I,:);
snr_i=P;
sir_i=Q;
phi=T;
delta=1;
eta = 0;
Qmat=[0:0.05:10];
Qmat=log(q_func(Qmat));
Pei_tot = 0;
Pec_tot = 0;
for k = 1:1:6
delta = (k-1)*0.1;
[Pei,Pei_mat]=pberdom(snr_i,sir_i,phi,delta,Qmat);
beta = 1-(1-2*Pei)*(1-eta);
[Pec] = pbermin(Pei_mat,snr_i/sir_i,1/sir_i,phi,delta,beta,Qmat);
Pei_tot = Pei_tot+Pei/6;
Pec_tot = Pec_tot+Pec/6;
end
g = [g snr_i];
e = [e sir_i];

% bitfoutenkans voor het sterkst verzonden signaal voor de Dual Signal Receiver
x = [x Pei_tot];
y = [y Pec_tot];

% voor de conventionele ontvanger is de bitfoutenkans 0.5 indien het
% sterkst verzonden signaal als zwakst wordt onvangen, dus y_C wordt:
y_C = [y_C 0.5];

end

psnrzwakk = 10.^((psnrzwakk/10)');
psir2zwak = 10.^((psir2zwak/10)');
pfaszwakk = pfaszwakk';
[M,N] = size (psnrzwakk);
[M,N] = size (psir2zwak);
[M,N] = size (pfaszwakk);
I = 1;
```

```
for I=1:1:M
P = psnrzwakk(I,:);
Q = psir2zwak(I,:);
T = pfaszwakk(I,:);
snr_i=P;
sir_i=Q;
phi=T;
delta=1;
eta = 0;
Qmat=[0:0.05:10];
Qmat=log(q_func(Qmat));
Pei_tot = 0;
Pec_tot = 0;
for k = 1:1:6
delta = (k-1)*0.1;
[Pei,Pei_mat]=pberdom(snr_i,sir_i,phi,delta,Qmat);
beta = 1-(1-2*Pei)*(1-eta);
[Pec] = pbermin(Pei_mat,snr_i/sir_i,1/sir_i,phi,delta,beta,Qmat);
Pei_tot = Pei_tot+Pei/6;
Pec_tot = Pec_tot+Pec/6;
end
gg = [gg snr_i];
ee = [ee sir_i];
xx = [xx Pei_tot];
yy = [yy Pec_tot];
end


psnrsterkkk = 10.^((psnrsterkkk/10)');
psir3sterk = 10.^((psir3sterk/10)');
pfassterkkk = pfassterkkk';
[M,N] = size (psnrsterkkk);
[M,N] = size (psir3sterk);
[M,N] = size (pfassterkkk);
I = 1;
for I=1:1:M
P = psnrsterkkk(I,:);
Q = psir3sterk(I,:);
T = pfassterkkk(I,:);
snr_i=P;
sir_i=Q;
phi=T;
delta=1;
eta = 0;
Qmat=[0:0.05:10];
Qmat=log(q_func(Qmat));
Pei_tot = 0;
Pec_tot = 0;
for k = 1:1:6
```

```
delta = (k-1)*0.1;
[Pei,Pei_mat]=pberdom(snr_i,sir_i,phi,delta,Qmat);
beta = 1-(1-2*Pei)*(1-eta);
[Pec] = pbermin(Pei_mat,snr_i/sir_i,1/sir_i,phi,delta,beta,Qmat);
Pei_tot = Pei_tot+Pei/6;
Pec_tot = Pec_tot+Pec/6;
end
GGG = [GGG snr_i];
EEE = [EEE sir_i];
XXX = [XXX Pei_tot];
YYY = [YYY Pec_tot];
end

psnrzwakkk = 10.^((psnrzwakkk/10)');
psir3zwak = 10.^((psir3zwak/10)');
pfaszwakkk = pfaszwakkk';
[M,N] = size (psnrzwakkk);
[M,N] = size (psir3zwak);
[M,N] = size (pfaszwakkk);
I = 1;
for I=1:1:M
P = psnrzwakkk(I,:);
Q = psir3zwak(I,:);
T = pfaszwakkk(I,:);
snr_i=P;
sir_i=Q;
phi=T;
delta=1;
eta = 0;
Qmat=[0:0.05:10];
Qmat=log(q_func(Qmat));
Pei_tot = 0;
Pec_tot = 0;
for k = 1:1:6
delta = (k-1)*0.1;
[Pei,Pei_mat]=pberdom(snr_i,sir_i,phi,delta,Qmat);
beta = 1-(1-2*Pei)*(1-eta);
[Pec] = pbermin(Pei_mat,snr_i/sir_i,1/sir_i,phi,delta,beta,Qmat);
Pei_tot = Pei_tot+Pei/6;
Pec_tot = Pec_tot+Pec/6;
end
ggg = [ggg snr_i];
eee = [eee sir_i];
xxx = [xxx Pei_tot];
yyy = [yyy Pec_tot];
end

% P1 en P2 zijn de verzamelingen van bitfouten kansen voor respectievelijk
% het sterkst verzonden signaal en het zwakst verzonden signaal
```

```
% voor antenne 1

P1 = [X YY];
P2 = [XX Y];

% evenzo voor antenne 2

P3 = [x yy];
P4 = [xx y];

% evenzo voor antenne 3

P5 = [XXX yyy];
P6 = [xxx YYY];

% We gaan nu de BER berekenen indien we indien twee of meer (drie)
% antennes hetzelfde bit als uitkomst geven die waarde nemen.

PFOUT = P1.*P3.*P5 + P1.*P3.*(1-P5) + P1.*P5.*(1-P3) + P3.*P5.*(1-P1);
Pfout = P2.*P4.*P6 + P2.*P4.*(1-P6) + P2.*P6.*(1-P4) + P4.*P6.*(1-P2);

% sterkst verzonden signaal zonder codering
bers = sum(PFOUT)/Size_of_vectors;

% zwakst verzonden signaal zonder codering
berz = sum(Pfout)/Size_of_vectors;

BERS = [BERS bers];
BERZ = [BERZ berz];

qq = qq+1
end

% single cell is de eerste 6 waarden
MBERS = BERS(1:6);
MBERZ = BERZ(1:6);

% Multiple cell met 100 % load is de tweede 6 waarden
MCBERS_100 = BERS(7:12);
MCBERZ_100 = BERZ(7:12);

% Multiple cell met 80 % load is de derde 6 waarden
MCBERS_80 = BERS(13:18);
MCBERZ_80 = BERZ(13:18);

% Multiple cell met 40 % load is de vierde 6 waarden
MCBERS_40 = BERS(19:24);
MCBERZ_40 = BERZ(19:24);
```

```
F = [0:4:20];

% De gemiddelde BER wordt uitgedrukt voor het sterkst verzonden signaal en het
% zwakst verzonden signaal als functie van Eb/No. Er is gebruik gemaakt van drie
% antennes voor de uplink. Voor de downlink is geen antenne diversity mogelijk.

subplot(2,2,1);
semilogy(F,MBERS,F,MBERZ);
title('single cell');

grid;

subplot(2,2,2);
semilogy(F,MCBERS_100,F,MCBERZ_100);
title('100 % load)');
grid;

subplot(2,2,3);
semilogy(F,MCBERS_80,F,MCBERZ_80);
title('80 % load)');
grid;
subplot(2,2,4);
semilogy(F,MCBERS_40,F,MCBERZ_40);
title('40 % load)');
grid;
MBERS = MBERS';
MBERZ = MBERZ';
MCBERS_100 = MCBERS_100';
MCBERZ_100 = MCBERZ_100';
MCBERS_80  = MCBERS_80';
MCBERZ_80  = MCBERZ_80';
MCBERS_40  = MCBERS_40';
MCBERZ_40  = MCBERZ_40';
save upl_c4.mat MBERS MBERZ MCBERS_100 MCBERZ_100 MCBERS_80 MCBERZ_80 MCBERS_40 MCBERZ_40
figure(1)
```

PROGRAM 2 MATLAB PROGRAM FOR DOWNLINK SIMULATIONS

This MATLAB program calculates the BER as a function of E_b/N_0 for a specific SIR and Cluster size for 40 %, 80 % and 100 % network load for multiple cell. The results for single cell are also calculated. The COST 207 macrocell power delay profile is used. The results are for the downlink without antenna diversity.

```
clear;
number = 2000;
i = sqrt(-1);
load bbmati2.dat;
load bbmatc2.dat;
LEV1 = (bbmati2==0);
LEV2 = (bbmatc2==0);
bbmati2 = bbmati2 + 1e-25*LEV1;
bbmatc2 = bbmatc2 + 1e-25*LEV1;
Pi = log(bbmati2);
Pc = log(bbmatc2);
Pi = reshape(Pi',1,12231);
Pc = reshape(Pc',1,12231);
bbmati2 = [];
bbmatc2 = [];
sir_i = 1000;
B = [0 -3.2 -5 -4.5 -3.6 -3.9 -3 -1.2 -5 -3.5];
B = 10.^(B/20);
B1 = B/sqrt(sum((B.^2)'));
B2 = B1/(sqrt(10^(sir_i/10)));
rand('seed',sum(100*clock));
phi1 = exp(2*pi*i*rand([10,number]));
phi2 = exp(2*pi*i*rand([10,number]));

u = [0 4 8 12 16 20];
% cluster size C = 4
% 100% load : C/I = 17.46
% 80% load : C/I = 17.46 - 10log10(0.8) = 18.43
% 40% load : C/I = 17.46 - 10*log10(0.4)= 21.44

% cluster size C = 3
% 100% load : C/I = 14.58
% 80% load : C/I = 14.58 - 10log10(0.8) = 15.55
% 40% load : C/I = 14.58 - 10*log10(0.4)= 18.56

MC = u - 10*log10(1+10.^((u-14.58)/10));
MC2 = u - 10*log10(1+10.^((u-15.55)/10));
MC3 = u - 10*log10(1+10.^((u-21.44)/10));
```

```
a = [u MC MC2 MC3];
for U = a

 Pec1 = [];
 Pec2 = [];
 snrmati = 20*log10(abs(B1*phi1))+U;
 snrmatc = 20*log10(abs(B2*phi2))+U;
 sirmati = snrmati-snrmatc;
 Lev1 = (sirmati>=0);
 sirmat =(Lev1.*sirmati)-((~Lev1).*sirmati);
 snrmat =(Lev1.*snrmati)+(~Lev1).*snrmatc;
 [Pbi,Pbc] = findpr(sirmat,snrmat,Pi,Pc);
 P1=((Lev1.*Pbi)+((~Lev1).*Pbc));
 P2=((Lev1.*Pbi)+((~Lev1)*0.5));
 Pav1 = sum(P1')/number;
 Pav2 = sum(P2')/number;

Ps = [Ps Pav1];

 end

% single cell is de eerste 6 waarden
% DSR
MBER = Ps(1:6);

% Multiple cell met 100 % load is de tweede 6 waarden
% DSR
MCBER_100 = Ps(7:12);

% Multiple cell met 80 % load is de derde 6 waarden
% DSR
MCBER_80 = Ps(13:18);

% Multiple cell met 40 % load is de vierde 6 waarden
% DSR
MCBER_40 = Ps(19:24);

F = [0:4:20];

subplot(2,2,1);
semilogy(F,MBER);
title('Single Cell');
grid;
subplot(2,2,2);
semilogy(F,MCBER_100);
title('100% load');
grid;
subplot(2,2,3);
semilogy(F,MCBER_80);
```

```
title('80% load');
grid;
subplot(2,2,4);
semilogy(F,MCBER_40);
title('40% load');
grid;
figure(1);
BER = MBER';
MCBER_100 = MCBER_100';
MCBER_80 = MCBER_80';
MCBER_40 = MCBER_40';
```

Performance analysis of a *Dual Signal Receiver - TDMA/SFH* system in a Macro-Cellular Environment

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ABSTRACT:

This paper presents evaluation results of a Time Division Multiple Access (TDMA) Slow Frequency Hopping (SFH) system, using a Dual-Signal Receiver (DSR) [1] in a Rayleigh fading channel. The DSR has the property of simultaneous reception of two narrowband Binary Phase Shift Keying (BPSK) co-channel signals. Both single and multiple-cellular systems are evaluated. The macro-cellular channel model used in this paper is the COST 207 channel model. The performance is evaluated in terms of average BER, capacity and spectral efficiency. The DSR is used in the uplink connection to allow two users simultaneously in a frequency band. This application of the DSR shows to enhance the system capacity significantly compared to a conventional receiver. Antenna diversity is used in the uplink connection to enhance performance results. Convolutional coding is used for uplink and for downlink.

I. Introduction

In the last decades, mobile communication has become a research subject of increasing importance. Because of the limited available bandwidth for radio communication services, the bandwidth has to be used as efficiently as possible. Protocols, modulation types and coding schemes are developed to achieve an optimum use of the available frequencies in a multi-user environment, where different users simultaneously operate without mutually destroying their information. In this paper a system based on Time Division Multiple Access (TDMA) and Slow Frequency Hopping (SFH) is proposed using a Dual Signal Receiver (DSR) concept [1]. In the basic concept TDMA/SFH-DSR, the unit of transmission is a series of modulated bits which is called a burst. Bursts have a finite duration, occupy a finite part of the radio spectrum and are sent in time and frequency windows called slots. Orthogonal hopping patterns are used so the number of hopping patterns per timeslot is equal to the number of frequency slots. The DSR can receive two narrowband Binary Phase Shift Keying (BPSK) signals of slightly different strength. By exploiting the capture effect, the large signal is demodulated and after remodulation it is subtracted from the composite received signal. After this subtraction, the smaller signal remains with some residual interference, and can be demodulated. A simplified model for the DSR is given in figure 1.

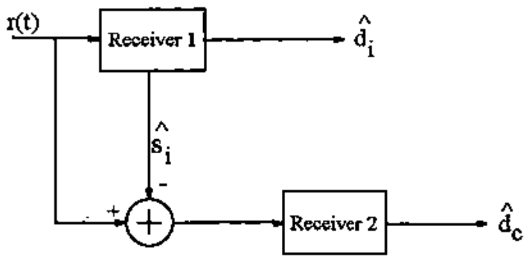

figure 1. Block diagram of the Dual Signal Receiver

In figure 1, r(t) is the composite received signal, $\hat{d}_i$ is the estimated data of the large signal, $\hat{s}_i$ is the remodulation of the estimated large data and $\hat{d}_c$ is the estimated data of the small signal. This property of the DSR is used to enhance the system capacity by allowing two co-channel users simultaneously in a frequency band in the uplink connection, using an average Signal to Interference Ratio (SIR) of 3 dB between the two signals. This means that one hopping pattern is allocated to two co-channel users simultaneously in the uplink. This gives a more efficient use of the available bandwidth than a conventional BPSK receiver. To generate BER results as a function of E_b/N_0, the Codit COST 207 outdoor macro-cell channel model [2] is used throughout this paper. This model is given in table 1 and shows to be a Rayleigh fading channel.

table 1 : Codit COST 207 macro-cell channel model

i	Relative delay τ_i (ns)	Relative power α^2_i (dB)
1	100	-3.2
2	200	-5.0
3	500	-4.5
4	600	-3.6
5	850	-3.9
6	900	0.0
7	1050	-3.0
8	1350	-1.2
9	1450	-5.0
10	1500	-3.5

Throughout the performance analysis only narrowband channels are considered and the paths are assumed to add non-coherently. This non-coherent addition corresponds to a phasor addition. The equivalent low pass channel that contains the discrete multipaths components is described by the time variant impulse response of the channel, derived from [3]:

$$c(t) = \sum_{i=1}^{n} \alpha_i \cdot \delta(t-\tau_i) \cdot e^{j\theta_i(t)} \qquad (1)$$

The phases θ_i of the signal are time variant due to the time variations of the medium and varying antenna positions. The values of θ_i are chosen randomly between 0 and 2π in time. The COST 207 macro-cell channel model is used to simulate the performance of the DSR. The DSR is useful in the mobile stations to suppress co-channel interference. The desired signal can be retrieved when a strong interferer is present. The advantage of the DSR compared with a conventional receiver becomes clear from figure 2, where the Outage Probability is given as a function of the average Signal to Interference Ratio (SIR). The outage probability is defined as Pr{BER > 1e-3}. The value 1e-3 is used as the Quality of Service requirement throughout this paper.

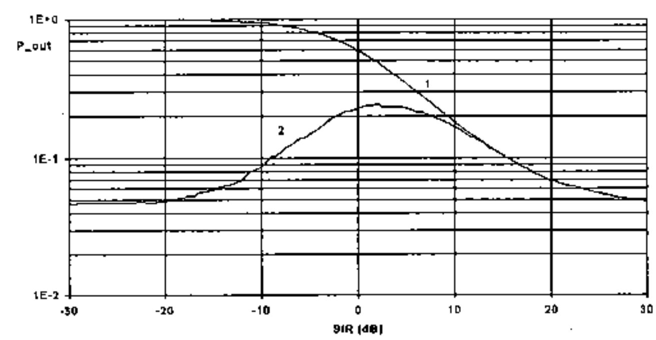

figure 2. Outage Probability for the COST 207 channel as a function of SIR for E_b/N_0 = 20 dB of the desired signal, with:
1. *A conventional receiver*
2. *The Dual Signal Receiver*

From figure 2 it can be seen that the desired signal can be received when SIR < 0. In this case a conventional receiver would be captured by the interfering signal. For the downlink, the DSR is implemented but with the method that is used to model inter-cell co-channel interference, the DSR has the same performance as a conventional receiver. This makes it possible to compare the system using the DSR concept, with the system using a conventional receiver by comparing uplink performance results with downlink performance results. In this paper, results are given using a system approach. A single cell, as well as a multiple cell system is investigated. Antenna diversity is used in the uplink to enhance system performance. Results are given in terms of capacity, spectral efficiency and allowable cell size. Finally a comparison is made between the performance of the DSR multiple cell system and the performance of the same system with a conventional BPSK receiver and conclusions are drawn from the obtained results.

II. BER results for single cell

Results are obtained by simulations using the COST 207 macro-cell channel model. In this section the single cell system is evaluated for uplink and downlink connection.

In figure 3 the average BER is given as a function of E_b/N_0 for the downlink connection.

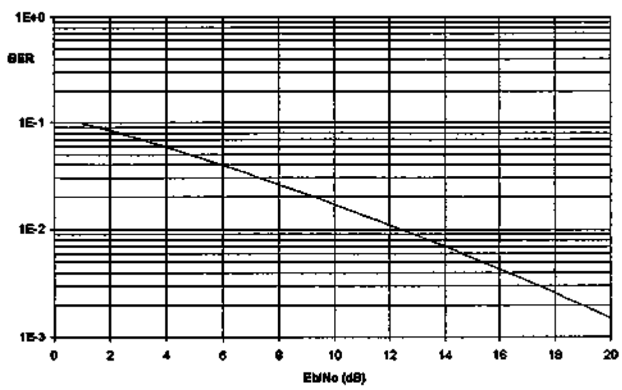

figure 3: Average BER as a function of E_b/N_0 for the COST 207 macro-cell channel for the downlink

The results in figure 3 are generated without error correcting coding. If convolutional coding with coding rate R = 1/2 and constraint length 8 is implemented, it is found from [4] that a BER before decoding of 3e-2 is required to result in a BER after decoding of 1e-3 which is the QoS requirement used throughout the performance evaluation. A value of E_c/N_0 = 7.8 dB is required to fulfil the QoS requirement.

In the uplink connection, two co-channel signals are present. The performance of the system is evaluated using an average received SIR between the both signals of 3 dB. Figure 4 gives the BER performance for the uplink connection.

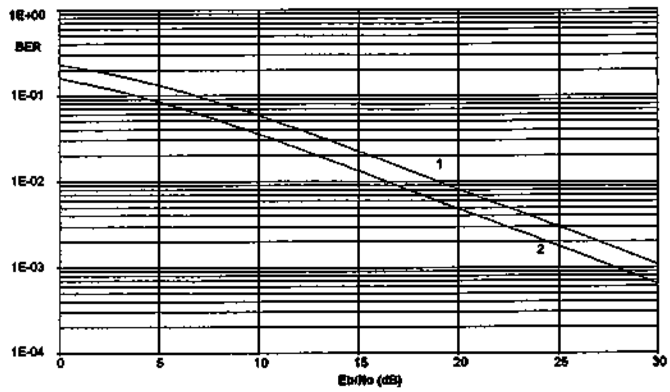

figure 4: Average BER as a function of E_b/N_0 for the COST 207 macro-cell channel for the uplink, SIR = 3dB, with:
1.BER for the small signal
2. BER for the large signal

The performance of the small signal can be assumed to be the same as the performance of the large signal (when coding is used in the large signal before the small signal is retrieved), so an E_c/N_0 = 11 dB is required to fulfil the QoS requirement when the proposed convolutional coding is used. This is 3 dB higher than the required E_c/N_0 for the downlink. The performance for the uplink connection can further be improved by using antenna diversity at the Base Station. The antenna diversity technique that was used is antenna diversity with 3 antennas using majority voting. The principle of this technique is shown in figure 4.

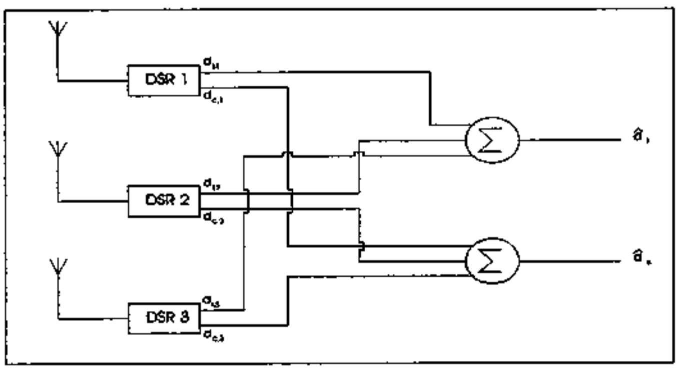

figure 5: Antenna diversity concept with 3 receiving antennas using majority voting

The three antennas are assumed to receive fully uncorrelated signals from a mobile station, which is achieved when the distance between the receiving antennas is large enough. Every antenna receives 2 signals both having a certain P_{el} and P_{ec} after demodulation. P_{el} and P_{ec} are respectively the probability of a bit error for the large and for the small signal. The received bits from the three antennas are compared for the large and the small signal. To define the resulting estimated bit for the large and for the small signal, majority voting is used. This means that the summation of the three bits determines the estimated value of the received data bit:

$$\hat{d} = \begin{cases} +1 & for \ \sum_{n=1}^{3} d_n > 0 \\ -1 & for \ \sum_{n=1}^{3} d_n < 0 \end{cases} \qquad (2)$$

If P_1, P_2, and P_3 are respectively the BER for a bit from antenna 1, 2 and 3, the probability that the data bit that was sent by a mobile user is received incorrectly is:

$$P_{biterror} = P_1 \cdot P_2 \cdot P_3 + (1-P_1) \cdot P_2 \cdot P_3 + P_1 \cdot (1-P_2) \cdot P_3 + P_1 \cdot P_2 \cdot (1-P_3) \qquad (3)$$

This formula is used for the small and for the large signal. Figure 5 gives the results for the uplink with antenna diversity.

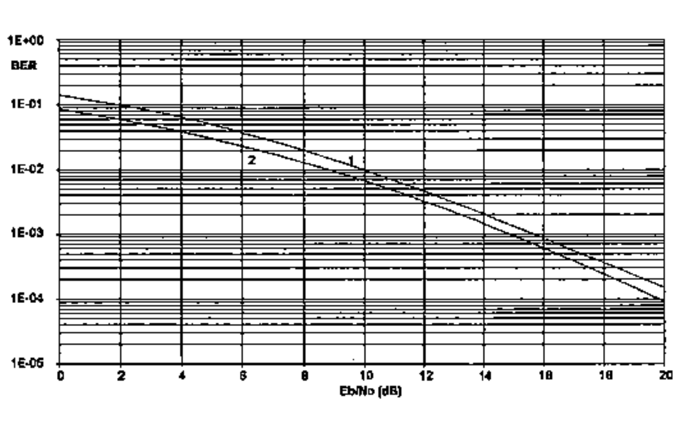

figure 5: Average BER as a function of E_b/N_0 for the COST 207 Macro-cell channel for the uplink with antenna diversity and SIR = 3dB, with:
1. BER for the small signal
2. BER for the large signal

Using convolutional coding R = 1/2 and constraint length 8, a E_b/N_0 = 4.9 dB is required to reach BER = 3e-2 before decoding and thus 1e-3 after decoding.

III. BER results for multiple cell

The multiple cell performance differs from the single cell performance because of the inter-cell interference that is present. A multiple cell system with cluster size C = 3 is considered. The reuse distance *d*, which is defined as the distance between two Base Stations using the same frequency band, is determined by [5]:

$$d = \sqrt{3C} \cdot R \qquad (4)$$

This parameter is used to determine the total inter-cell interference. In [6] a model is given for the inter-cell interference in case of power control. This is necessary in the system investigated here to obtain an average SIR of 3 dB between the large and the small signal in an uplink frequency band. The interference from the six nearest interfering cells is calculated. This set of cells is called the first tier. This value is the average value for all possible terminal positions, assuming that the probability of the positions in a cell is uniformly distributed. The interference from the second tier can be neglected for the used value for the path loss factor $\beta = 4$. In the uplink two users share one frequency band so the total inter-cell co-channel interference from the first tier consists of the interference of 12 users. In the downlink the total co-channel interference from the first tier consists of the interference of 6 users. The total average interference I from the first tier for 100 % system load is then:

Downlink:

$$I = 6 \cdot P_{RB,1^{st}\,tier} \qquad (5)$$

with

$$P_{RB}(d) = 2 \cdot S_p \left[2\left(\frac{d}{R}\right)^2 \ln\left(\frac{\left(\frac{d}{R}\right)^2}{\left(\frac{d}{R}\right)^2 - 1}\right) - \frac{4\left(\frac{d}{R}\right)^4 - 6\left(\frac{d}{R}\right)^2 + 1}{2\left(\left(\frac{d}{R}\right)^2 - 1\right)^2} \right] \qquad (6)$$

$P_{RB,1^{st}\,tier}$ is the average interference power from one neighbouring cell in the first tier. This formula is a closed expression for β=4. The open form is given in [6]. The average interference from one cell is calculated from the formula for $P_{RB}(d)$The received desired signal is S_p so the total average C/I for downlink becomes:

$$\frac{C}{I} = \frac{S_p}{6 \cdot P_{RB,1^{st}\, tier}} \tag{7}$$

For uplink 12 interferers from the first tier are present because two users per uplink channel are present. Because of the SIR of 3 dB, the total interfering power for uplink is 1.5 times the interfering power for downlink:

Uplink:

$$I = 9 \cdot P_{RB,1^{st}\, tier} \tag{8}$$

so the total average C/I for uplink becomes:

$$\frac{C}{I} = \frac{S_p}{9 \cdot P_{RB,1^{st}\, tier}} \tag{9}$$

P_{RB} is the average interference power from one neighbour cell.. The total interference that we calculated is assumed to be Gaussian distributed. From [7] it is found that the summation of a number of interferers which are Rayleigh distributed can be modelled by a Gaussian distribution if C/I is large enough and the number of interferers is large enough. Using this model for the co-channel interference, the interference adds to the AWGN and the BER performance for multiple cell 100% system load and for single cell for the downlink is given in figure 6.

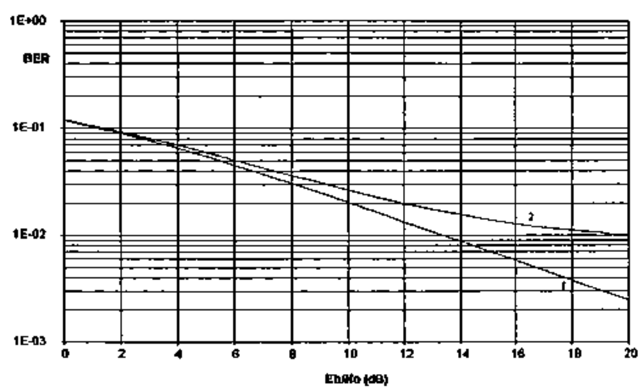

figure 6: Average BER for the downlink for the COST 207 macro-cell channel, with:
1. Single cell
2. Multiple cell, C = 3 and 100 % system load

For multiple cell with cluster size C = 3 and 100 % system load, the required E_c/N_0 = 9.4 dB to obtain BER = 1e-3 after decoding. For the uplink, the BER performance is given in figure 7.

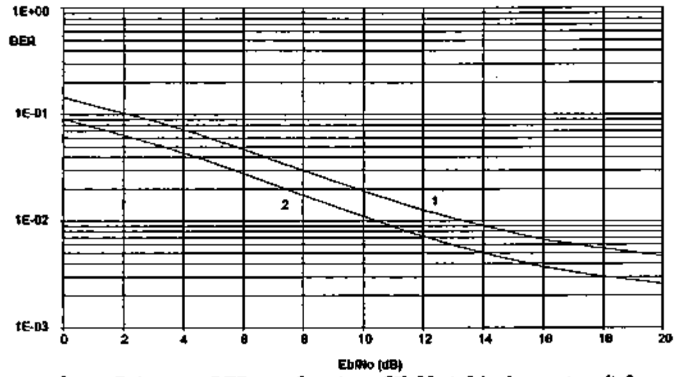

figure 7: Average BER as a function of E_b/N_0 (of the large signal) for multiple cell uplink for the COST 207 macro-cell channel model. SIR = 3 dB, C = 3 and network load = 100 %, with:
1. BER for the small signal
2. BER for the large signal

From both figure 6 and 7 it can be seen that the system is interference limited. This is expressed for large values of E_b/N_0. From figure 7 a required E_c/N_0 = 5.7 is found to fulfil the QoS requirement.

IV. Capacity, spectral efficiency and coverage analysis

In II and III the required values for E_c/N_0 are found to fulfil the QoS requirement. These values are used in this section for the coverage analysis. First capacity and spectral efficiency are defined and values are given for single cell and multiple cell.

$$\text{Capacity}\,(\eta) = \frac{\text{Traffic load}\,(\lambda)}{\text{Bandwidth}\,(W)} \left[\frac{\text{users}}{\text{cel} \cdot \text{MHz}}\right]$$

$$\text{Spectral efficiency}\,(\upsilon) = \frac{\text{Traffic load}\,(\lambda) \cdot \text{Average user bitrate}\,(R)}{\text{Bandwidth}\,(W)} \left[\frac{\text{kbit/s}}{\text{cell} \cdot \text{MHz}}\right]$$

Values for the capacity and spectral efficiency are given in table 2 for speech service of 12 kbit/s and data service of 128 kbit/s for multiple cell with cluster size C = 3.

table 2 : Capacity and spectral efficiency for multiple cell with cluster size C = 3

Service C=3	Required E_b/N_0 (dB) for the large signal	Bitrate [kbit/s]	Capacity (η) [users/MHz/cell]	Spectral efficiency [kbit/s/MHz/cell]
Data (uplink)	5.7	128	2.59	333
Speech (uplink)	5.7	12	27.66	333
Data (downlink)	9.4	128	1.30	166
Speech (downlink)	9.4	12	14.83	166

Using the values of table 2, the maximum allowable cell size can be evaluated. To do this, a link budget evaluation is made. In this analysis, aspects like handover gain and power control margin are considered. The following expression is used for this evaluation [8, p.440]:

$$\frac{C}{N} = \frac{P_{EIRP} G_{FS} G_{AR}}{k T_{syst} B} \quad (10)$$

In decibel units, E_b/N_0 received at the detector input in the receiver is related to the link parameters by:

$$\left(\frac{E_b}{N_0}\right)_{dB} = (P_{EIRP})_{dBW} - (L_{FS})_{dB} + (G_{AR})_{dB} + 204 - R_{dB} - NF_{dB} + G_{HO} - G_{PC} \quad (11)$$

In formula 10, P_{eirp} is the effective isotropic radiated power, NF is the noise factor of the receiver, L_{FS} is the path loss, G_{AR} is the antenna gain of the receiver, G_{HO} is the handover gain and G_{PC} is the power control margin.

The handover gain G_{HO} is an indication of the gain a mobile station would have when at the border of a cell it would choose the the base station providing the best link, instead of the physically closest. For the power control margin, a value of 3 dB is taken because of the imperfectness of the (slow) power control. The distance dependent part of the static macro cell propagation model is the Hata model [9] converted to a frequency band around 2 GHz (BS antenna height equals 30 m, Suburban model). The attenuation of the signal power as a function of the distance d is given by [2, p.31] for β=4:

$$L_{FS} = 123 + 40\log(d) + \zeta \quad (12)$$

where d is the distance measured in km and ζ is a Gaussian stochastic variable which models the shadow fading. ζ = 6 dB is used for the shadowing margin ζ in L_{FS}.. The results for the coverage analysis are given in tables 3 and 4 for data and for speech service.

Table 3: Multiple Cell Coverage analysis results for data service (user bitrate is 128 kbit/s and coding rate R=1/2) with cluster size C = 3

cluster size C=3	Required E_b/N_0 (dB)	Transmitted Power (W)	ξ (dB)	Cell Radius (km)
Downlink	9.4	5	6	3.46
Uplink	5.7	1	6	2.87
Uplink	5.7	0.125	6	1.71

Table 4: Multiple Cell Coverage analysis results for speech service (user bitrate is 12 kbit/s and coding rate R=1/2) with cluster size C = 3

cluster size C=3	Required E_b/N_0 (dB)	Transmitted Power (W)	ξ (dB)	Cell Radius (km)
Downlink	9.4	5	6	6.24
Uplink	5.7	1	6	5.16
Uplink	5.7	0.125	6	3.25

The value β = 4 is a high value for the outdoor environment. Values smaller than β = 4 are often used. The results in table 3 and table 4 are therefore a little pessimistic.

V. Comparison of the DSR with a conventional receiver

The performance of the DSR in the uplink is compared with the performance of a conventional receiver. The results are given in figure 8.

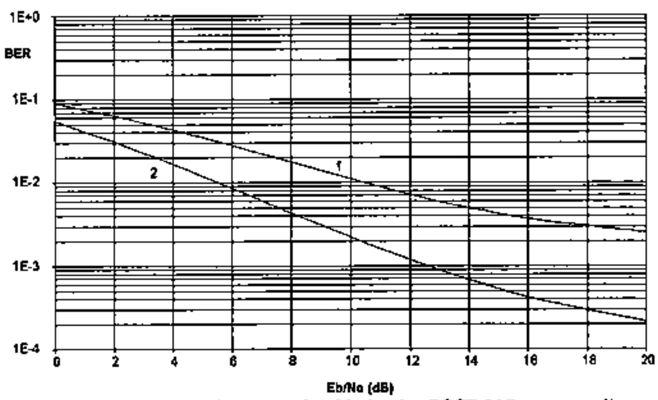

figure 8: BER as a function of E_b/N_0 for the COST 207 macro-cell channel with cluster size C=3 for 100 % system load, with:
1. BER for the large signal for the DSR with SIR = 3 dB
2. BER for a conventional BPSK receiver

If we compare the required E_c/N_0 of the conventional BPSK receiver with the required E_c/N_0 for the large signal using the DSR to reach BER = 3e-2 before decoding, we see a difference of 3.7 dB between the large signal of the DSR and the signal of the conventional BPSK receiver. If we assume that the performance for the small signal is the same as the performance of the large signal this would imply that, because SIR = 3 dB, the required E_c/N_0 of the small signal is only 3.7 - 3 = 0.7 dB higher than the required E_c/N_0 if a conventional receiver is used to reach the required BER = 3e-2 before decoding. So for the small signal the presence of the large signal costs only 0.7 dB.

At this moment GSM uses a conventional receiver (which has no DSR properties) and another modulation type than BPSK wich was investigated here. In GSM, the modulation type Gaussian Minimum Shift Keying (GMSK) is used. The results that are obtained in this paper are very interesting if we look at the main goal of implementing the DSR in the uplink connection : capacity enhancement. From table 2, we find that the capacity for the uplink is twice as big as the capacity for a conventional receiver (which is the performance of the downlink in the method that was used to model inter-cell interference), requiring a higher value of E_c/N_0 to fulfil the QoS requirement. Therefore it is worth investigating the performance of the DSR for GMSK modulation. Using this technique the available bandwidth for GSM and other narrowband mobile communication systems can be improved significantly.

VII. Conclusions

Results were presented for a Dual Signal Receiver in a TDMA/SFH protocol, using a system approach. It was shown that it improves the capacity in mobile cellular networks, compared with a conventional receiver. However, more transmitting power per bit is required to obtain the same performance. Also the range was computed using a link budget evaluation. The use of convolutional coding and antenna diversity showed to bring down the required value E_b/N_0 to fulfil the QoS requirement. These techniques result in a wider coverage but the results show that for macro-cell communications the energy requirements have to be brought down to provide wide area coverage. The system was evaluated for BPSK modulation. To evaluate the enhancement that the DSR would give in an existing mobile communication system like GSM, the evaluation should also be done for GMSK modulation. Using the DSR, the available bandwidth for narrowband mobile communication systems can be improved significantly.

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