Chapter 2
Radio Resource Management Performance for the GSM/EDGE Radio Access Network

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2.1 Introduction

This chapter presents a broad study on the potential of applying RRM techniques to the global system for mobile communication (GSM)/enhanced data rates for GSM evolution (EDGE) system. Even though the presented results are focused on the GSM/EDGE system, the principles employed by most of the considered RRM techniques can be applied to other radio access networks (RANs). We thus hope that the reader will also learn about RRM strategies and adapt these concepts to the RAN of his/her own interest.

The provision of multiple services is one of the key features of GSM/EDGE, which has been the focus of several studies, such as [22, 24, 25, 37]. Additionally, some of the themes that have been covered by recent research include power control [36, 41, 45], dynamic channel allocation [20, 46, 47, 49, 60], multi-antenna techniques [18, 19, 34, 48, 50, 51], among others.

In this chapter, the RRM techniques are placed within the context of GSM/EDGE and results are presented, which indicate the achievable gains in terms of capacity and/or quality of service (QoS). The results demonstrate that, by using appropriate RRM techniques, the GSM/EDGE radio access network capacity remains competitive with other emerging access technologies, thus allowing for substantial operational cost reductions for the already deployed infrastructure.

The remaining of this chapter is organized as follows: Section 2.2 briefly describes the architecture of the GSM/EDGE radio access network, along with its protocol stack and standard RRM functionalities. Section 2.3 presents the RRM techniques considered in the scope of this chapter, which are power control, dynamic channel allocation, management of multiple services, and multi-antenna techniques. Next, in Section 2.4, some aspects concerning the simulation and modeling of GSM/EDGE networks are discussed. The achieved simulation results are presented in Section 2.5 for the studied RRM techniques. Finally, trends and directions for the further evolution of RRM in GSM/EDGE networks are discussed in Section 2.6.

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2.2 Fundamentals of RRM in GSM/EDGE

The second generation of cellular systems was marked by a transition from analog-to-digital radio communications. GSM emerged in this context, with its phase 1 specification and initial deployment dating back to the early 1990s. GSM had a significant role in unifying the previously diverging European standards. The ubiquity of GSM, which facilitated international roaming among operators, the creation of the low-cost short message service, the support for circuit-switched data connections, as well as further improvements of the technology, such as the introduction of more efficient speech codecs, led to widespread GSM availability throughout the world, reaching the expressive mark of over 3 billion subscribers by the end of 2007 [27].

The provision of data services was improved with the introduction of the general packet radio service (GPRS) in 1997, which added support for packet-switched connections and provided four different coding schemes, with rates ranging from 8 to 20 kbit/s. In 1999, the enhanced GPRS (EGPRS) was introduced and then adopted as the packet system of the GSM/EDGE radio access network, which is the focus of this section. In the following section, an overview of GSM/EDGE is presented along with some of its standard functionalities.

2.2.1 GSM/EDGE Radio Access Network Overview

The GSM/EDGE radio access network (GERAN) represents the evolution of the GSM system for providing improved packet data transmission. The GPRS and EGPRS are radio technologies that provide packet-switched connections between MS and BS, while the GERAN is composed of several network elements that are interconnected through standard interfaces. The two main elements of the radio access network are

- **Base station subsystem (BSS)**: It comprehends the base transceiver station (BTS), or simply BS, which is the onsite base station, and the base station controller (BSC), which is a controlling unit responsible for a group of BTSs.
- **Core network**: It has functionalities such as mobility management, authentication, charging, among others, and also provides access to networks outside of the cellular system. In the case of circuit-switched connections, the mobile switching center (MSC) is the main element of the core network, providing accessibility to the conventional public-switched telephone network (PSTN). In the case of packet-switched connections, the main elements are the service GPRS support node (SGSN) and the gateway GPRS support node (GGSN). The former performs routing and delivery of packets within the cellular system and the latter provides connectivity to external data packet networks, such as the Internet.

With the purpose of maintaining compatibility with the existing GSM infrastructure, the EDGE technology has many parameters in common with GSM, including the sharing of the same frequency spectrum. EGPRS introduces some improvements with regard to GPRS, such as the 8-PSK (phase-shift keying) modulation and additional modulation and coding schemes (MCSs). Due to this improved physical layer, EGPRS provides high data rates, reaching 384 kbit/s or more when multiple timeslots are reserved to a single MS.

Through the adequate configuration of parameters from the protocol layers of GSM/EDGE, it is possible to provide multiple services. The data transmission of EGPRS may be adjusted, for example, to support applications with different quality-of-service (QoS) requirements, such as World Wide Web (WWW), file transfer protocol (FTP), and streaming of audio/video files.

The standardization of the GSM/EDGE radio access network is coordinated by the 3rd. Generation Partnership Project (3GPP). The standards have undergone some major revisions, with Rel-8 being the latest release as of 2008.

The GSM/EDGE network is already well established, in terms of technical maturity as well as market deployment. This, however, has not stopped the development of new techniques for improving its performance and providing capacity gains.

2.2.1.1 Channel Structure

The GSM/EDGE system implements multiple access through frequency division as well as through time division. Each frequency carrier has a cyclic time structure associated, which is composed of hyperframes, superframes, multiframes, frames, and timeslots [7], as it can be seen in Fig. 2.1.

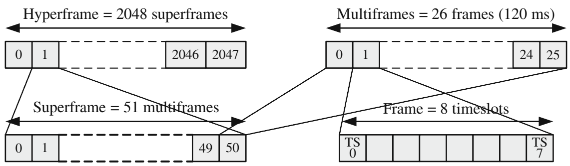


Fig. 2.1 GSM/EDGE frame structure.

Besides the displayed frame structure, which is employed for traffic channels, there is an alternative signaling frame structure that defines a multiframe with 51 frames and a superframe with 26 multiframes.[2.1]

The basic time unit is the timeslot, which is equivalent to roughly 0.577 ms. A sequence of eight timeslots defines a time division multiple access (TDMA) frame, and a group of four frames composes a TDMA radio block.

Among the 26 frames of the multiframe structure in Fig. 2.1, the 13th and the last frame are reserved for control and other functionalities. Therefore, the other 24 frames may be employed for traffic, i.e., six radio blocks.

A physical channel is defined by the pair timeslot/frequency. A logical channel, on the other hand, corresponds to the information flow between a BS and an MS.

[2.1] A superframe still contains 26×51 frames in total.

The logical channels may be divided into traffic and control channels. Next, some of the most relevant logical channels are presented:

- **Traffic channel (TCH)**: This is a circuit-switched traffic channel used for voice as well as circuit-switched data transmission.
- **Packet data traffic channel (PDTCH)**: This is a packet-switched traffic channel used for data transmission.
- **Broadcast control channel (BCCH)**: This is a downlink control channel that distributes general information to the MSs concerning the system configuration. The information may include number of common control channels, possible combinations of control channels, whether support for packet-switched traffic is enabled, among others. There is also the packet broadcast control channel (PBCCH), which is the corresponding channel for data MSs.
- **Common control channel (CCCH)**: It corresponds to a set of common control channels that are used for implementing access management functions.
- **Dedicated control channel (DCCH)**: It corresponds to a set of dedicated control channels that are used for measurements, signaling, among other functionalities. The main circuit-switched dedicated control type channels are the slow associated control channel (SACCH) and fast associated control channel (FACCH), which provide connection-specific signaling information concerning the channels they are associated to, and the stand-alone dedicated control channel (SDCCH), which can be used for signaling during call setup. These channels can be used both in the uplink and downlink.

The SACCH, for example, is important for transmitting information related to signal level and signal quality measurements. The reporting periods have a duration of 480 ms (104 TDMA frames) and they are employed, e.g., by the power control (PC), handover, and link adaptation (LA) algorithms discussed later in this chapter.

The actual radio transmission requires that the different logical channels be mapped onto the physical channels. A physical channel may be composed of only control channels, two half-rate traffic channels plus control channels, or a full-rate traffic channel plus control channels. The combination of traffic and control channels is possible by either employing the previously described reserved frames, in the case of SACCH, or by stealing slots from traffic channels, in the case of FACCH. A more detailed description of the possible channel combinations can be found in [6].

2.2.1.2 Protocols

This section presents an overview of the protocol stack of the GSM/EDGE network for packet data transmission. Figure 2.2 shows how the user plane[2.2] protocol layers are organized among the different network elements in the 3GPP standards.

The focus of this chapter lies on the radio link between the MS and the BS, which is supported by the following three protocol layers:

[2.2] There are protocol stacks for the user plane and control plane. The former refers to the actual data transmission and the latter is used for control and signaling.

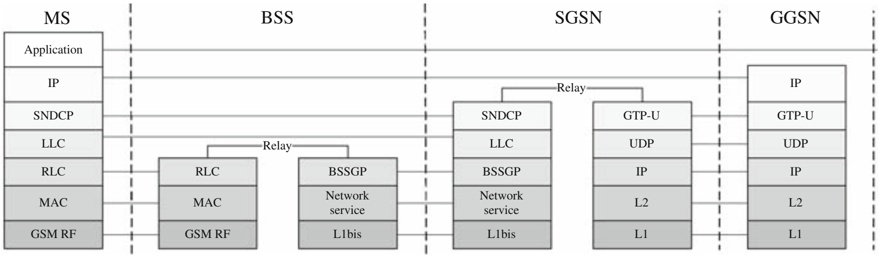


Fig. 2.2 Packet-switched user plane protocols of the GSM/EDGE network.

- **Link layer control (LLC)**: It offers a reliable and secure logical link between the MS and the SGSN for superior layers. One of its main functionalities consists of performing the segmentation of packets arriving from higher layers.
- **Radio link control (RLC) and medium access control (MAC)**: These protocols provide services for the transfer of information over the physical layer. Among their functionalities are the error-correcting procedures enabled through the selective retransmission of erroneous blocks. The RLC function offers a reliable radio link to the higher layers, while MAC treats issues such as channel allocation and the multiplexing/scheduling of MSs.
- **GSM RF or physical layer**: It provides data transfer services over the physical channel between the BS and the MS. Among its functionalities are the coding of data and the detection/correction of transmission errors in the physical medium.

The data transmission process can be briefly described as follows. The packets that arrive from the internet protocol (IP) and sub-network-dependent convergence protocol (SNDCP) layers are segmented into LLC layer frames. The LLC frames are segmented into RLC/MAC blocks as they are being requested by the system. The RLC/MAC blocks are transmitted over the four bursts of a TDMA radio block, where the term burst corresponds to the transmission of data during the time of a timeslot.

The RLC/MAC blocks are composed of header and data fields, which have variable lengths depending on the current MCS. EGPRS has nine MCSs, with the first four employing Gaussian minimum shift keying (GMSK) modulation and the remaining ones 8-PSK. The lowest MCSs transport a smaller amount of information data per block, but are more robust to variations in the link quality.

The MCSs are organized into different families, each with a certain base payload [4]. In the case of retransmissions, only an MCS of the same family may be chosen. MCSs 7–9 transmit two RLC/MAC blocks per TDMA radio block, while the others transmit only one block. A summary of the main MCS parameters is presented in Table 2.1. The interested reader can refer to [3, 5] in order to obtain a detailed description of each protocol in the GSM/EDGE protocol stack.

Table 2.1 MCS parameters (with data rate in kbit/s and payload in bytes per TDMA radio block).

MCS	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**
Modulation	⊢	GMSK		⊣	⊢		8-PSK		⊣
Code rate	0.53	0.66	0.85	1.0	0.37	0.49	0.76	0.92	1.0
Data rate	8.8	11.2	14.8	17.6	22.4	29.6	44.8	54.4	59.2
Payload	22	28	37	44	56	74	112	136	148
Family	C	B	A	C	B	A	B	A	A

2.2.2 Link Adaptation

The link adaptation (LA) mechanism of EGPRS, which is described in [8], tries to provide the best possible quality to the MS through the modification of the current MCS. This adaptation occurs according to the availability of link quality measurements and it intends to exploit the channel diversity and maximize data rates by suitably selecting an MCS according to the channel state.

Ideally, LA could be employed on a per-block basis, i.e., a new MCS would be selected for each radio block (20 ms) [38, 39]. In practice, however, the standard LA mechanism uses the same link quality estimations used by power control, which are periodically reported to the BS each 480 ms [9].

Figure 2.3, based on results presented in [22], shows how LA behaves according to the SIR. For a given SIR, the MCS providing the highest data rate should be selected.

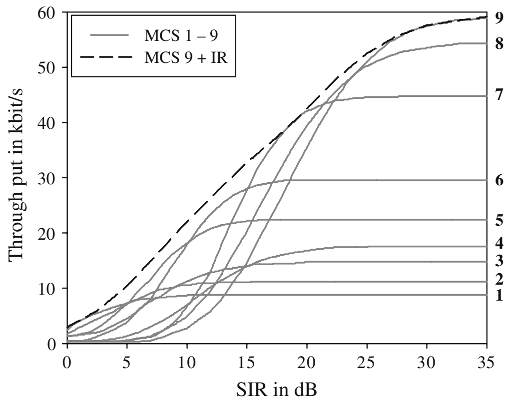


Fig. 2.3 Link adaptation with pedestrian mobility (3 km/h).

Another mechanism, which may replace or act together with LA, is the incremental redundancy (IR), which is also shown in Fig. 2.3. With IR the amount of redundancy is increased for each additionally required retransmission. IR improves the reception of retransmissions by combining the retransmitted blocks with the data already available at the receiver. More details on IR can be found in [23, 40].

2.2.3 Frequency Hopping

The frequency hopping (FH) technique consists of periodically changing the transmission frequency with the purpose of introducing diversity. The diversity effect may include both frequency and interference diversity, which are illustrated in Fig. 2.4 for two co-channel MSs.

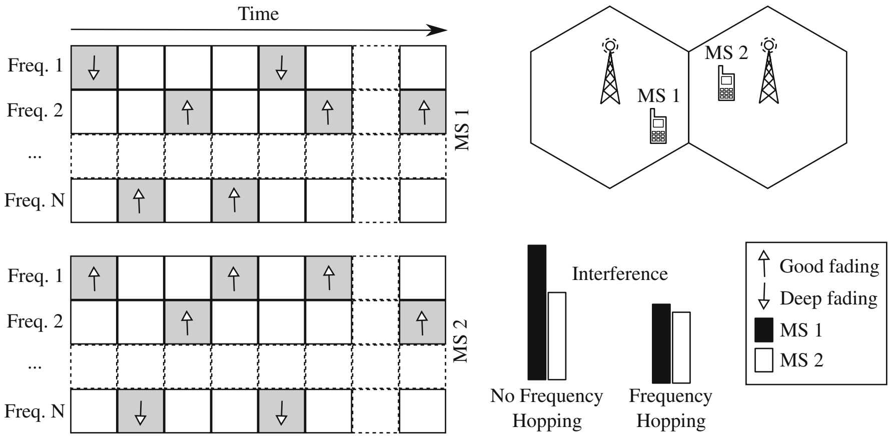


Fig. 2.4 Frequency hopping.

Because a different frequency is used after each hop, the MSs perceive different fading gains at each time, as indicated by the arrows in Fig. 2.4. Consequently, the users' links do not remain for a long time in a deep fading and a more reliable communication might be achieved.

Regarding interference, suppose some MSs perceive strong interference while other MSs perceive weak interference (or no interference at all), as illustrated in Fig. 2.4. Without frequency hopping, the links of some users would be subject to high interference for a long time. By hopping across different frequencies, the set of co-channel interferers seen by a user changes after each hop and the MSs would experience periodically changing interference profiles, so that almost the same average interference would be perceived by all MSs.

Thus, the main benefit of frequency hopping is to average out the fading and interference effects, thus allowing the use of more aggressive reuse patterns, such as 1/3 and 1/1.[2.3] The types of frequency hopping are described as follows:

[2.3] This notation indicates the cluster size of the frequency reuse pattern, i.e., the group of cells/sectors within which there can be no reuse of the available frequencies.

- **Random frequency hopping (RFH)**: It performs the hopping in an unordered fashion, according to a pseudo-random sequence determined based on system parameters and the algorithm presented in [6].
- **Cyclic frequency hopping (CFH)**: It performs the hopping in an ordered fashion, according to a previously established cyclic sequence.

RFH provides both frequency and interference diversity, while CFH presents frequency diversity, but not interference diversity, since the co-channel MSs hop over the same frequencies. The main system parameters involved with the frequency hopping algorithms are presented in Table 2.2.

Table 2.2 Description of the frequency hopping parameters.

Parameter	Description
MAL	Mobile allocation list containing the frequencies available for allocation
MAIO	Mobile allocation index offset indicating the offset within the MAL
N_{freq}	Number of frequencies per mobile allocation list
FN	TDMA frame number currently in use
HSN	Hopping sequence number allocated to each sector
MAI	Mobile allocation index referencing the frequency of MAL to be used

The frequency hopping algorithm proposed in [6] can be described as

$$\begin{cases} \mathrm{MAI} = (\mathrm{FN} + \mathrm{MAIO}) \bmod (N_{\mathrm{freq}}), & \text{if } \mathrm{HSN} = 0, \\ \mathrm{MAI} = (S + \mathrm{MAIO}) \bmod (N_{\mathrm{freq}}), & \text{if } \mathrm{HSN} \neq 0, \end{cases} \tag{2.1}$$

where the first equation relates to CFH and the second to RFH. The S variable corresponds to the generation of the pseudo-random sequence, which is a function of the FN, HSN, N and the hopping table defined in [6]. The standard defines 64 possible orthogonal hopping sequences. The orthogonality is assured for MSs that have the same HSN but different MAIOs, i.e., they never become co-channel interferers.

The MAIO allocation to MSs entering the system can be done either at random, in the case of RFH, or it can follow a certain allocation algorithm, such as the one defined in Section 2.3.2, in the case of CFH. More details on the implementation of frequency hopping for GSM/EDGE can be found in [6, 42].

2.3 Advanced Radio Resource Management for GSM/EDGE

Since the deployment of the first GSM-based networks, the demand for mobile communication has increased enormously. Conventional GSM/EDGE networks have reached their capacity limits and, in order to serve the growing demand for mobile communication, solutions to increase system capacity are required.

Spectrum is a scarce resource whose use is granted and regulated by estate institutions, such that a capacity expansion through the acquisition of new frequency bands may become very expensive. In this context, techniques to optimize the usage of radio resources, i.e., RRM techniques, gain importance as alternative to enhance the system capacity without needing additional spectrum.

This section describes several advanced RRM techniques contextualized in a GSM/EDGE network. The RRM techniques discussed here are not necessarily an integral part of the GSM/EDGE standard, but might be incorporated as part of proprietary solutions.

2.3.1 Power Control

Power control (PC) is a well-established RRM technique which aims at, mainly, reducing interference levels in a wireless network and conserving battery power of terminals. A detailed discussion about PC has already been provided in Chapter 1 of this book. In this section, some particular aspects of PC in the context of GSM/EDGE networks are detailed.

PC is well described in the standard for the uplink of GSM/EDGE networks [9, 10]. In the downlink, PC is not a mandatory feature. However, a downlink PC algorithm can also be implemented at the BSs as long as the restrictions imposed by the standard are respected. The power characteristics of base and mobile stations are described in [9, 10] and among the standard restrictions are, for example, the usage of discrete power levels in steps of 2 dB and dynamic power ranges limited to 30 dB and to 10 dB for voice and data services, respectively.

The standard PC algorithm for the uplink in the GSM/EDGE network is a variable-step up-down power control (UDPC) algorithm with some allowed step sizes (all in integer multiples of 2 dB) [9, 10]. However, arbitrarily large power adjustments, such as 30 dB at once, are not foreseen. The up-down algorithm described in Chapter 1 is the PC algorithm that can more easily be related to the standard algorithm considered in the uplink of the GSM/EDGE network. In this chapter, the up-down algorithm of Chapter 1 is referred to just as UDPC.

In order to apply PC, link measurements are required. Indeed, the actuation frequency and performance of PC depend directly on the availability of such measurements at the BS.

For the voice service, two standard measurements performed by the MS are the received signal level (RXLEV) and the received signal quality (RXQUAL). RXLEV values are measured in the range of -110 to -48 dBm for each TDMA frame within one SACCH multiframe and are mapped afterward to one of 64 possible levels. Average RXLEVs are then reported by the MS to its serving BS. For the RXQUAL, the GSM/EDGE standard states that its value must be related to the BER before decoding, also termed raw bit error rate (RBER). RBER values can be estimated, e.g., as part of the channel equalization or decoding processes. For example, a method to estimate RBER values consists of comparing a reencoded version of a correctly decoded frame with the originally received frame, which is required due to the data interleaving done across the half-bursts of the frame. Based on the estimated RBER values, an average RBER over one SACCH multiframe should be computed, mapped to one of eight possible discrete values, and reported by the MS to its serving BS, where it can be used to perform, e.g., downlink PC, LA, and handovers. Average RXLEV and RXQUAL values become available at the BS at the end of the subsequent SACCH multiframe, thus imposing a delay of 480 ms between measurement and availability of the measured values at the BS. Thus, it should be noted that PC actuates at a very low frequency in the GSM/EDGE network with two power adjustments each second. Some particular cases in which this frequency might be higher are also defined by the standard.

For data services, the GSM/EDGE standard states that the link quality measurements should be related to the bit error probability (BEP) within each burst of the radio blocks received within one SACCH multiframe. The BEP of the four bursts of a radio block are used to calculate the mean BEP (MEAN_BEP) and the coefficient of variation of the BEP (CV_BEP) of the block, as described in [9]. MEAN_BEP and CV_BEP values are computed for all the correctly decoded radio blocks within the duration of one SACCH multiframe. The average of the MEAN_BEP and CV_BEP values are calculated from these values, mapped to 32 and 8 values, respectively, and reported back by the MS to its serving BS, where they can be used to perform, e.g., downlink PC, LA, and handovers. Since data is interleaved across the bursts of a block, the BEP of each burst should be estimated using, e.g., the same method described for the voice service, i.e., the comparison of a reencoded version of a correctly decoded block with the originally received one.

For data services in the GSM/EDGE network, PC is a particularly challenging task because of the bursty nature of data traffic. The size of data packets may vary in a broad range of values and such packets require quite different amounts of time to be transmitted. For small packets, iterative PC may not have enough time to converge to the target SINR before the packet is completely transmitted. Moreover, depending on the adopted scheduling discipline the MS transmitting on the shared channel may change each 20 ms, thus also affecting the convergence of PC algorithms.

In the GSM/EDGE network, LA has priority over PC and the dynamic power range available for PC is limited to 10 dB for data services instead of the 30 dB used for the voice service. The reduced dynamic power range leads to higher average interference levels [45].

One reason for a higher minimum transmit power is to ensure a higher reliability in the reception of the uplink state flag (USF) transmitted within the downlink RLC blocks. This 3-bit flag is stored in the header of RLC blocks sent on the downlink and is used to coordinate channel accesses in the uplink. The USF is decoded by all MSs sharing a downlink channel via TDMA and indicates which MS gets access to the uplink channel during the next radio block. Thus, in order to enable scheduling in the uplink, all MSs must be able to reliably decode the USF independently of their positions within the cell. Throughout this chapter, every time the transmit power range or the transmit antenna pattern are modified in the downlink, there is a risk that the USF might not be decoded correctly. This is a practical GSM/EDGE issue that the RRM algorithms need to overcome.

PC and LA have somewhat conflicting objectives. The former usually aims at providing just the minimum required quality to the links, for example, a target SINR, thus reducing power consumption at MSs and the overall interference in the system. The latter aims at providing the highest possible data rate according to the current link quality while employing maximum transmit power. In order to avoid concurrence between PC and LA, previous works on PC for data services in the GSM/EDGE network considered high target SINR values lying outside the interval in which LA works [45]. However, with such high target SINR values, only a very small number of links benefit from PC, while most of them transmit at full power.

EGPRS can be considered an energy-efficient service, since it maximizes throughput for a fixed transmit power by LA and, consequently, MSs' data sessions will probably take shorter times in average. However, maximizing instantaneous rates might not always be the best policy, especially when considering mixed-service scenarios where one service may be experiencing an excess quality while the other services are below their QoS limit. This subject (co-existence of multiple services) will be further explored in this chapter.

Different downlink PC algorithms are considered later in Section 2.5.2 of this chapter, which involve non-standard features including power adjustments of up to 30 dB at once and higher actuation frequency, such as one power adjustment at each 120 or 20 ms. Regarding data services, the impact of increasing the dynamic power range of PC from 10 to 30 dB is also investigated.

2.3.2 Dynamic Channel Allocation

The channel allocation procedure consists, essentially, of distributing a finite number of channels among the several base and mobile stations within a cellular network. An efficient channel allocation algorithm may lead to benefits for the system, be it in terms of reduced blocking rates or QoS improvements. Classically, algorithms may be classified as fixed or dynamic [33], even though there are also those which combine characteristics of both, which are called hybrid.

Dynamic channel allocation (DCA) assumes that there is a central channel pool, from which the channels may be allocated on-demand, i.e., there is no fixed distribution of the channels among the cells. This higher flexibility allows that the fluctuations in the offered traffic and co-channel interference be treated with a higher efficiency. DCA thus requires that the network be capable of offering all frequencies within each cell, as well as providing reliable measurements of the parameters used by the DCA algorithms (e.g., number of MSs and radio link quality).

The DCA algorithms may be classified according to the metric they optimize [59] or to their degree of centralization [15]. The first criterion considers characteristics such as adaptability to traffic and interference, as well as channel reusability. The second form of classification concerns the degree of centralization of the algorithm, e.g., centralized, distributed, or locally distributed. Locally distributed algorithms allow the exchange of information among nearby BSs, aiming at the improvement of the quality estimation procedure and consequently of the allocation decisions.

The centralized approach has a rather high implementation cost, mainly due to the excessive signaling. The fully distributed strategy may also not be adequate, since it does not provide the necessary means for reliable interference estimation. A locally distributed algorithm represents the most feasible approach, allowing the exchange of information among the neighboring BSs in order to achieve more precise interference estimates. For these reasons, and since high interference tight reuse patterns provide the highest capacity potential, here we focus on a locally distributed interference adaptive algorithm.

The application of DCA to an actual system must take into account the characteristics and practical limitations of the cellular network. Some works have proposed and evaluated DCA algorithms specifically for GSM/EDGE, such as the dynamic frequency and channel allocation [46, 47].

It is worth mentioning that the use of DCA in this context is not compatible with random hopping. Due to the fact that the MSs are constantly hopping over the frequencies in an unordered fashion, the channel selection procedure becomes irrelevant, since with each hop the set of co-channel interferers may change completely. It is therefore required that frequency hopping be disabled or that a coordinated cyclic hopping be implemented, in which the groups of co-channel interferers hop over the same frequencies.

2.3.2.1 Measurements and SIR Estimation

The GSM/EDGE cellular network does not count with direct SIR measurements. The SIR must be inferred based on the measurement report mechanisms available in the network [9].

Each MS monitors the power levels arriving from nearby BSs. This information is accumulated and periodically reported to the BS to which the MS is connected. The measurements are done for the BCCH channel and the report period is of 104 TDMA frames (480 ms). For each frame a different BS is measured. The number of BCCH carriers and the measurement order are parameters defined by the system. As an example, suppose that the BCCH list contains 32 elements, then there will be (104/32) received power measurements for each carrier, i.e., three or four samples per BS.

After the measurements within the report period are concluded, the MS has to organize the data and prepare the report to be sent to the BS. The samples of each carrier are averaged, and among all measured carriers only the six with the highest received power levels are included within the report. The transmission occurs through the SACCH control channel during the next 480 ms. The total delay until the report is available at the BS, including the measurement and transmission times, is therefore of roughly 1 s.

The original purpose of this measurement mechanism would be to aid in the handover decisions, indicating which cells offer the best signal quality to the MS. Nevertheless, it may also be employed to produce channel SIR estimates.

The reports are based on measurements of the BCCH channel, which is always active. In order to obtain a more realistic SIR estimation, the actual channel activity of the BSs should be taken into account. However, this channel activity information is not globally available, it has to be shared among the BSs [46]. In practical terms this information exchange could be feasible, representing a signaling increase within the backbone of the cellular network. Note that if power control is employed in the system, it would also be necessary to share the power adjustments of each BS, since the BCCH measurements are done for full power.

2.3.2.2 Channel Selection and Admission Control

The considered DCA algorithm prioritizes the channel presenting the best SIR. In the case when several channels perceive no interference, or when the estimated SIR is the same, the choice is done at random among them.

The admission control corresponds to an optional stage of the DCA algorithm. Differently from the case with RFH, with DCA the MS will be subject to the same interference profile for a certain period of time, therefore it is important to guarantee that the channel is offering a minimum acceptable quality.

Even though the channel selection procedure prioritizes the channel perceiving the best SIR, high-load situations may occur, for which even the best channel would not be able to offer a satisfactory quality to the MS. In such cases, blocking the MS might be a better option than letting it enter the system, since it is expected that its QoS will probably not be satisfied. Besides, the additional interference that would be introduced in the system is avoided.

The admission criterion can be based on a minimum SIR threshold, which may be defined based on link-level simulation results. In the case of the enhanced full rate (EFR) speech codec, for example, a minimum SIR of 8 dB is required in order to assure that the frame erasure ratio will be kept at acceptable levels [24].

Another aspect that may be taken into account by the admission control is related to the impact that the introduction of a new MS might have over the quality of the MSs already allocated in the system. This "impact test" corresponds to performing an estimate of how the SIR of the co-channel MSs would be degraded after admitting a new MS. In case the admission would result in the SIR of any of the co-channel MSs being reduced to a value below the threshold, blocking would be activated. Note that the complexity for implementing such estimate might be prohibitive in practical terms.

2.3.3 Management of Multiple Services

The support of multiple services, such as web-browsing, e-mail, audio/video streaming, among others, is one of the main features of the third generation of cellular systems and beyond. Since these services have different characteristics and requirements, the use of efficient RRM techniques is therefore essential to ensure their QoS levels and to optimize system capacity.

In systems with multiple services the available radio resources may be either segregated or shared, i.e., different frequency groups may be reserved for the services or it may be allowed that they have access to the total set. The isolated approach is not very efficient, since for situations of asymmetric load the overloaded service does not have access to the channels reserved to the other services, even though they may be unoccupied. The sharing of channels avoids problems of this nature, but has the side-effect of creating a situation in which the different services cause interference among each other, which may have certain implications on the capacity of interference-limited systems.

Since the services have different QoS requirements and different traffic patterns, their combination within interference-limited scenarios implies that the more demanding service will limit system capacity, even though the other services still may perceive sufficient QoS. In [54], a method has been proposed for balancing the QoS in code division multiple access (CDMA) systems, based on a power allocation methodology that allows the joint service capacity to be increased.

A more general definition for QoS balancing was presented in [22, 25], which was called per-service capacity balancing. The referred work has demonstrated that the system capacity is maximized when the per-service capacities are reached for the same load. In the case of interference-limited systems, it has also been shown that the capacity balancing may be achieved through an interference balancing process, such as the service-based power setting (SBPS) technique for mixed-service GSM/EDGE networks [24], which consists of applying offsets in the transmission powers of the different services.

2.3.4 Multi-antenna Techniques

The application of multi-antenna techniques to mobile communication systems is capable of providing capacity gains as well as the improvement of the quality of service perceived by the subscribers. The spatial filtering realized through the use of narrow beams, which can be either selected from a fixed set (switched fixed beams) or adaptively steered toward the desired MSs (adaptive beamforming), is able to significantly reduce the co-channel interference levels. This approach can be considered a more flexible and evolved form of the sectorization that is employed as a baseline in most mobile communication systems. The interference reduction allows for the implementation of more aggressive frequency reuse patterns, such as 1/3 or 1/1, thus resulting in higher spectral efficiencies.

The switched fixed beams, as well as adaptive beamforming based on direction-of-arrival estimation techniques, are more adequate to macrocellular environments with low angular spread and strong line-of-sight [55]. In the case of indoor or microcellular environments presenting strong multi-path components, the adaptive arrays must rely on the estimation of the complex channel coefficients in order to adapt to the current channel conditions [11].

The gains provided by multi-antenna techniques, however, are not limited to those of the spatial filtering functionality. Techniques that take advantage of the spatial diversity associated with the presence of multiple antennas may also be applied, such as the maximum ratio combining (MRC) or the interference rejection combining (IRC).

In the context of the application of adaptive antennas to multiple services in GSM/EDGE networks there are some practical aspects that should be taken into account, mainly with regard to the data service over EGPRS. For the voice service there would be no problem with replacing the sector antennas with antenna arrays, other than the restriction that they may not be applied to the BCCH carriers, for the same reason that other techniques such as FH and PC may not be used, which is to guarantee that all MSs have uninterrupted access to the broadcast channel.

The results section carries out the evaluation of some multi-antenna strategies for scenarios where both voice and data share the same frequency spectrum (see Section 2.5.4). It assesses, among other things, the performance of a strategy for which the adaptive antennas are applied only to the voice service, in order to avoid the problems associated with their use within EGPRS. Since all MSs hop over the same set of frequencies, the interference reduction for the voice MSs is also reflected upon the data MSs, thus bringing benefits for both. The performance of the strategy combining different antennas for the different services is also compared to the fully multi-antenna case, i.e., with both services employing multiple antennas.

2.4 Simulation and Modeling of GSM/EDGE Networks

Studying the performance of modern wireless networks, such as GSM/EDGE, is a complex task. Due to the large number of variables and mechanisms involved, a pure analytical study is not feasible and computer simulations are applied to investigate the system's characteristics of interest [31, 35].

In this chapter, the strategy of dividing the simulations into link and system levels is employed. These two types of simulators are then connected through an appropriate interface. This is a complexity reduction strategy, which is discussed in more details in Chapter 7 of this book. We also employ dynamic system simulations, which include channel and traffic variabilities with time. In the following, we describe several simulation models, which are used later to assess the performance of RRM techniques in a GSM/EDGE network. In spite of considering here a GSM/EDGE network, many of the models introduced in this section are quite general and can be used to evaluate the performance of other modern wireless networks.

2.4.1 Cellular Grid, Frequency Reuse, and Mobility Models

In this chapter, the cellular network is modeled as a macrocellular system composed of tri-sectored cells organized in 1/3 or 1/1 uniform frequency reuse patterns [59]. Figure 2.5 illustrates a cellular grid with a 1/3 frequency reuse.

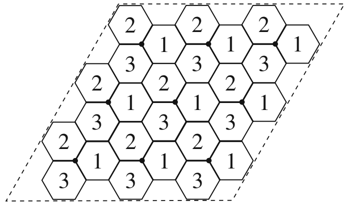


Fig. 2.5 Uniform cellular grid with 1/3 frequency reuse.

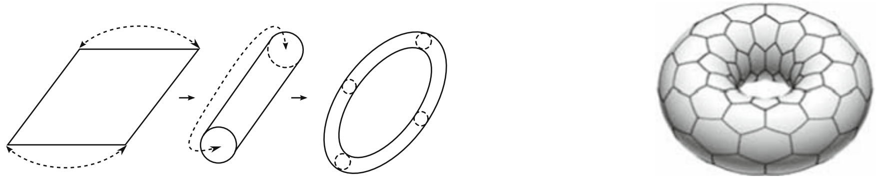
In Fig. 2.5, the sectors with same number employ the same set of channel frequencies and are co-channel interferers. A number of MSs are distributed over the area covered by the cellular grid and can freely move within it. Each cell sector in Fig. 2.5 is assumed to use a typical sector antenna, such as that presented in [53].

MSs' mobility can be modeled according to a random-walk (Markovian) mobility pattern [16, 17]. Only pedestrian mobility is considered in this chapter, which assumes an average speed of 3 km/h. The current MS' speed and direction of movement are uniformly distributed within [1 km/h, 5 km/h] and $[0, 2\pi]$, respectively. These are held until the MS walks a distance of 5 m, after which new speed and direction are sorted. Other mobility models, including that of vehicular mobility, can be found in [16, 17].

Because MSs move over the grid shown in Fig. 2.5 and because only a limited area is covered, MSs could eventually leave the coverage area. Additionally, due to the geographic distribution of the co-channel sectors in Fig. 2.5, sites on the border of the grid perceive less interference than those in middle of the grid, which is termed a border effect. In order to allow infinite mobility over a limited region and to avoid border effects, which are undesired, a wrap-around technique is usually considered. Wrap-around techniques are usually based on cell replication or on a geometric model which yields homogeneous average interference levels in the whole grid. Herein, the wrap-around technique described in [59] is employed, which consists of bending the grid in order to form a torus surface, as illustrated in Fig. 2.6. Note that the described grid and mobility models can be directly applied to other wireless networks.

2.4.2 Propagation Models

As it has been discussed in Chapter 1, radio communication is affected by large- and small-scale fading, which ultimately result from reflection, refraction, and diffraction of the transmitted radio waves [44, 56]. These effects are assumed in the



(a) Torus model (b) Torus mapping

Fig. 2.6 Infinite mobility model.

GSM/EDGE simulations considered in this chapter and their models are described in the sequel.

There are different average path loss models, which are adequate for different propagation scenarios [44, 56]. Herein, the Okumura–Hata model is employed, which applies to urban and suburban environments where the average building height is approximately uniform [53]. Considering this model and denoting by d the distance in km between a BS and an MS, by f_c the system central carrier frequency in MHz, and by h_{BS} the BS height in meters and measured with respect to the average rooftop of buildings, the average path loss L_{pl} is given by

$$L_{pl} = 40(1 - h_{BS} \times 4 \times 10^{-3}) \log(d) - 18 \log(h_{BS}) + 21 \log(f_c) + 80 \text{ in dB}. \quad (2.2)$$

As the MS moves in the coverage area, large obstacles such as buildings may obstruct the propagation path between the BS and MS causing fluctuations on the received signal power, i.e., shadowing the received signal. Shadowing is usually modeled by a lognormal random variable with standard deviation σ_{sf} [44, 56]. Because shadowing relates with the position of large obstacles in the coverage area, it is position-dependent and spatially correlated. The modeling of the spatially correlated shadowing can be accomplished by sorting independent lognormal shadow samples for a rectangular grid composed of points uniformly separated by a shadowing decorrelation distance d_{sf} [59]. This model is termed a shadowing map and for each BS in the network such a shadowing map is created. Then, the shadowing $L_{sf}(x,y)$ associated with an arbitrary position (x,y) in the grid can be obtained through linear interpolation.

Besides the spatial correlation of shadowing captured by the above model, it is also worth modeling the spatial correlation of the shadowing perceived in the links between different BSs and the same MS. This kind of correlation occurs, for example, when the MS moves within a tunnel. This inter-BS shadowing correlation can be modeled with help of an additional shadowing map, which is associated with all MSs. In order to obtain the shadowing sample $L_{sf}^{(b,m)}(x,y)$ for the link between a BS b and an MS m, the shadowing samples $L_{sf}^{(b)}(x,y)$ and $L_{sf}^{(m)}(x,y)$, obtained respectively from the BS' and MSs' shadowing maps, are combined as

$$L_{sf}^{(b,m)}(x,y) = \sqrt{1-\rho_{sf}} L_{sf}^{(b)}(x,y) + \sqrt{\rho_{sf}} L_{sf}^{(m)}(x,y), \quad (2.3)$$

where $0 \leq \rho_{\mathrm{sf}} \leq 1$ is a coefficient which controls the amount of shadowing correlation among BSs [59].

Multi-path fading leads to deep fluctuations in the received signal power. It has been modeled in the link-level simulations, where time-correlated fast fading is generated using the well-known Jakes' model [30].

2.4.3 Link Quality Measurements

Considering the presented radio propagation models, the measures considered in the system simulations to evaluate the communication links between BSs and MSs are discussed in the following.

The power P_{r} received by the MS depends on the transmit power P_{t} used by the BS and on the particular state of the link between BS and MS. The received power P_{r} of an MS can be expressed as

$$P_{\mathrm{r}} = P_{\mathrm{t}} + G_{\mathrm{ant}} - L_{\mathrm{pl}} - L_{\mathrm{sf}} \text{ in dBm,} \tag{2.4}$$

where the antenna gain G_{ant}, the average path loss L_{pl}, and the shadow fading L_{sf} can be obtained considering the relative position of the BS and MS. Additionally, extra gains/losses, such as cabling losses at the BS or additional antenna gains at the MS, can be easily added/subtracted in (2.4). For simplicity, such additional gains/losses are not considered here.

Co-channel sectors share the same frequencies and generate interference. A common measure of the link quality corresponds to its SINR. Assume that MS i is served by the sector i and let N_{ci} denote the number of interfering co-channel sectors. Denoting by $p_{\mathrm{r}_{i,j}}$ the power received by MS i from sector j, and by ν the average noise power, the SINR γ_i of MS i is given by

$$\gamma_i = \frac{p_{\mathrm{r}_{i,i}}}{\sum\limits_{j=1,\, j\neq i}^{N_{\mathrm{ci}}} p_{\mathrm{r}_{i,j}} + \nu}. \tag{2.5}$$

For interference-limited scenarios, the average noise power ν can be neglected and the SINR in (2.5) reduces to the SIR.

In the link-level simulations, curves such as bit error rate (BER), block error rate (BLER), or frame erasure rate (FER) as functions of the SINR (or SIR) are obtained by averaging the link performance over a long period of time. In this way, the mean characteristics of mechanisms like data interleaving, fast fading, and fast interference variations are captured into the link-level results. In the system-level simulations, measures of the SINR (or SIR) can be easily obtained and can be mapped afterward into BER, BLER, or FER values using an adequate link-level curve.

Considering voice services, the average SINR of the eight half-bursts composing a voice frame is mapped to an FER value. Considering data services, the average SINR of the four bursts composing a radio block is mapped to a BLER value. Based on the FER, in the case of the voice service, or on the BLER, in the case of data services, a random test is used to determine whether a transmission has been successful.

For the simulations considered in this chapter, the RXQUAL, MEAN_BEP, and CV_BEP values shortly described in Section 2.3.1 cannot be measured as described in the standard because they depend directly on RBER. Instead of the standard measurements, the average SINR $\bar{\gamma}_s$ over one reporting period is considered as link quality measurement. It is computed by averaging the mean SINR of each block within the reporting period and it is sent back to the BS by the MS. This model is applied here to both voice and data services and a reporting period equal to one SACCH multiframe is considered by default.

2.4.4 Traffic Models

Voice and data services have very different traffic patterns, thus requiring elaborate traffic models adequate to their peculiarities. They define the activity of the MSs and how the traffic is generated.

Dynamic arrival and departure of MSs' voice calls or data sessions are considered in the system. The arrival process is modeled in the dynamic simulations through Poisson processes with specific arrival rates for each traffic type. In both voice and data traffic cases, the interval between consecutive arrivals of new MSs in the system is modeled by a negative exponentially distributed random variable [43].

In the modeling of the voice service, two aspects are taken into account: the duration of the call and the speech activity during a call. The first aspect is modeled through an exponential distribution with a 120 s mean. The activity model, however, depends on the use or not of discontinuous transmission (DTX). In the case in which DTX is disabled, the BS continuously transmits speech frames, even during the periods in which the speaker is silent, thus generating interference during the whole call. In the case in which DTX is used as an interference reduction mechanism, voice activity must be modeled.

A slow voice activity detector is considered, which fits well in to the global system for mobile communication (GSM) voice service whose minimum talking/silent periods are of one SACCH multiframe [9]. The adopted voice activity model considers a two-state Markov chain for simulating the transition between active and silent states [26].

The state transition probabilities from active-to-silent $P_{a\to s}$ and from silent-to-active $P_{s\to a}$ can be determined from the equations:

$$P_{a\to s} = 1 - \exp(-T_{rep}/T_a) \quad \text{and} \quad P_{s\to a} = 1 - \exp(-T_{rep}/T_s), \tag{2.6}$$

respectively, where T_{rep} represents the duration of a reporting period and T_a and T_s correspond to the mean duration of the active and silent stages. This leads to a mean voice activity of $T_a/(T_a + T_s)$. Typically, a mean voice activity of 60% is considered [22]. For the GSM/EDGE network considered in this chapter, T_{rep} corresponds by default to the duration of an SACCH multiframe and during silent periods DTX disables the transmission of voice frames, thus reducing interference in the system. Note that by adapting the values of T_{rep}, T_{a}, and T_{s} the presented voice traffic model can be easily employed in other wireless networks.

The data traffic model of the WWW interactive service is fairly different from that of speech, having a strong bursty traffic characteristic. The adopted WWW model considers the arrivals of packets within a session, the times between packets, and the packet lengths [32]. It is a simplified version of the WWW model presented in [53], which additionally considers packet calls. In [53], there are random variables modeling the number of packets per packet call and the inter-arrival time between such packets. The traffic model of [32] concentrates packets belonging to a packet call into a single large packet. Table 2.3 presents a summary of the parameters, distributions, and values of the WWW traffic model.

Table 2.3 world wide web (WWW) traffic model.

Parameter	Value
Sessions	
Distribution of the number of packet calls per session	Geometric
Mean number of packet calls per session	10
Packet calls	
Distribution of the reading time between packet calls	Truncated Pareto
Mean reading time between packet calls μ_{TP}	10 s
Pareto distribution parameters α_{TP}, k_{TP}, and m_{TP}	1.4, 3.45, and 120 s
Number of packets per packet call	1
Packets	
Distribution of packet sizes	Lognormal
Mean packet size	4,100 bytes
Standard deviation of packet size	30,000 bytes
Maximum packet size	100,000 bytes

2.4.5 Evaluation Metrics

Key performance indicators in wireless networks are mainly related to quality and capacity measures. Other measures, such as blocking rate and channel reallocation rate, may also be relevant in certain situations. The analyses presented in this chapter are focused on the relative performance of the different RRM techniques, rather than on absolute figures. In this section, some evaluation metrics and requirements are defined.

The voice quality is expressed in terms of the FER, which takes into account the percentage of lost speech frames with regard to the total number of speech frames.

The data quality, on the other hand, is defined as the mean packet bit rate r during the session of an MS.

The individual QoS criteria, which determine the satisfaction of each MS, are defined taking as reference the values presented in [24]. The following services are considered: enhanced full rate (EFR), multi-rate at 5.9 kbit/s with full rate (MR59FR), and World Wide Web (WWW). The QoS requirements are given by

$$\begin{cases} \text{Voice (EFR):} & FER^{\text{req}} \leq 1\%, \\ \text{Voice (MR59FR):} & FER^{\text{req}} \leq 0.6\%, \\ \text{Data (WWW):} & r^{\text{req}} \geq 10\ \text{kbit/s}. \end{cases} \tag{2.7}$$

The global QoS criteria of the system are defined as the percentual of satisfied users of each service. For voice a 95% satisfaction is required and for data either 95% or 90%, depending on the scenario.

Besides the QoS requirements, which apply to interference-limited systems, there is also the blocking criterion for voice. The blocking limit tolerated by the system is of 2%, where the blocking rate is defined as the relation between the number of blocked MSs and the number of births that occurred during the period for collecting the statistics.

The system capacity C is expressed in terms of spectral efficiency, i.e., the offered load per cell normalized by the amount of utilized spectrum. The spectral efficiency is expressed as $C = r_{\text{tot}}/(B_{\text{tot}} N_{\text{cell}})$, where r_{tot} indicates the total load offered to the system in bit/s, B_{tot} represents the frequency bandwidth available for traffic, and N_{cell} corresponds to the number of cells.

In the case of the voice service, the offered load is measured in Erlangs, while for the data service it is measured in terms of the transmission rate (bit/s). The spectral efficiency units are therefore measured in Erl/MHz/cell and bit/s/Hz/cell, for the voice and data services, respectively.

The network capacity limit $C^{\max}$ corresponds to the maximum load that can be offered to the system before the QoS or blocking requirements are violated. In some situations the capacity is also expressed in its normalized form, which consists of dividing the capacity value by the maximum service capacity, i.e., $C_{\text{norm}} = C/C^{\max}$.

Although the required QoS values introduced in this section are meant for a GSM/EDGE network, the same framework can be applied to other wireless networks with different QoS requirements.

2.5 RRM Performance in GSM/EDGE

In this section, the performance of different RRM techniques is evaluated in a GSM/EDGE network by means of dynamic system level simulations employing the models described in the previous section. It should be noted that the performance assessment through simulations involves simplifications, which are required in order to make this task mathematically and computationally tractable.

Due to such simplifications, the absolute performance results obtained through the modeling and simulation of wireless networks might differ from those observed in real systems. However, the relative analyses conducted by means of simulative studies are valid since all the model parameters are kept consistent across the different scenarios. Moreover, a significant part of the performance gains observed in simulation studies can be usually obtained in real systems. In this way, important insight on the performance of RRM strategies applied to wireless networks is obtained, which is of crucial strategic importance in decision-making processes involved in the design and operation of these systems.

Therefore, in this section it is important to focus on the relative performance gains obtained by the RRM techniques compared to reference scenarios. It is also worth mentioning that there exists extensive literature on RRM applied to GSM/EDGE and other wireless networks. It is not our intent to be exhaustive in covering the existing literature, so that we restrict ourselves to providing only key references concerning each topic.

2.5.1 Overall scenario

In this chapter, a GSM/EDGE network covering an urban macrocellular scenario is considered. Because frequency spectrum is a limited and expensive resource, tighter frequency reuse patterns like 1/3 and 1/1 are usually pursued in the GSM/EDGE networks. Tight frequency reuses allow for obtaining higher spectral efficiencies. They lead, however, to additional co-channel interference that must be efficiently handled by means of RRM techniques. To investigate the performance of voice and data services, simulations considering different RRM techniques have been done for cellular grids implementing 1/3 and 1/1 frequency reuses and the obtained spectral efficiency values have been compared with those obtained in reference scenarios. The most relevant parameter values are shortly summarized in Table 2.4.

A central carrier frequency $f_c = 2,000$ MHz and a BS height of 15 m above the rooftop of buildings are considered. A system bandwidth of 2.4 MHz is taken into account, which corresponds to 12 GSM carriers. RFH will be often employed and in each simulation a total number of 10,000 calls or sessions are simulated. In most of the considered cases, an interference-limited system is considered, i.e., noise power is negligible and SINR values are equivalent to SIR values.

In the system, a shadowing standard deviation σ_{sf} value of 6 dB, a shadowing decorrelation distance d_{sf} of 110 m, and an inter-BS shadowing correlation factor ρ_{sf} of 0.4 are considered [58].

2.5.2 Power Control

In this section, the performance of some of the PC algorithms discussed in Chapter 1 is evaluated in the downlink of a GSM/EDGE network. The voice service

Table 2.4 Simulation parameters.

Parameter	**Value**
	System
Cell type	Tri-sectored
Frequency reuse pattern	1/3 or 1/1
Frequency of operation	2,000 MHz
System bandwidth	2.4 MHz (12 GSM carriers of 200 kHz)
# of transceivers per sector	4 in frequency reuse 1/3, 12 in the frequency reuse 1/1
# of timeslots per carrier	8
Frequency hopping	Random or cyclic
Transmission direction	Downlink
Maximum transmit power	35 dBm
MS mobility	Random-walk pedestrian model (3 km/h avg. speed)
# of simulated calls or sessions	10,000
Satisfaction degree	95% of voice calls, 90 or 95% of WWW sessions
	Propagation effects
Average path loss	$128.15 + 37.60 \log(d)$ in dB, cf. (2.2)
Shadowing standard deviation	6 dB
Shadowing decorrelation distance	110 m
Inter-BS shadowing correlation factor	0.4
Fast fading	Considered at the link-level
	Services
Average voice call duration	120 s
Voice codec	EFR, MR59FR
Voice call average FER	1% with EFR, 0.6% with MR59FR
Voice blocking limit	2%
Discontinuous transmission	Enabled, disabled
Mean active voice period	1.1 s
Mean silent voice period	0.7 s
Mean voice activity	≈60%
WWW traffic	Modeled according to [32], cf. Section 2.4
Link adaptation	Enabled in ideal or non-ideal mode
Average session throughput	10 kbit/s

performance analysis considers two voice codecs, namely the EFR and the MR59FR. WWW traffic is considered as the data service. The system performance with and without PC is investigated for these services. In this section, a fraction of 95% of the MSs should be satisfied either for the voice or the WWW service.

The following PC algorithms are considered in this section: the one proposed in [21], which is termed here autonomous SINR balancing power control (ASBPC); the one proposed in [13, 57], which is termed here soft dropping power control (SDPC); and the up-down power control (UDPC) algorithm. All these algorithms have been previously discussed in Chapter 1.

In this section, the reported $\bar{\gamma}_s$ is represented by γ for simplicity of notation. The three considered PC algorithms are closed-loop algorithms based on the reported

SINR values. The algorithms share a similar mathematical structure, given by

$$P^{(k+1)} = P^{(k)} - c_{\mathrm{PC}} \varepsilon_\gamma^{(k)} \text{ in dBm}, \tag{2.8}$$

where $P^{(k)}$ represents the transmit power at iteration k, c_{PC} is a PC feedback constant, and $\varepsilon_\gamma^{(k)}$ is the feedback error signal of PC [28].

For all algorithms, the time delay compensation (TDC) technique discussed in Chapter 1 is employed to avoid instabilities due to measurement and report delays [28]. Considering the delay of one period associated with reporting the average SINR γ back to the BS, (2.8) becomes

$$P^{(k+1)} = P^{(k)} - c_{\mathrm{PC}} \varepsilon_\gamma^{(k-1)} \text{ in dBm}. \tag{2.9}$$

As a consequence, the power adjustment at iteration $k+1$ is performed based on outdated information, since the effect of the power adjustment associated with iteration k is captured only in the SINR made available at the BS at iteration $k+2$. In order to avoid this, the effect of the power adjustment done at iteration k is predicted and introduced in the feedback error signal under the assumption that the referred PC command was successful. Thus, considering TDC, the PC iteration becomes

$$P^{(k+1)} = P^{(k)} - c_{\mathrm{PC}} \varepsilon_\gamma^{k} + P^{(k)} - P^{(k-1)} \text{ in dBm}. \tag{2.10}$$

Considering TDC, the iterative formulation of the ASBPC, SDPC, and UDPC algorithms of Chapter 1 can be written as

$$p^{(k+1)} = p^{(k)} \left(1 - \beta_{\mathrm{F}} + \beta_{\mathrm{F}} \frac{\gamma_{\mathrm{t}} p^{(k-1)}}{\gamma^{(k)} p^{(k)}} \right) \text{ in W}, \tag{2.11}$$

$$P^{(k+1)} = P^{(k)} - \beta_{\mathrm{SD}} \left(\Gamma^{(k)} + P^{(k)} - P^{(k-1)} - \Gamma_{\mathrm{t}}(P^{(k)}) \right) \text{ in dBm, and} \tag{2.12}$$

$$P^{(k+1)} = P^{(k)} - \delta_{\mathrm{UD}} \operatorname{sign}(\Gamma^{(k)} + P^{(k)} - P^{(k-1)} - \Gamma_{\mathrm{t}}) \text{ in dBm}, \tag{2.13}$$

respectively.

For the ASBPC and SDPC algorithms, $\varepsilon_\gamma^{(k)}$ of (2.8) and (2.10) relates to the difference between the measured SINR γ and the target SINR γ_{t}, while the constant c_{PC} relates to the amount of the difference between measured and target SINRs that is compensated at each PC iteration. For the UDPC, $\varepsilon_\gamma^{(k)}$ relates to the sign of the difference between the measured and target SINRs, which is limited to either -1, 0, or $+1$, and the constant c_{PC} relates to the power step δ_{UD} introducing a fixed compensation of the feedback error signal. For more details on the formulation of power control iterations as control systems with feedback error signals refer to [28].

In particular for the voice service, several target SINR Γ_{t} values for the PC algorithms have been tested and the value resulting in the best spectral efficiency figures has been selected. For UDPC and ASBPC, Γ_{t} values 2 dB lower or higher than the selected ones resulted in worse system performance. The target SINR values considered in this section are higher than the 8 and 4 dB required with the EFR and

MR59FR codecs to achieve average FERs of 1 and 0.6%, respectively. Such differences are due to the particular models described in Section 2.4. In fact, an outer loop PC would be required in order to determine the most adequate target SINR for each scenario [29]. However, such an outer loop PC is not considered here.

In the SDPC case, a fixed target SINR is not specified, but a value for the parameter β_{SD} controlling the relationship between demanded power and target SINR. For the SDPC, β_{SD} value has been found experimentally, as suggested in [13], with $\beta_{SD} = 0.6$ providing the best results.

For the data service, a target SINR of 35 dB has been considered in order to avoid the concurrence between PC and LA. In order to investigate the impact of the actuation frequency of PC, different reporting periods have also been considered. Such parameter settings will be discussed in more detail together with the results obtained in the corresponding cases. The most relevant power control parameters are listed in Table 2.5.

Table 2.5 Power control parameters.

Parameter	Value
PC algorithms	ASBPC, SDPC, and UDPC
Maximum transmit power	35 dBm
Minimum transmit power	5 dBm
Power levels	Discrete in steps δ_{UD} of 2 dB
Power control time-step	1 iteration at each 480, 120, or 20 ms
Time delay compensation (TDC)	Enabled
β_F for ASBPC	1 (fastest convergence) [21, 28]
β_{SD} for SDPC	0.6 (defined experimentally, as in [13])
Minimum target SINR $\Gamma_{t,min}$ for SDPC	8 dB for EFR, 4 dB for MR59FR, 6 dB for WWW

In this section, scenarios implementing tight frequency reuses of 1/3 and 1/1 are considered and PC is employed to manage the co-channel interference and improve the system performance. RFH and pedestrian mobility, cf. Table 2.4, are considered.

2.5.2.1 Power Control Performance for the Voice Service

Initially, the performance of PC for the voice service considering the EFR codec is investigated. In order to determine the system capacity, simulations with increasing offered loads have been conducted until reaching the QoS limits given in Table 2.4. Figure 2.7 presents the percentual of satisfied MSs as a function of the voice spectral efficiency in Erl/MHz/cell considering the EFR codec and the 1/3 frequency reuse. In the cases in which PC is applied, DTX is also enabled.

In Fig. 2.7, the capacity limits by interference and by blocking are also shown. The system capacity is limited by interference whenever the blocking rate is below 2% but the fraction of satisfied MSs is lower than 95%, which corresponds to having more than 5% of the MSs perceiving average FER higher than the values specified

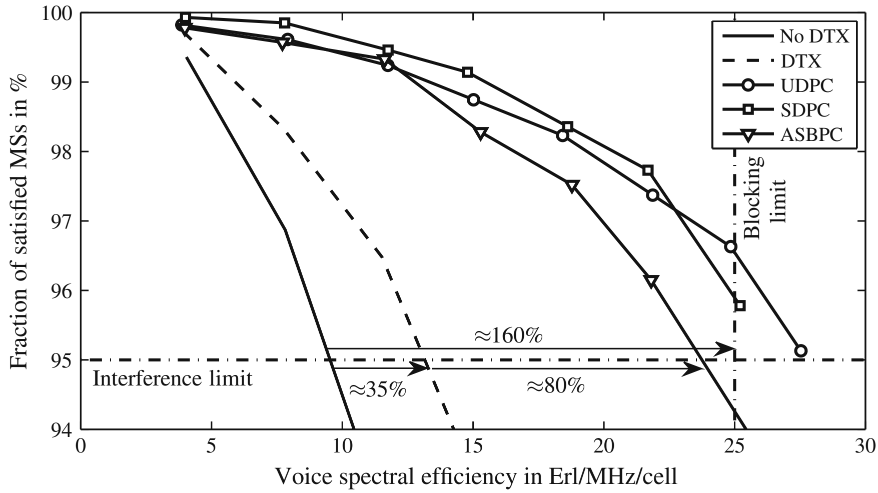


Fig. 2.7 PC performance with EFR voice codec and reuse 1/3 ($\Gamma_t = 16$ dB for UDPC and ASBPC).

in Table 2.4. Oppositely, the system capacity is limited by blocking if more than 2% of the arriving calls are blocked due to unavailability of free channels while the ongoing calls perceive acceptable average FER values. Given the number of channels per sector and the target blocking rate, the blocking limit of the system can be easily calculated using standard traffic engineering methods [44, 56]. The capacity limits by blocking and interference are drawn for all figures in this section, but are labeled only in Fig. 2.7.

In Fig. 2.7, it can be seen that the system capacity without PC and DTX is considerably lower than when these two features are enabled. Indeed, by enabling DTX a considerable amount of unnecessary interference is eliminated from the system and a capacity gain of more than 35% is obtained. By using ASBPC, a voice spectral efficiency of more than 23.5 Erl/MHz/cell is obtained, which represents an additional gain superior to 80% compared to the case with DTX only. Compared to the case in which both PC and DTX are disabled, the obtained spectral efficiency is about 1.5 times higher, which shows that ASBPC can substantially improve the system capacity. Such a high-capacity improvement comes from the reduction of the co-channel interference performed by the PC algorithms, which employ only as much power as required to attain the target QoS levels.

In spite of being able to perform larger power adjustments, the ASBPC provided lower spectral efficiency figures than the UDPC. This result might be explained by the selection of $\beta_F = 1$, which according to [21, 28] results in the fastest convergence. However, this value of β_F is not necessarily optimal and might lead to instability of the ASBPC algorithm, as discussed in [28]. Thus, it is possible that lower values of β_F could lead to better spectral efficiencies. Anyway, the optimization of the parameter β_F has not been performed for the ASBPC algorithm. In this case, the UDPC works in a more stable fashion leading to higher spectral efficiency values.

Indeed, it performs as good as the SDPC algorithm, which is allowed to make larger power adjustments like the ASBPC but used a more conservative value for β_{SD} and involves a self-regulation of the target SINR [13, 28, 57]. The UDPC and SDPC algorithms outperform here the ASBPC algorithm providing spectral efficiency gains higher than 160% compared to the case without PC and DTX. As it can be seen in Fig. 2.7, the UDPC and SDPC algorithms reach the blocking limit of 2% while there are more than 95% of satisfied MSs in the system, i.e., in the considered scenario PC turns the system into a blocking limited system.

Because the system capacity became limited by blocking, voice calls perceive QoS levels higher than the target ones. This suggests tightening the frequency reuse in order to make more channels available per sector and to potentially support higher number of voice calls. The frequency reuse tightening increases the number of available channels per sector, but it incurs in a considerable increase of co-channel interference due to the smaller reuse distance. In Fig. 2.8, the fraction of satisfied MSs against the voice spectral efficiency is shown for a GSM/EDGE network implementing a 1/1 frequency reuse and considering the EFR codec.

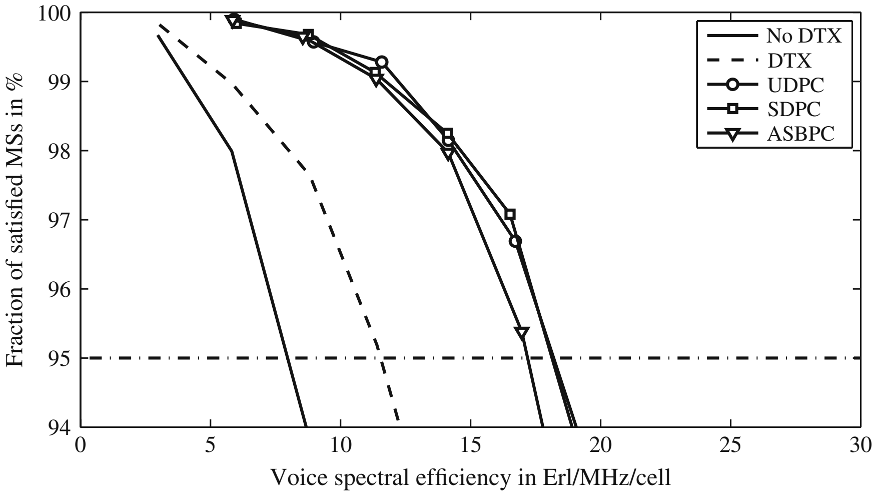


Fig. 2.8 PC performance with EFR voice codec and reuse 1/1 ($\Gamma_t = 18$ dB for UDPC and ASBPC).

It can be seen in Fig. 2.8 that the frequency reuse tightening results in a too large increase of the average co-channel interference and causes an overall reduction of the system spectral efficiency varying between 15 and 25%. The system might support higher voice loads if more robust voice codecs, such as the MR59FR, are used by the MSs. Such a codec allows the system to operate with acceptable voice quality at considerably lower SINR levels.

In the following, the spectral efficiency of the system considering the MR59FR codec is evaluated.

Figure 2.9(a) shows the fraction of satisfied MSs against the achieved voice spectral efficiency considering the MR59FR codec and a 1/3 frequency reuse. In this figure, it can be seen that considerably higher voice loads are supported in the system and, consequently, higher spectral efficiency values are achieved. Moreover, it can be seen that the use of DTX without PC is already sufficient to bring the system to a capacity limitation by blocking. Compared to the case without DTX and PC, the obtained spectral efficiency gains are of about 25%. Using PC in this scenario would only improve the QoS levels perceived by the MSs resulting in average FER per call considerably lower than the target values given in Table 2.4. Such a case is shown in Fig. 2.9(a) for the UDPC algorithm only, which provides a satisfaction level much higher than that achieved with DTX and no PC with more than 99% of satisfied MSs. In this case, a tightening of the frequency reuse might also be advisable. The results, analog to Fig. 2.8 but considering the MR59FR codec, are shown in Fig. 2.9(b).

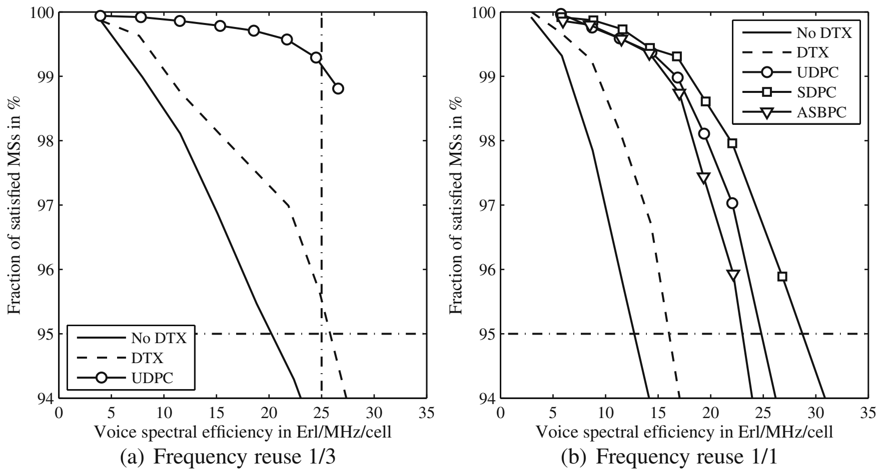


Fig. 2.9 PC performance with MR59FR codec.

Differently from Fig. 2.8, where a tightening of the frequency reuse resulted in too much interference to be managed, in Fig. 2.9(b) the use of a more robust codec allows a larger amount of interference to be supported. In Fig. 2.9(b) the spectral efficiency value achieved with the ASBPC at the capacity limit is almost the same as that achieved in the 1/3 frequency reuse with the EFR codec, in spite of the additional interference of the 1/1 frequency reuse. The same is valid for the UDPC, which achieves about 25 Erl/MHz/cell as in Fig. 2.7. In Fig. 2.9(b), it can be seen that SDPC outperforms the UDPC algorithm and achieves a spectral efficiency value of about 28.5 Erl/MHz/cell. Thus, a spectral efficiency gain of about 15% is obtained with acceptable QoS for voice calls. Moreover, no blocking has been verified since a larger number of channels are available in the 1/1 scenarios.

Additional capacity gains can be obtained by increasing the actuation frequency of the PC. In order to do this, alternative schemes have been proposed in [2, 52], which suggest to modify the PC signaling mechanisms in order to improve PC performance. The first approach consists of modifying the SACCH channel structure so that PC commands and link quality measurements could be transmitted in each SACCH burst, i.e., at each 120 ms. This corresponds to multiplying the rate of the power adjustments by 4. A second approach is to transmit PC commands and link quality measurements in-band, resulting in a 24-fold improvement in the PC actuation frequency. Nevertheless, it is worth mentioning that both strategies might reduce the protection of the control information sent over the SACCH and TCH channels, since some more bits would be allocated for PC purposes [52]. Even considering such improvements, the PC actuation frequency in the GSM/EDGE network is still too low to compensate for fast fading, especially for fast moving MSs. As a comparison, the highest PC actuation frequency, i.e., 50 Hz considering one iteration at each 20 ms, is still 30 times lower than that considered in WCDMA systems, which corresponds to 1.5 kHz. According to [29], even when operating at such frequency, wideband code division multiple-access (WCDMA) fast PC is not able to perfectly compensate for fast fading of MSs with speeds superior to 50 km/h.

Table 2.6 shows the performance of the SDPC algorithm considering different reporting periods, i.e., different actuation frequencies for the PC. SDPC is considered because it performed as good as or outperformed the ASBPC and UDPC algorithms. A frequency reuse of 1/3 is used with the EFR service because better spectral efficiency values have been achieved in this case. For the same reason, the 1/1 frequency reuse is selected when considering the MR59FR codec.

Table 2.6 Performance of the SDPC algorithm with reduced reporting periods (higher actuation frequency).

	EFR, 1/3			MR59FR, 1/1		
Reporting period in ms	480	120	20	480	120	20
Capacity in Erl/MHz	25	25	25	28.5	35	31.3
% of satisfied MSs	96	98	98	95	95	95

In Table 2.6, it can be seen that either QoS (with EFR) or capacity gains (with MR59FR) can be obtained by increasing the actuation frequency of the PC. For the MR59FR, a spectral efficiency gain of about 20% is achieved by performing PC adjustments at each 120 ms instead of at each 480 ms. However, adjusting transmit powers at each 20 ms does not enhance spectral efficiency. The reason for the latter result is that an increase of the actuation frequency leads to a higher variance of the interference in the system, which is harder to be tracked and compensated by the PC. Comparing the 35 Erl/MHz/cell supported in the 1/1 frequency reuse with MR59FR and power adjustments at each 120 ms, a spectral efficiency gain superior to 170% is obtained compared to the scenario without PC and DTX.

2.5.2.2 Power Control Performance for the WWW Service

In the GSM/EDGE network, the LA works on the same time basis as PC, i.e., with one adjustment at each 480 ms. Such adjustments can be made based on the reported average MEAN_BEP and average CV_BEP measured during one SACCH frame. Similarly to the voice service, a delay of one reporting period is involved before measurements become available at the BS.

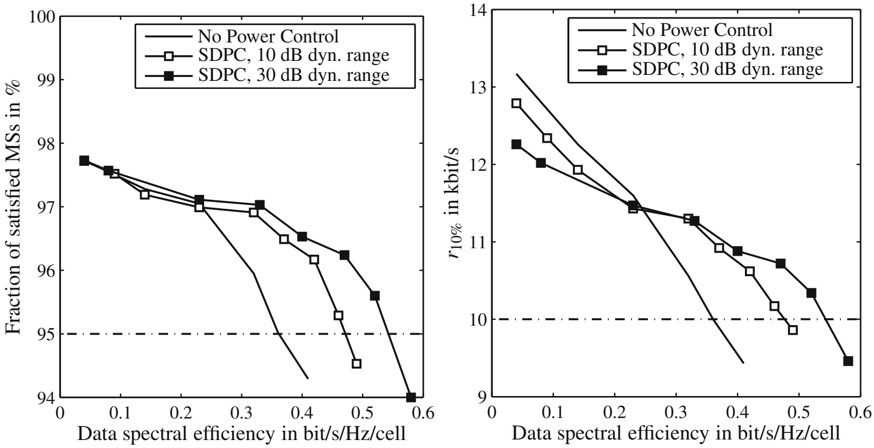
If the target SINR of the PC algorithm is set outside of the range of SINR values covered by the LA, the performance of PC is strongly limited and no considerable capacity gain is obtained through PC, as it has been shown in [37]. This occurs mainly because PC will only apply for a few connections, whose link quality is really high, for instance, with SINR values above 35 dB. In [37], it has also been shown that an increase in the actuation frequency of LA leads to capacity gains of about 20%, which are similar to those previously shown for the voice service when increasing the actuation frequency of PC. Anyway, PC control does not provide any considerable capacity gain in this case too.

PC can provide capacity gains to the WWW service if the dynamic power range is extended from 10 to 30 dB and if the target SINR of the PC algorithm is allowed to lie inside the range covered by the LA, i.e., if the prioritization of LA over PC and the minimum transmit power constraint for the reliable reception of the USF are disabled.

In order to show this, the SDPC algorithm is considered in the sequel for the WWW service. A minimum target SINR of 6 dB has been considered, which corresponds to the QoS requirement of 10 kbit/s according to the link-level curves in Fig. 2.3. As before, the value 0.6 has been used for the parameter β_{SD}. In order to reduce the impact of the scheduling algorithm over the PC, the well-known first-in-first-served (FIFS) scheduling discipline is used, which maintains a channel allocated to the same MS for the total time needed to transmit its current packet. Considering these assumptions, Fig. 2.10 shows the capacity and QoS gains achieved with PC for the WWW service.

In Fig. 2.10(a), the fraction of satisfied MSs against the data spectral efficiency in bit/s/Hz/cell is shown. As it can be noted, PC can provide spectral efficiency gains of about 30% if a 10 dB dynamic power range is assumed and the prioritization of LA over PC is ignored. Additional gains of 15% are obtained by increasing the dynamic power range from 10 to 30 dB.

Figure 2.10(b) shows the 10th percentile of the MSs' average packet throughput, denoted by $r_{10\%}$, against the data spectral efficiency. In this figure, it can be seen that for low offered loads the average throughput achieved by 90% of the MSs is higher when PC is not used. Oppositely, for higher loads the average throughput perceived by 90% of the MSs is higher than when PC is disabled. This effect is more accentuated for a 30 dB dynamic power range. Consequently, the curves for the average throughput of 90% of the MSs with and without PC cross each other. The reasons for this crossing are as follows.



(a) Fraction of satisfied MSs (b) Average packet throughput

Fig. 2.10 PC performance for the WWW service in frequency reuse 1/3.

For lower loads, a smaller number of data sessions are present in the system and LA maximizes the throughput of each connection providing high rates, while PC reduces the rates of the MSs to attain only the target values. Because for low offered loads there are enough resources (timeslots) to serve the connections without incurring excessive co-channel interference, as it can be seen in Fig. 2.10(a), LA can operate at peak rates, and the rate reductions promoted by PC in order to attain the target SINRs could be eventually seen as unnecessary. In the truth, an outer loop PC should ideally be employed in this case to adjust the target SINR of the MSs and give them the higher rates that they could achieve without compromising system performance, i.e., without leading to excessive co-channel interference. When the load increases, co-channel interference increases and, in order to keep it under control, PC reduces transmit powers and consequently lowers the rates of several connections, which does not occur when considering LA without PC. The increased co-channel interference levels are efficiently combated by PC so that higher offered loads can be supported while maintaining the QoS levels at acceptable values and, consequently, better throughput values are attained by the MSs. This result is even better when considering a wider dynamic power range, i.e., 30 dB.

In total, the capacity gains provided by PC for the WWW service are modest (about 50% only) compared to those achieved for the voice service. Moreover, these gains are only achieved by violating some restrictions imposed by the standard, which suggests that PC is not very efficient for interactive data services. This is due to several aspects such as the bursty nature of data services, the MSs multiplexing performed by scheduling algorithms, and the competition between LA and PC, among other reasons.

2.5.3 Dynamic Channel Allocation

In this section, the different stages of the dynamic channel allocation algorithm are evaluated taking into account both ideal situations as well as those for which practical restrictions limit the efficiency of the algorithm.

The evaluation of DCA has been performed considering a GSM/EDGE cellular system in a macrocellular environment employing a 1/1 frequency reuse pattern. Such reuse pattern implies that all hopping frequencies are available within each sector, which corresponds to a scenario adequate to the application of DCA, since there exists the flexibility of distributing the channels freely among the cells. The voice service is considered in the evaluations using the EFR codec and considering DTX. In order to assess the impact of DCA on the system performance, scenarios employing either RFH or DCA combined with CFH are considered. RFH and CFH have been described in Section 2.2.3.

The evaluation begins with the simulation of the DCA algorithm under ideal circumstances (DCA ideal), i.e., with perfect SIR estimation. Some admission control alternatives are also considered. Finally, practical restrictions are introduced and some solutions for improving DCA performance are evaluated.

Figure 2.11 shows the performance of the DCA ideal algorithm with regard to RFH, in terms of quality of service and blocking rate. The value in dB represents the minimum SIR criterion of the admission control, and the term "impact" (DCA impact) refers to the impact test described in Section 2.3.2 with a 10 dB threshold.

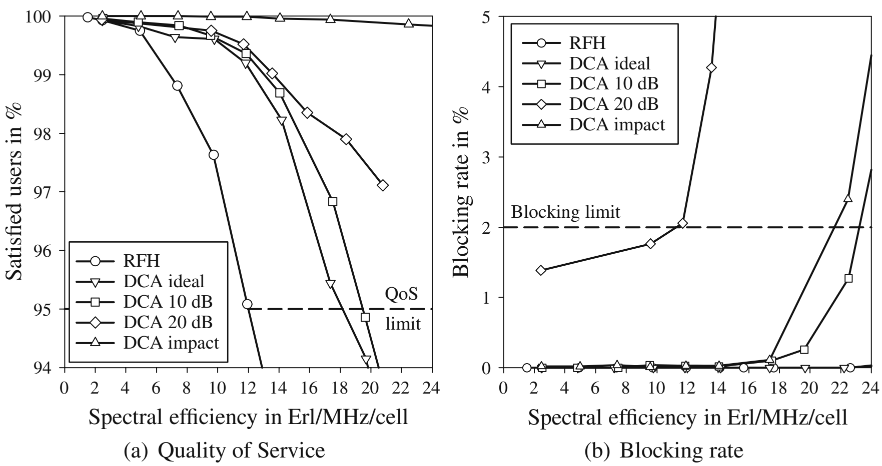


Fig. 2.11 Performance of ideal DCA and admission control.

Observing Fig. 2.11(a), it can be seen that the use of DCA provides significant capacity gains when compared to RFH. By applying the admission control with 10 and 20 dB thresholds, which indicate that users with SIRs below these levels are blocked, the quality of service is additionally improved. An even higher gain is provided by the impact test.

The admission control algorithm, even though presenting better quality of service to the MSs within the system, has its negative side, which is to increase the blocking error rate. This behavior can be seen in Fig. 2.11(b), where the presented metric includes the blocking of incoming calls as well as the dropping of ongoing calls. For a 10 dB threshold the blocking rate is significantly increased when compared to the case without admission control, but the interference still limits the offered load. For a 20 dB threshold, blocking becomes excessive, limiting capacity too early. Since, according to link-level results, an 8 dB SIR would be enough to guarantee minimum quality, the use of a threshold with such a high margin is not recommended. The best situation for the admission control would be the one in which the interference and blocking limits would meet at the same load, which might occur for a threshold slightly above 10 dB.

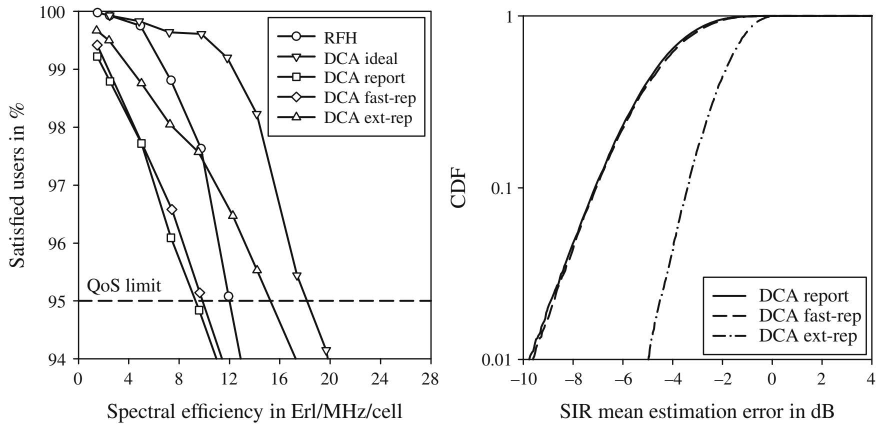
The impact test provides excellent QoS levels, almost reaching 100% satisfaction even for high offered loads. The blocking rate limits capacity at about 22 Erl/MHz/cell, which is still better than the other strategies. A threshold lower than 10 dB could provide better results, in the sense of equilibrating interference and blocking. It is important to emphasize, however, that this impact test is highly idealized, being its practical implementation probably not feasible after all restrictions are taken into account.

The measurement report mechanism, described in Section 2.3.2, is also used for performing SIR estimation in a practical system. As previously mentioned, two aspects might contribute to the imprecision of the estimates: the report delay and the number of reported cells.

Figure 2.12(a) shows how the DCA performance is compromised when assuming non-ideal SIR estimation (see the DCA report curve). The system capacity with dynamic channel allocation becomes inferior to the case with RFH, dropping from approximately 18 to 9 Erl/MHz/cell. The distribution (in logarithmic scale) of the mean SIR estimation error per user can be seen in Fig. 2.12(b), where the error is defined as the difference between the actual value and the estimated value. Since all values are negative, it may be concluded that the SIR is being overestimated.

The impact of faster report updates on the DCA performance is depicted by the DCA fast-rep curve, which assumes a delay corresponding to half the actual delay perceived within the system. It can be seen that the reduction of the delay has only a slight impact on the system capacity and that the distribution of the SIR estimation error was practically unaltered. Since the DCA algorithm employs cyclic hopping, which does not provide interference diversity after each hop, the interference profile perceived by the users barely changes within a measurement reporting period.

The use of extended reports, i.e., with a larger number of reported cells, presented good capacity results, as it can be seen in Fig. 2.12(a). By using measurement reports with 10 reported BSs (DCA ext-rep) instead of 6, the DCA capacity once again surpassed that of RFH, approaching the ideal performance and reaching a spectral efficiency of 15 Erl/MHz/cell. The improved DCA performance is due to the more



(a) Quality of Service

(b) Error cumulative distribution

Fig. 2.12 Impact of the measurement reports on the DCA performance

precise SIR estimates (see Fig. 2.12(b)), which are achieved by the larger number of reported BSs, since more interferers are taken into account.

Table 2.7 presents a summary of the spectral efficiency achieved by the different algorithms. Note that the capacity is limited by interference for almost all cases, except for those marked with *, which are limited by blocking.

Table 2.7 Capacity results of the different algorithms.

Strategy	Spectral efficiency	Strategy	Spectral efficiency
RFH	12.0	DCA ideal	18.2
DCA report	9.3	DCA 10 dB	19.5
DCA fast-rep	9.8	DCA 20 dB	11.3*
DCA ext-rep	15.3	DCA impact	21.6*

The DCA algorithm is also analyzed for data services. Since the channels may be shared by several users, the selection procedure first prioritizes the empty channels with the best SIR, in order to avoid excessive queuing delays. If there are no empty channels, the selection is based exclusively on the SIR criterion. The ideal DCA algorithm has been assumed and a data user is deallocated if there is currently no other packet at the BS to be transmitted, which means that several channel reassignments may take place during the session of a data user.

Figure 2.13 shows the performance of RFH and DCA for a WWW service considering a FIFS scheduling discipline. The QoS is expressed in terms of the previously described $r_{10\%}$ rate metric. It can be seen that DCA provides significant capacity gains, reaching almost double the capacity of RFH, which demonstrates the potential of DCA for data services. The gains are mainly due to the bursty nature of WWW, which leads to high reassignment rates and allows the DCA algorithm to act quite often. Notice that lower gains are expected if practical constraints are taken into account, such as measurement reports and channel release timers, as well as in the case of non-bursty data traffic, such as download.

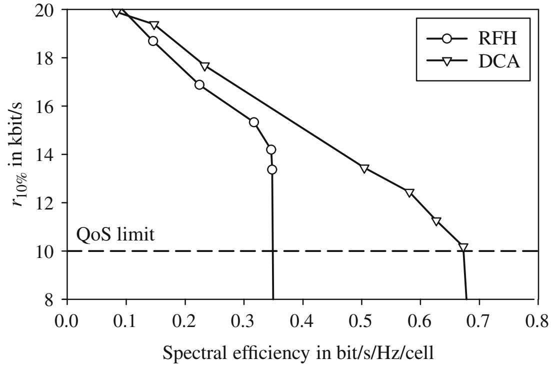


Fig. 2.13 DCA performance for the WWW data service.

2.5.4 Multi-antenna Techniques for Multiple Services

In order to provide capacity enhancements for GSM/EDGE networks with a mix of a voice and a data service, this section evaluates three strategies concerning the application of multi-antenna techniques, which are single-antenna transmission (SAT) with both services using a single sector antenna; single- and multi-antenna transmission (SMT) with a single sector antenna applied to the data service and switched fixed beams applied to voice; and multi-antenna transmission (MAT) with both services using switched fixed beams.[2.4]

For the cases in which switched fixed beams are applied, four-element uniform linear antenna arrays are considered for each sector of the BSs. In a sector, the antenna array is configured to steer four narrow beams covering the sector area. The voice service using the EFR codec and the WWW service are considered. For the voice service, DTX and PC are applied. For the WWW service, IR is employed considering only the MCS-9 of EGPRS.

The performance of the individual services may be seen in Fig. 2.14. The parameters are defined as voice service spectral efficiency (C_{V}), data service spectral efficiency (C_{D}), and the maximum offered spectral efficiency supported by the voice ($C_{\mathrm{V,SA}}^{\max}$), and data ($C_{\mathrm{D,SA}}^{\max}$) services with a single antenna.

It can be seen that the antenna arrays provide significant capacity gains, due to their spatial filtering and interference reduction characteristics. The capacity gains

[2.4] Note that MAT disregards the USF issue. A possible solution for a multi-antenna EGPRS system is the beam-based scheduling algorithm proposed in [12].

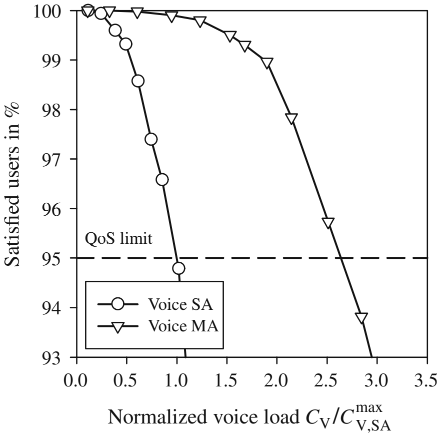


Fig. 2.14 Performance of the individual services: voice (left) and data (right).

may also be seen as quality of service gains, in case it is preferred to maintain the same load and offer better quality to the MSs, which would be adequate to the data service, since it would lead to higher transmission rates.

Figure 2.15 shows the values of the mean interference per MS for each service, with a single antenna (SA) and multiple antennas (MA). The mean interference is measured from the simulation of the individual services. Since we are simulating an interference-limited network that employs RFH, it may be assumed that the interference distribution will not significantly vary with the service mix. Therefore, out of simplicity, the individual services are taken as reference for calculating the power offset, corresponding to a mix of 100% of one service and 0% of the others.

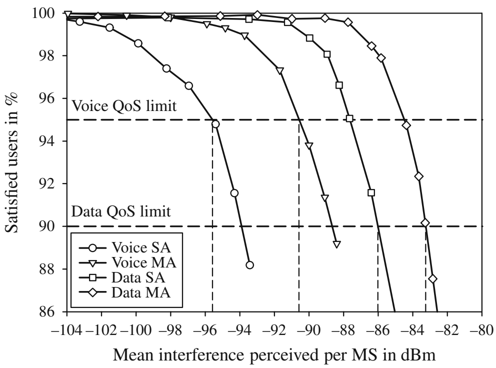


Fig. 2.15 Maximum interference supported by each service.

The application of multiple antennas results in an increase of the maximum interference supported by the services. The reason for this behavior might be the higher interference diversity introduced by the switched fixed beams.

As it has been discussed in Section 2.3.3, the system capacity is maximized when the maximum interference levels are balanced, which may be accomplished by adjusting the transmission power of the service which is more robust to interference (SBPS technique). The determination of the power offset may be done based on the interference levels supported by each service. From Fig. 2.15, a rough estimate of the power offset of each service/antenna configuration may be calculated as the absolute difference between the maximum interference levels, e.g., the difference between voice SA and data SA for the SAT configuration. This leads to offsets of approximately 9 dB (SAT), 4 dB (single and multi-antenna transmission (SMT)), and 6 dB (MAT). Note that the actual offset obtained from Fig. 2.15 for the MAT case is of roughly 7 dB, but it has been verified through simulations that the ideal offset for maximizing capacity is about 1 dB lower.

Next, the joint simulation of voice and data assumes that the frequency spectrum is shared among the services. The antenna strategies are initially evaluated for fixed voice load scenarios, in order to assess the achievable data capacities.

Figure 2.16 shows the data performance of the system for a scenario in which the normalized voice load is fixed at 0.75, i.e., 75% of the total supported voice load. The graphic is presented in terms of quality of service as a function of the data load. For each antenna strategy, quality curves are presented for voice (VQoS) and data (DQoS). The points at which the QoS requirements are met are indicated as well.

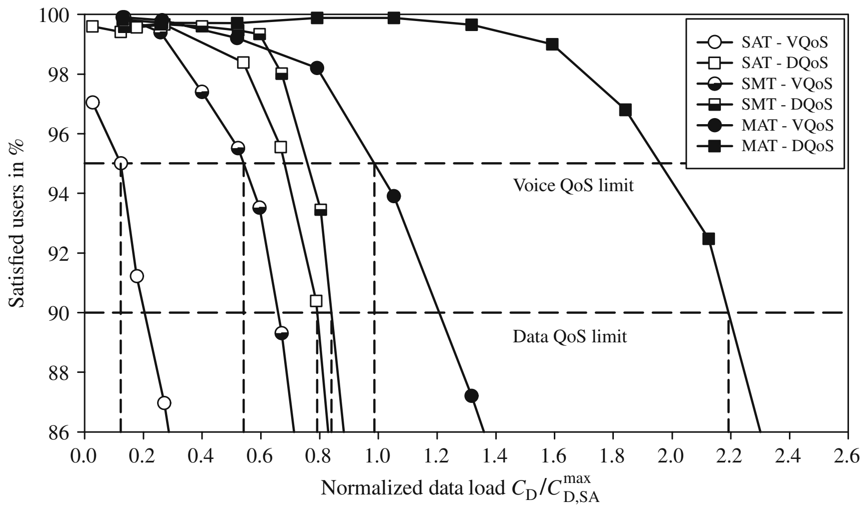


Fig. 2.16 Data service capacity for the different antenna strategies.

For the case of the strategy employing only single-antenna transmission (SAT), it can be seen that the voice service limits the data capacity right after the introduction of a few MSs, resulting in a capacity of less than 20% of the load supported by the data service individually.

The application of multi-antenna transmission only for the voice service (SMT) provides a reasonable increase in terms of data capacity, since the random frequency hopping algorithm allows the interference reduction of switched fixed beams to be perceived by all services. The multi-antenna transmission for both services (MAT) also provides improvements on the data capacity, as it can be seen in Fig. 2.16.

For all considered strategies, the voice service always limits system capacity, due to its more restrictive QoS requirements. The data service, which has less restrictive QoS requirements than voice, would be able to support higher offered loads if the voice restriction could be disregarded. This indicates that the system is unbalanced and that the SBPS concept may be applied.

The application of the power offset to each of the antenna strategies presents quite satisfactory results, as it can be seen in Fig. 2.17. The total system capacity considerably increases, and the QoS requirements are reached for similar offered loads, confirming the initial expectations regarding SBPS.

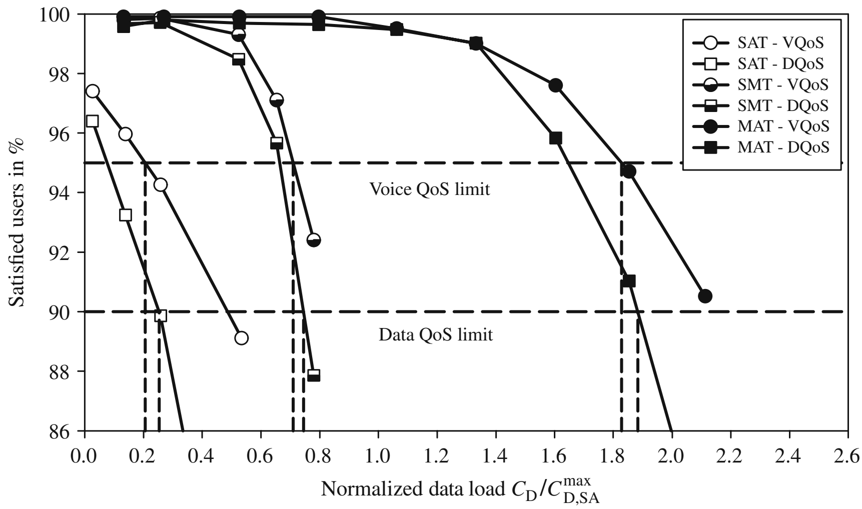


Fig. 2.17 Application of the power offset for the different antenna strategies.

Next, both voice and data loads are jointly varied for each service mix in order to obtain the capacity regions. The capacity region corresponds to the area within a data load vs. voice load graphic, for which the QoS requirements of both services are satisfied. Combinations of voice and data outside this region are not supported by the system. The left hand side of Fig. 2.18 shows the capacity regions – both simulated and theoretical – for the evaluated antenna strategies.

The extreme points correspond to the individual capacities of the isolated services. In [22] it has been shown that, for interference-limited scenarios, the capacity region of a balanced system is linear, i.e., it consists of a straight line connecting the individual capacity extremes. These regions are indicated as (Theory - SBPS).

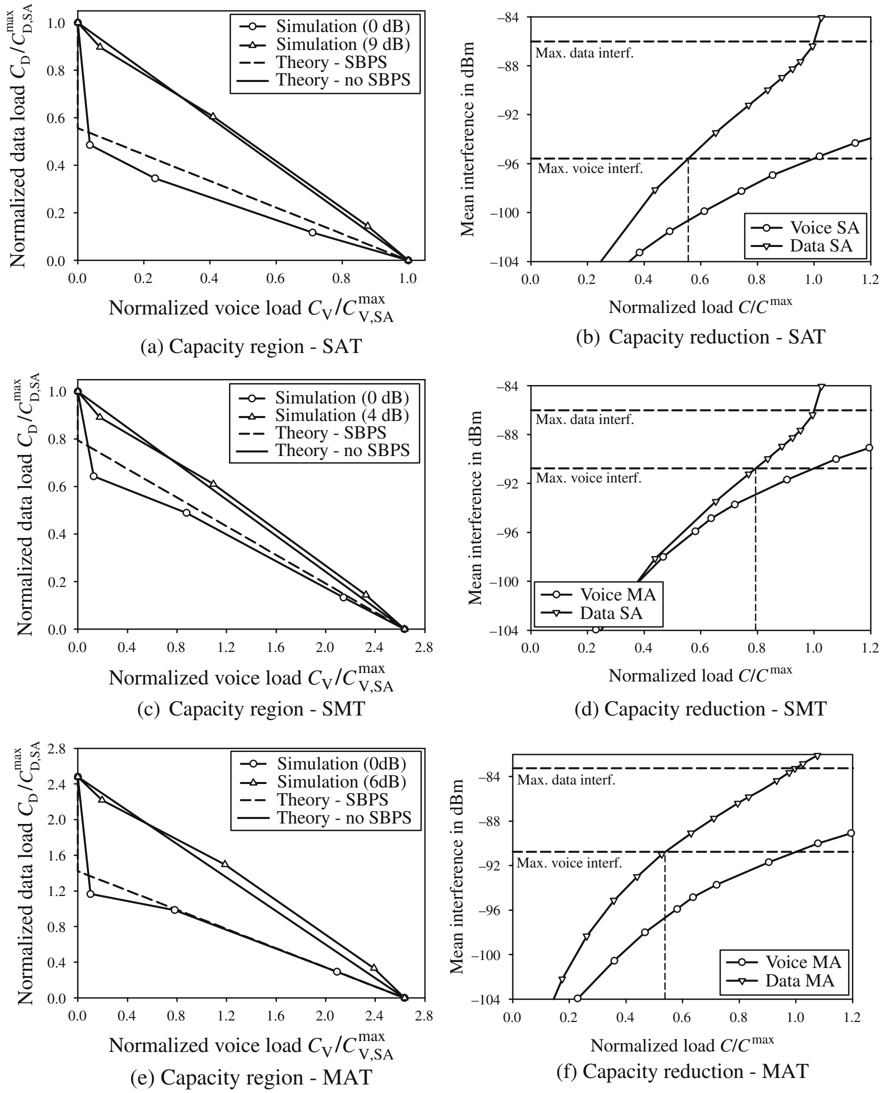


Fig. 2.18 Capacity regions (*left-hand side*) and capacity reduction (*right-hand side*) for the mix of voice and data services.

It can be seen that the application of the previously estimated power offsets has been fairly efficient, approximating the theoretical capacity regions for all cases. This result confirms that the application of antenna arrays, for one or more services, requires different power offsets in order to perform the interference balancing.

The system capacity prior to the SBPS application, mainly for the MAT case, has a much worse performance than the balanced system. As soon as a few voice users are introduced, the data capacity suffers a sudden capacity drop, and from then on it starts to decrease linearly with the increase in voice capacity.

This capacity drop is caused by the unbalanced interference distribution. Since the data service supports interference levels much higher than the voice service, the introduction of the latter in the system requires that the interference be decreased until the point at which it becomes possible to offer the minimum QoS requirements for voice. The larger the difference between the maximum interference supported by the services, the larger the sudden capacity drop effect.

The theoretical curves of the unbalanced system may be obtained from the results that relate load and interference, such as the ones shown on the right-hand side of Fig. 2.18. In these results, the load is normalized per service, i.e., the unit load corresponds to the maximum load supported by voice as well as by data. The maximum interference limits are those determined based on the quality requirements of each service (see Fig. 2.15).

Since the voice service is the most sensitive to interference, the crossing of the data curve with the maximum voice interference indicates how much the data load has to be reduced in order to satisfy the interference requirements of the voice service. The theoretical curve (Theory - no SBPS) may then be drawn as the maximum data capacity dropping until the estimated reduction point, followed by a straight line connecting it to the maximum voice load point.

Figure 2.19 shows the simulated capacity regions for each antenna strategy in a same graphic, with and without the application of SBPS. These figures provide a clearer view of the performance of the antenna strategies for multiple services. The application of the SMT technique, besides providing significant gains for the voice service itself, also provides gains to the data service. The capacity region area is expanded and certain voice load situations, for which it was previously prohibitive to add data users, now have a larger tolerance for data users. The application of MAT, on the other hand, results in a significant increase of the whole capacity region of the system making it possible to support high offered loads for both services.

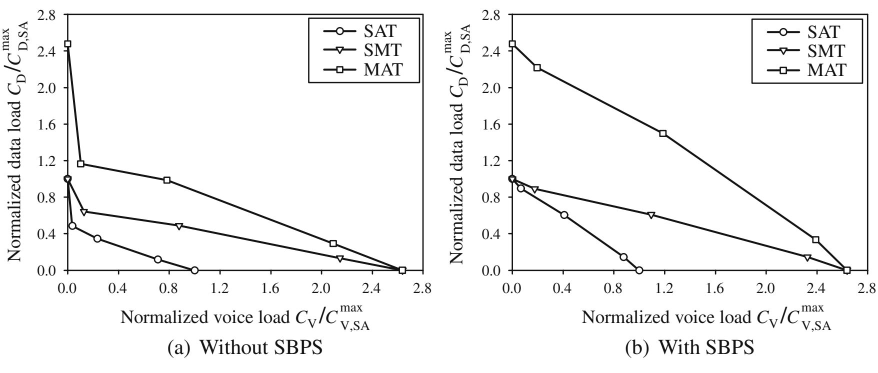


Fig. 2.19 Capacity regions for the different antenna strategies.

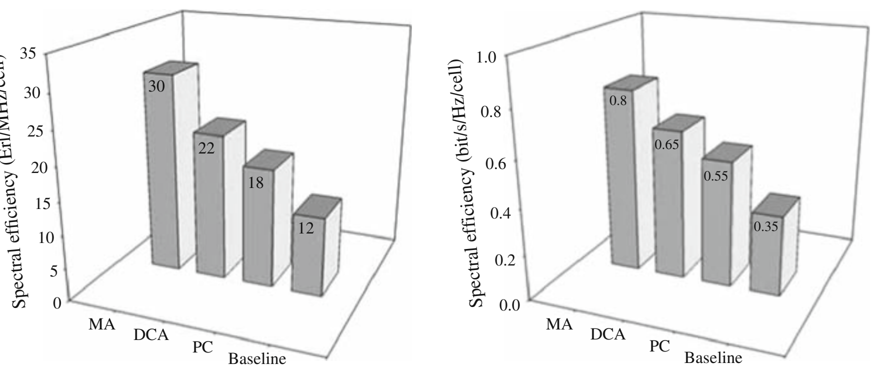
The application of SMT thus represents a sensible first step toward higher data capacities, especially when the system is still dominated by voice users.

2.6 Conclusions and Research Directions

This chapter has shown how the efficient application of adequate RRM techniques can provide significant gains to the GSM/EDGE system. The techniques were employed to different services and considered different scenarios.

The PC technique is an efficient method for improving the capacity of both voice and data services. Due to the distributed nature of the considered algorithms, low implementation complexity is required and they may be implemented using the standard measures available on GSM/EDGE DCA, on the other hand, requires a certain cooperation among BSs in order to achieve reliable SIR estimates of the available channels. The increased complexity of DCA results in significant gains with regard to standard mechanisms such as random hopping, especially when applied in combination with admission control. Finally, the multi-antenna techniques, which require the deployment of additional hardware to the BSs, present the most promising capacity improvements, for both single and multi-service scenarios.

In Fig. 2.20, a summary of the spectral efficiencies achieved by the different RRM techniques is presented for both voice (EFR) and data (WWW) services considering a 1/1 frequency reuse. The best algorithm of each RRM technique is taken into account and the baseline corresponds to a simple random hopping scenario. It can be seen that the proposed techniques can provide considerably higher spectral efficiencies than the baseline case. The multi-antenna (MA) strategy provides the highest gains for both voice and data services, followed by the DCA and PC algorithms.



(a) Voice (EFR) service (b) Data (WWW) service

Fig. 2.20 Spectral efficiency achieved by the RRM algorithms assuming a 1/1 reuse.

A rough comparison of GSM/EDGE with a baseline high-speed downlink packet access (HSDPA) system with 5 MHz bandwidth, which is shown in Chapter 3 to be of about 0.7 bit/s/Hz/cell, indicates that RRM can potentially boost the GSM/EDGE capacity to approach and even surpass that of HSDPA. Notice, however, that this comparison is just illustrative of the potential capacity of GSM/EDGE with advanced features, as the HSDPA capacity figure does not consider similar improvements.

The further evolution of GSM/EDGE, as pointed out by [14], could include dual-antenna terminals, multi-carrier data services, mobility enhancements, high-order modulations, among other features. The evolved EDGE technology is expected to reach peak theoretical user data rates of 1.2 Mbit/s in the downlink and 473 kbit/s in the uplink [1]. Besides these system modifications, further studies on radio resource management techniques might still be able to boost system performance, especially with regard to the co-existence of multiple services.

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