Universidade de Aveiro Departamento de
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ESQUEMAS DE DIVERSIDADE COOPERATIVA PARA SISTEMAS SEM FIOS

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Cooperative diversity schemes for wireless communication systems

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Dissertação apresentada à Universidade de Aveiro para cumprimento dos requesitos necessários à obtenção do grau de Mestre em Engenharia Electrónica e Telecomunicações, realizada sob a orientação científica do Prof. Dr. Adão Silva, professor auxiliar convidado do Departamento de Electrónica, Telecomunicações e Informática da Universidade de Aveiro e Prof. Dr. Atílio Gameiro, professor associado do Departamento de Electrónica, Telecomunicações e Informática da Universidade de Aveiro.

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palavras-chave

MIMO virtual, Sistemas com Relay, Diversidade cooperativa, *Amplify-and-forward*, *Decode-and-Forward*, *Decode-and-Forward selectivo*, Diversidade espacial, Propagação multipercurso, Modulação multiportadora, OFDMA.

resumo

A presente dissertação insere-se na área das comunicações sem fios, ou mais especificamente na temática da diversidade cooperativa.

Neste trabalho é feito o estudo, implementação e avaliação do desempenho de esquemas de diversidade cooperativa de baixa complexidade para sistemas de comunicação móvel. Estes esquemas são mapeados em modelos de simulação baseados em OFDMA e são completamente simulados em `CoCentric System Studio`. Os resultados obtidos com os modelos desenvolvidos mostram que os esquemas de diversidade cooperativa atenuam os efeitos do *desvanecimento* induzido pela *propagação multipercurso*, aumentando desta forma a capacidade e cobertura dos sistemas wireless. Os ganhos são particularmente altos quando as perdas de percurso são consideráveis, como é o caso das zonas urbanas densas.

keywords Virtual MIMO, Relay system, Cooperative diversity, Amplify-and-forward, Decode-and-Forward, Selective decode-and-Forward, Space diversity, Multipath propagation, Multicarrier modulation, OFDMA.

abstract This dissertation is inserted into the wireless communication, or more specifically, into the cooperative diversity field.

within this thesis, the performance of low-complexity cooperative diversity schemes projected for mobile communication systems are studied, implemented and evaluated. These schemes are mapped into simulation models based on OFDMA and are fully simulated in the `CoCentric System Studio` environment. The obtained results show that the proposed cooperative schemes for the uplink communication mitigate *fading* induced by *multipath propagation*, thereby increasing the capacity and coverage of wireless systems. Cooperation gains are particularly high when multipath losses are considerable, as is the case for dense urban regions.

Contents

Acronyms and Abbreviations

3G	Third Generation Networks
3GPP	3rd Generation Partnership Project
4G	Fourth Generation Networks
ACK	Acknowledgement
AF	Amplify-and-Forward
AMC	Adaptive Modulation and Coding
BLAST	Bell-Labs Layered Space-Time
Bps	Byte per second
BPSK	Binary Phase Shift Keying
BS	Base Station
CDF	Cumulative Density Function
CDMA	Code Division Multiple Access
CODIV	Enhanced Wireless Communication Systems Employing **CO**operative **DIV**ersity
CSI	Channel State Information
DOA	Direction of Arrival
DF	Decode-and-Forward
EDGE	Enhanced Data Rates for GSM Evolution
EF	Equalize-and-Forward
ETSI	European Telecommunications Standards Institute
FDD	Frequency Division Duplexing
FDMA	Frequency Division Multiple Access
FEC	Forward Error Correction
GI	Guard Interval
GPRS	General Packet Radio Service
GSM	Global System for Mobile telecommunication
IEEE	Institute of Electrical and Electronics Engineers
iid	Independent and Identically Distributed
ITU	International Telecommunication Union
IP	Internet Protocol
kbps	kilobit per second
LAN	Local Area Network
LDPC	Low Density Parity Check
LOS	Line of Sight
LTE	Long Term Evolution

MAC	Media Access Control
MAN	Metropolitan Area Network
MC-CDMA	Multiple Carrier - Code Division Multiple Access
Mbps	Megabit per seconde
MIMO	Multiple Input Multiple output
MISO	Multiple Input Single Output
MMR	Mobile Multi-hop Relay
MRC	Maximum Ratio combining
MT	Mobile Terminal
NLOS	Non Line of Sight
OFDM	Orthogonal Frequency Division Multiplexing
OFDMA	Orthogonal Frequency Division Multiple Access
P2P	Peer-to-Peer
PDF	Probability Density Function
PHY	Physical Layer
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
SIMO	Single Input Multiple Output
SINR	Signal-to-Interference Noise Ratio
SNR	Signal-to-Noise Ratio
STC	Space Time Coding
STBC	Space Time Block Coding
SDF	Selective Decode-and-Forward
SDMA	Space Division Multiple Access
TDD	Time Division Duplexing
TDM	Time Division Multiplexing
TDMA	Time Division Multiple Access
ULA	Uniform Linear Array
UT	User Terminal
UMTS	Universal Mobile Telecommunications System
VMIMO	Virtual MIMO
WiBro	Wireless Broadband
WiMAX	Worldwide Interoperability Access
WLAN	Wireless Local Area Network
WMAN	Wireless Metropolitan Area Network

WiFi	Wireless Fidelity
WSSUS	Wide-Sense Stationary Uncorrelated Scattering Channel
WiMax	Worldwide Interoperability for Microwave Access

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Chapter 1

INTRODUCTION

As is often the case in engineering, solutions that effectively overcome one challenge may aggravate another. Design trade-offs have to be made to find the right balance among competing requirements – for example, coverage and capacity. Advances in computing power, hardware miniaturization, and signal-processing algorithms, however, enable increasingly favorable tradeoffs, albeit within the fundamental bounds imposed by laws of physics and information theory. Despite these advances, researchers continue to be challenged as wireless consumers demand even greater performance.

— Jeffrey G. Andrews

1.1 Introduction to Broadband Wireless

Today's Wireless communication is the result of ever-increasing mobility and service area requirements. Cellular telephones are commonplace, satellites broadcast television direct to home and offices are replacing internet cables with wireless networks.

Nevertheless, our society needs mobility and higher data rates and this fact has created endless challenges to the communications engineers with respect to spectrum scarceness. The inability to offer bit rates that meet multimedia communication requirements and to accommodate more users and services justifies efforts to explore higher frequency bands, as is proved by the spectra allocation of recent wireless technologies. The trend to move to higher frequency bands is likely to remain(Fig. 1.1), as mobility and higher bit rates must be provided for more people and services. Indeed, it is expected that 4G will support a wide range of services and multimedia communications effectively, meaning that the current systems will not be able to meet future demands.

In order to get a better understanding over this scenario, this section includes sections 1.1.1 up to 1.1.4. 1.1.1 starts by discussing the main features of second-generation broadband wireless systems. Then 1.1.2 reviews the technologies that are common nowadays and 1.1.3 provides further insight into 3G technologies, giving particular emphasis to HSPA and 1x EV-DO technologies. Finally it is provided section 1.1.4 in which the 3GPP LTE (Long Term Evolution) is introduced by describing its current state, architecture highlights and air interface. Further details concerning these technologies can be found in [1],[7] and [58].

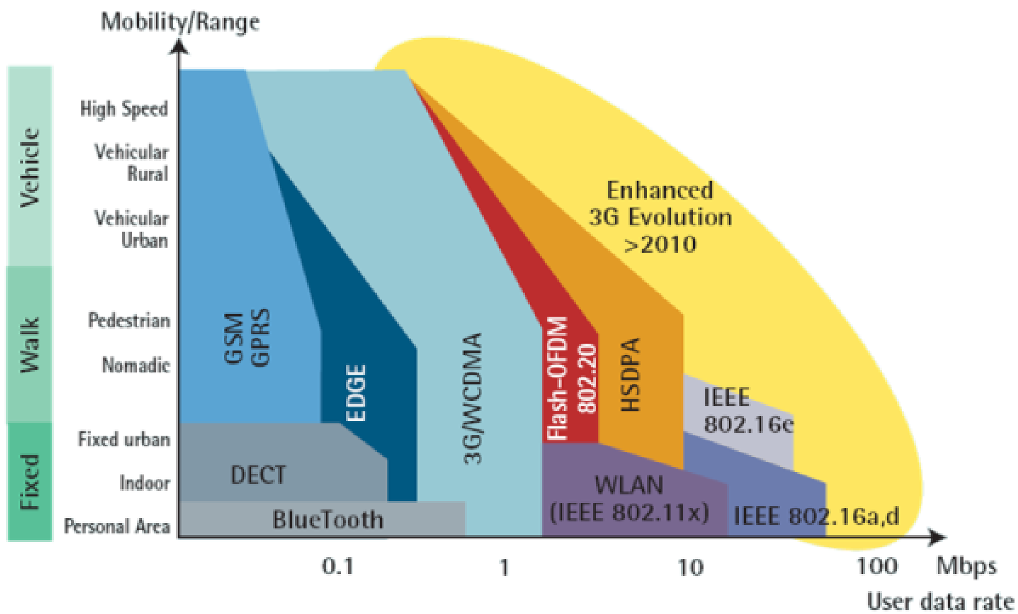


Figure 1.1: There is a trend to move wireless systems to higher frequency bands so that higher bit rates can be offered [57].

1.1.1 Overview of 2G cellular systems

Second-generation broadband wireless systems were able to overcome the LOS issue and to provide more capacity then previous 1G systems. This was done through the use of a cellular architecture and implementation of advanced-signal processing techniques to improve the link and system performance under multipath conditions. Several start-up companies developed advanced proprietary solutions that provided significant performance gains over first-generation systems. Most of these new systems could perform well under non-line-of-sight (NLOS) conditions, with customer-premise antennas typically mounted under the leaves or lower. Many solved the NLOS problem by using techniques such as *code division multiple access* (CDMA) and *multiple antennas* (i.e., space diversity). A few megabits per second throughput over cell ranges of a few miles had become possible with second generation fixed wireless broadband systems.

1.1.2 Review of current wireless data standards

A wide variety of different wireless data technologies now exist, some in direct competition with one another, others designed to be optimal for specific applications. Wireless technologies can be evaluated by a variety of different metrics described below.

Table 1.1 resumes and classifies the mentioned standards according to the network size they are meant to.

Of the standards evaluated these can be grouped as follows:

UWB, Bluetooth, ZigBee, and Wireless USB are intended for use as so called Wireless PAN systems. They are intended for short range communication between devices typically controlled by a single person. A keyboard might communicate with a computer, or a mobile phone with a handsfree kit, using any of these technologies.

Wide Area	Local Area	Personal Area
iBurst	WiFi: 802.11a, 802.11b, 802.11g, 802.11n standards	Bluetooth
WiMAX: 802.16e standard (also known as Mobile WiMAX)		Wibree
UMTS over W-CDMA		ZigBee
UMTS-TDD		Wireless USB
EV-DO		UWB
HSPA D and U standards		EnOcean
RTT		
GPRS		
EDGE		

Table 1.1: Standards classification according to network size.

WiFi is the most successful system intended for use as a WLAN system. A WLAN is an implementation of a LAN over a microcellular wireless system. Such systems are used to provide wireless Internet access (and access to other systems on the local network such as other computers, shared printers, and other such devices) throughout a private property. Typically a WLAN offers much better bandwidth and latency than the user's Internet connection, being designed as much for local communication as for access to the Internet, and while WiFi may be offered in many places as an Internet access system, access speeds are usually more limited by the shared Internet connection and number of users than the technology itself. Other systems that provide WLAN functionality include DECT and HIPERLAN.

GPRS, EDGE and 1xRTT are extensions to existing 2G cellular systems, providing Internet access to users of existing 2G networks (it should be noted that technically both EDGE and 1xRTT are 3G standards, as defined by the ITU, but are generally deployed on existing networks). 3G systems such as EV-DO, W-CDMA (including HSDPA and HSUPA) provide combined circuit switched and packet switched data and voice services as standard, usually at better data rates than the 2G extensions. All of these services can be used to provide combined mobile phone access and Internet access at remote locations. Typically GPRS and 1xRTT are used to provide stripped down, mobile phone oriented, Internet access, such as WAP, multimedia messaging, and the downloading of ring-tones, whereas EV-DO and HSDPA's higher speeds make them suitable for use as a broadband replacement.

Pure packet-switched only systems can be created using 3G network technologies, and UMTS-TDD is one example of this. Alternatively, next generation systems such as WiMAX also provide pure packet switched services with no need to support the circuit switching services required for voice systems. WiMAX is available in multiple configurations, including both NLOS and LOS variants. UMTS-TDD and WiMAX are used by Wireless ISPs to provide broadband access without the need for direct cable access to the end user.

Some systems are designed for P2P LOS communications; once 2 such nodes get too far apart to directly communicate, communication fails. Other systems are designed to form a wireless mesh network using one of a variety of list of ad-hoc routing protocols. In a mesh network, when 2 nodes get too far apart to directly communicate, they can still indirectly communicate through intermediate nodes.

1.1.3 Overview of 3G cellular systems

The evolution from 2G to 3G represents a change in many aspects: new technology, change of focus from voice to mobile multimedia, simultaneous support of several QoS classes in a single radio interface. Around the world, mobile operators are upgrading their networks to 3G technology to deliver broadband applications to their subscribers. Mobile operators using GSM (global system for mobile communications) are deploying UMTS (universal mobile telephone system) and HSDPA (high speed downlink packet access) technologies as part of their 3G evolution. Traditional CDMA operators are deploying 1x EV-DO (1x evolution data optimized) as their 3G solution for broadband data. In China and parts of Asia, several operators look to TD-SCDMA (time division-synchronous CDMA) as their 3G solution. All these 3G solutions provide data throughput capabilities on the order of a few hundred kilobits per second to a few megabits per second.

Let us briefly review the capabilities of HSPA and 1x EV-DO technologies.

HSPA

HSDPA is a downlink-only air interface defined in the Third-generation Partnership Project (3GPP) UMTS Release 5 specifications. HSDPA is capable of providing a peak user data rate (layer 2 throughput) of 14.4Mbps, using a 5MHz channel. Realizing this data rate, however, requires the use of all 15 codes, which is unlikely to be implemented in mobile terminals. Using 5 and 10 codes, HSDPA supports peak data rates of 3.6Mbps and 7.2Mbps, respectively. Typical average rates that users obtain are in the range of 250kbps to 750kbps. Enhancements, such as spatial processing, diversity reception in mobiles, and multiuser detection, can provide significantly higher performance over basic HSDPA systems. It should be noted that HSDPA is a downlink-only interface; hence until an uplink complement of this is implemented, the peak data rates achievable on the uplink will be less than 384kbps, in most cases averaging 40kbps to 100kbps. An uplink version, HSUPA (high-speed uplink packet access), supports peak data rates up to 5.8Mbps and is standardized as part of the 3GPP Release 6 specifications; deployments are expected in 2007. HSDPA and HSUPA together are referred to as HSPA (high-speed packet access).

1x EV-DO

1x EV-DO is a high-speed data standard defined as an evolution to second-generation IS-95 CDMA systems by the 3GPP2 standards organization. The standard supports a peak downlink data rate of 2.4Mbps in a 1.25MHz channel. Typical user-experienced data rates are in the order of 100kbps to 300kbps. Revision A of 1x EV-DO supports a peak rate of 3.1Mbps to a mobile user; Revision B will support 4.9Mbps. These versions can also support uplink data rates of up to 1.8Mbps. Revision B also has options to operate using higher channel bandwidths (up to 20MHz), offering potentially up to 73Mbps in the downlink and up to 27Mbps in the uplink. In addition to providing high-speed data services, 3G systems are evolving to support multimedia services. For example, 1x EV-DO Rev A enables voice and video telephony over IP. To make these service possible, 1xEV-DO Rev A reduces air-link latency to almost 30ms, introduces intrauser QoS, and fast intersector handoffs. Multicast and broadcast services are also supported in 1x EV-DO. Similarly, development efforts are under way to support IP voice, video, and gaming, as well as multicast and broadcast services over UMTS/HSPA networks. It should also be noted that 3GPP is developing the next major revision to the 3G standards.

The objective of this long-term evolution (LTE) is to be able to support a peak data rate of 100Mbps in the downlink and 50Mbps in the uplink, with an average spectral efficiency that is three to four times that of Release 6 HSPA. In order to achieve these high data rates and spectral efficiency, the air interface will likely be based on OFDM/OFDMA and MIMO (multiple input/ multiple output), with similarities to WiMAX.

Similarly, 3GPP2 also has longer-term plans to offer higher data rates by moving to higher bandwidth operation. The objective is to support up to 70Mbps to 200Mbps in the downlink and up to 30Mbps to 45Mbps in the uplink in EV-DO Revision C, using up to 20MHz of bandwidth. It should be noted that neither LTE nor EV-DO Rev C systems are expected to be available until about 2010.

1.1.4 Overview of LTE as a possible 4G

Despite the high capacity offered by the 3G technology, the rapid growth of Internet services and increasing interest in portable computing devices are likely to create a strong demand for high-speed wireless data services, presumably with a maximum information bit rate of more than 2-20Mbps in a vehicular environment and possibly 50-100Mbps in indoor to pedestrian environments, using a 50-100MHz bandwidth [35]. Especially in the downlink, high throughput is needed since the number of downloads of large data files from web sites and servers will increase and broadcast / multicast services may become a reality, that has to be accommodated by 4G systems. The European vision for this new generation is one of a fully IP-based integrated system offering all services, all the time and designed to support multiple classes of terminals.

According to [1], *the 3rd Generation Partnership Project (3GPP) is a collaboration agreement that was established in December 1998. The collaboration agreement brings together a number of telecommunications standards bodies and its initial scope was to produce globally applicable Technical Specifications and Technical Reports for a 3rd Generation Mobile System based on evolved GSM core networks and the radio access technologies that they support (i.e., Universal Terrestrial Radio Access (UTRA) both Frequency Division Duplex (FDD) and Time Division Duplex (TDD) modes). The scope was subsequently amended to include the maintenance and development of the Global System for Mobile communication (GSM) Technical Specifications and Technical Reports including evolved radio access technologies (e.g. General Packet Radio Service (GPRS) and Enhanced Data rates for GSM Evolution (EDGE)).*

Therefore, the 3GPP goals include:

- improving spectral efficiency;
- lowering costs;
- improving services;
- making use of new spectrum;
- better integration with other open standards.

The LTE air interface will be added to the specification in Release 8 and can be found in the 36-series of the 3GPP specifications [1]. Although it is an evolution of UMTS, the LTE air interface is a completely new system based on OFDMA in the downlink and SC-FDMA (DFTS-FDMA) in the uplink that efficiently supports multi-antenna technologies (further details can be found in section 3). The architecture that will result from this work is called EPS (Evolved Packet System) and comprises E-UTRAN (Evolved UTRAN) on the access side and EPC (Evolved Packet Core) on the core side.

Current State

While 3GPP Release 8 has yet to be ratified as a standard, much of the standard will be oriented around upgrading UMTS to a so-called fourth generation mobile communications technology, essentially a wireless broadband Internet system with voice and other services built on top.

The standard includes:

- Peak download rates of 326.4 Mbit/s for 4x4 antennas, 172.8 Mbit/s for 2x2 antennas for every 20 MHz of spectrum;
- Peak upload rates of 86.4 Mbit/s for every 20 MHz of spectrum;
- 5 different terminal classes have been defined from a voice centric class up to a high end terminal that supports the peak data rates. All terminal will be able to process 20 MHz bandwidth;
- At least 200 active users in every 5 MHz cell. (i.e., 200 active data clients);
- Optimal cell size of 5 km, 30 km sizes with reasonable performance, and up to 100 km cell sizes supported with acceptable performance;
- Co-existence with legacy standards (users can transparently start a call or transfer of data in an area using an LTE standard, and, should coverage be unavailable, continue the operation without any action on their part using GSM/GPRS or W-CDMA-based UMTS or even 3GPP2 networks such as CDMA or EV-DO)
- Possibility to deliver services such as Mobile TV using the LTE infrastructure, and is a competitor for DVB-H-based TV broadcast.

A large amount of the work is aimed at simplifying the architecture of the system, as it transits from the existing UMTS circuit and packet switching combined network to an all-IP flat architecture system.

Preliminary requirements have been released for LTE-Advanced, expected to be part of 3GPP Release 10. LTE-Advanced will be a software upgrade for LTE networks and enable peak download rates over 1Gbit/s that fully supports the 4G requirements as defined by the ITU-R. It also targets faster switching between power states and improved performance at the cell edge. A first set of requirements has been approved in June 2008.

Timetable

The LTE standard reached the functional freeze milestone in March 2008. Stage 2 Freeze was scheduled for mid 2008 and official ratification happens in December 2008. The standard has been complete enough that hardware designers have been designing chipsets, test equipment and base stations for some time. LTE test equipment has been shipping from several vendors since early

2008 and Motorola demonstrated a LTE RAN standard compliant eNodeB and LTE chipset at Mobile World Congress 2008.

Fig. 1.2 resumes the expected evolution of the major wireless technologies. It contrasts EDGE, HSPA, LTE, EV-D0 and WiMAX. Here the throughput rates are peak network rates and that rates refer to initial network deployment except 2006 (which shows available technologies at that year).

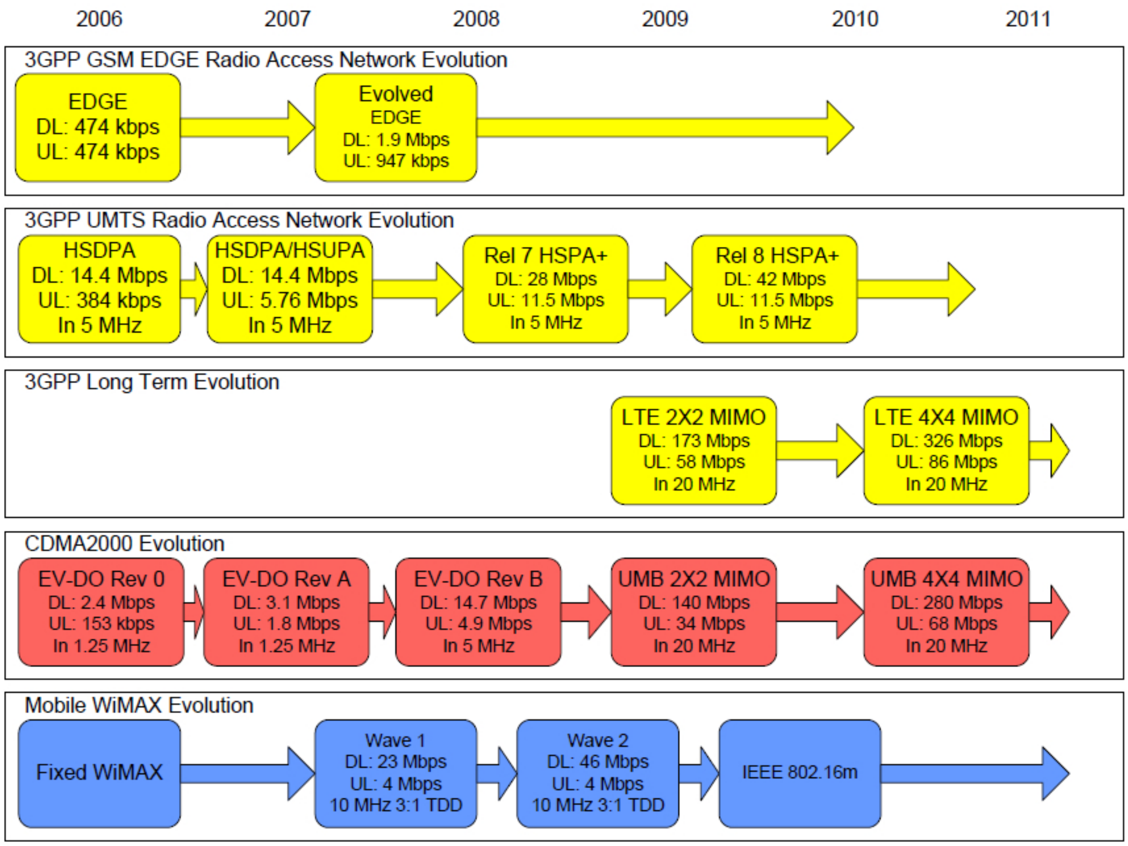


Figure 1.2: Expected evolution of wireless technologies[47].

An "All IP Network" (AIPN)

A characteristic of so-called "4G" networks such as LTE is that they are fundamentally based upon TCP/IP, the core protocol of the Internet, with higher level services such as voice, video, and messaging, built on top of this. In 2004, the 3GPP proposed this as the future of UMTS and began feasibility studies into the so-called All IP Network (AIPN). These proposals form the basis of the effort to build the higher level protocols of evolved UMTS. The LTE part of this effort is called the *3GPP System Architecture Evolution*.

At a glance, the UMTS back-end becomes accessible via a variety of means, such as GSM's/UMTS's own radio network (GERAN, UTRAN, and E-UTRAN), WiFi, and even competing legacy systems such as CDMA2000 and WiMAX. Users of non-UMTS radio networks would be provided with an entry-point into the IP network, with different levels of security depending on the trustworthiness of the network being used to make the connection. Users of GSM/UMTS networks would use an integrated system where all authentication at every level of the system is covered by a single system, while users accessing the UMTS network via WiMAX and other similar technologies would handle the WiMAX connection and the UMTS link-up in different ways.

E-UTRA Air Interface

Release 8's air interface, E-UTRA (Evolved UTRA, the E- prefix being common to the evolved equivalents of older UMTS components) would be used by UMTS operators deploying their own wireless networks. It's important to note that Release 8 is intended for use over any IP network, including WiMAX and WiFi, and even wired networks.

The proposed E-UTRA system uses OFDMA for the downlink (tower to handset) and Single Carrier FDMA (SC-FDMA) for the uplink and employs MIMO with up to four antennas per station. The channel coding scheme for transport blocks is turbo coding and a contention-free quadratic permutation polynomial (QPP) turbo code internal interleaver.

The use of OFDM, a system where the available spectrum is divided into thousands of very thin carriers, each on a different frequency, each carrying a part of the signal (see more details section 4.1), enables E-UTRA to be much more flexible in its use of spectrum than the older CDMA based systems that dominated 3G. CDMA networks require large blocks of spectrum to be allocated to each carrier, to maintain high chip rates, and thus maximize efficiency. Building radios capable of coping with different chip rates (and spectrum bandwidths) is more complex than creating radios that only send and receive one size of carrier, so generally CDMA based systems standardize both. Standardizing on a fixed spectrum slice has consequences for the operators deploying the system: too narrow a spectrum slice would mean the efficiency and maximum bandwidth per handset suffers; too wide a spectrum slice, and there are deployment issues for operators short on spectrum. This became a major issue with the US roll-out of UMTS over W-CDMA, where W-CDMA's 5 MHz requirement often left no room in some markets for operators to co-deploy it with existing GSM standards.

OFDM has a Link spectral efficiency greater than CDMA, and when combined with modulation formats such as 64QAM, and techniques as MIMO, E-UTRA has proven to be considerably more efficient than W-CDMA with HSDPA and HSUPA.

- **Downlink**

The subcarrier spacing in the OFDM downlink is 15 kHz and there is a maximum of 1200 subcarriers available. The number of subcarriers is dependent on the used bandwidth (1.4MHz and up to 20Mhz),subcarriers don't occupy 100% of the used bandwidth as Cyclic Prefixes (Guards) occupies a part of it.The Mobile devices must be capable of receiving all subcarriers but a base station need only support transmitting 72 subcarriers. The transmission is divided in time into time slots of duration 0.5 ms and subframes of duration 1.0 ms. A radio frame is 10 ms long.

Supported modulation formats on the downlink data channels are QPSK, 16QAM and 64QAM.

- **Uplink**

The currently proposed uplink uses SC-FDMA multiplexing, and QPSK or 16QAM (64QAM optional) modulation. SC-FDMA is used because it has a low Peak-to-Average Power Ratio (PAPR). Each mobile device has at least one transmitter. If virtual MIMO / Spatial division multiple access (SDMA) is introduced the data rate in the uplink direction can be increased depending on the number of antennas at the base station. With this technology more than one mobile can reuse the same resources.

1.2 Motivation and Objective

1.2.1 Future demands high bit rates

Wireless technology is one of the key components for enabling the information society and is advancing at a rapid pace. With the emerging of new technologies and the phenomenal growth of wireless services, requirements for the radio frequency spectrum are increasing at an astronomical rate.

It is expected that the demand for wireless services will continue to increase in the near and medium term, therefore calling for more capacity and creating the need for cost effective transmission techniques that can exploit scarce spectral resources efficiently.

It is anticipated that the broadband mobile component of beyond 3G systems must be able to offer bit rates in excess of 100Mbps in indoor and picocell environments.

To achieve such high bit rates, so as to meet the quality of service requirements of future multimedia applications, *Orthogonal Frequency Division Multiple Access* (OFDMA) has been adopted in different flavors of broadband wireless systems [32, 23]. OFDMA is a robust and yet spectrally efficient communication strategy. In essence, it is about splitting the available spectrum in several narrowband frequency bands and distribute them among the users.

Besides OFDMA, the use of spatial diversity has also been proposed. Multiple antennas at transmitter/receiver side is commonly referred to as Multiple input and multiple output (MIMO), and is a very promising technique to mitigate the channel fading and thus improving the cellular system capacity. By configuring multiple antennas at both the base station (BS) and mobile terminal (MT), the channel capacity may be improved proportionally to the minimum number of the antennas at the transmitter and receiver [15]. However, using an antenna array at the MTs may not be feasible due to size, cost and hardware limitations. Moreover, if the MTs are equipped with multiple antennas, the spatial separation between antennas must be great enough to guarantee the statistical independence of faded signals for optimal performance [18]. These devices are usually small and light and thus this spatial separation requirement is difficult to satisfy. This limitation is the reason why cooperative communications have been proposed as a solution for future uplink communication in wireless systems scenarios.

Cooperative communications allows single antenna devices to gain some benefits of spatial diversity without the need for physical antenna arrays [16]. The underlying idea is to program the mobile terminals (MTs) to send their own information and also to cooperate with its neighbors by relaying their information. By doing so they create a *virtual* array and they might end up with gains that are comparable to MIMO.

1.2.2 Cooperative Diversity History

A cooperative communication scenario contrasts to communication from a single source to a single destination without the help of any other communicating terminal, which is called direct, single-user or point-to- point communication (P2P), as it can be seen in Figure 1.3.

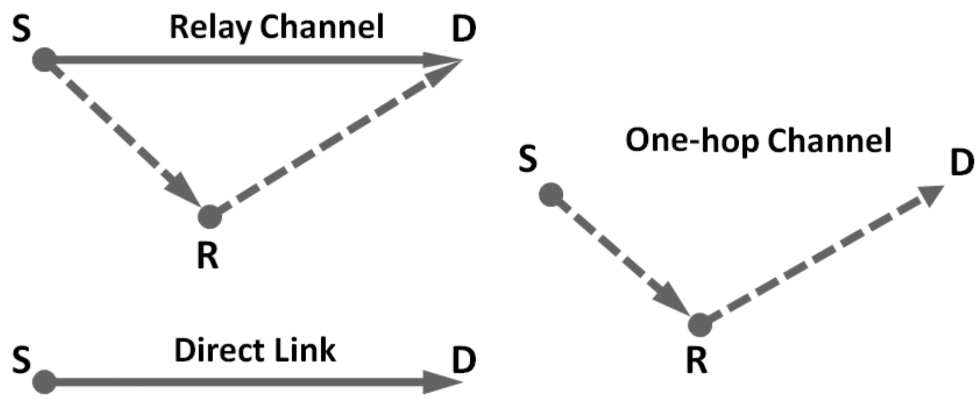


Figure 1.3: Direct, two-hop and relay communications.

User-cooperation is possible whenever there is at least one additional node willing to aid in communication. The simplest and oldest form of user-cooperation is perhaps multi-hopping, which is nothing but a chain of point-to-point links from the source to the destination (Figure 1.3 shows one-hop communication). No matter what the channel, there is some attenuation of the signal with distance, which makes long-range P2P communication impractical.

Research on cooperative diversity can be traced back to the pioneering papers of Van der Meulen [38] and Cover, El Gamal [12] on the information theoretic properties of the relay channel. They introduced and discussed the three-terminal relay channel (depicted in Figure 1.3). At the time, we have results for upper and lower bounds on the capacity of the relay channel, but the capacity of the general relay channel is still unknown.

Explicit cooperation of neighboring nodes was considered in [49, 13, 31]. In such cooperative transmission scenarios, two or more sources (genuine sources or relays) transmit the same information to a destination, generating a *virtual antenna array*. In [13],[31], the use of orthogonal space-time block coding (STBC) in a distributed fashion for practical implementation of user cooperation has been proposed. Several authors have also addressed the search and design of practical distributed space-time codes for cooperative communications [51]. A cooperative scheme for the UL OFDMA has been proposed in [22]. In this scheme each user transmits his partner's and his own data on different subcarriers.

1.2.3 Preliminaries of Relaying

The relay channel is the three-terminal communication channel shown in Figure 1.4. The terminals are labeled the source (S), the relay (R) and the destination (D). These three nodes are conceptually divided into two subsets by two cuts of interest: C_1 or the broadcast cut which separates S from R, D, and C_2 or the multiple-access cut, which separates S, R from D. The channel input at S is given by X, the input at R is W, and the outputs at R and D are V and Y respectively.

All information originates at S, and must travel to D. The relay aids in communicating information from S to D without actually being an information source or sink. The signal being transmitted from the source is labeled X. The signal received by the relay is V. The transmitted

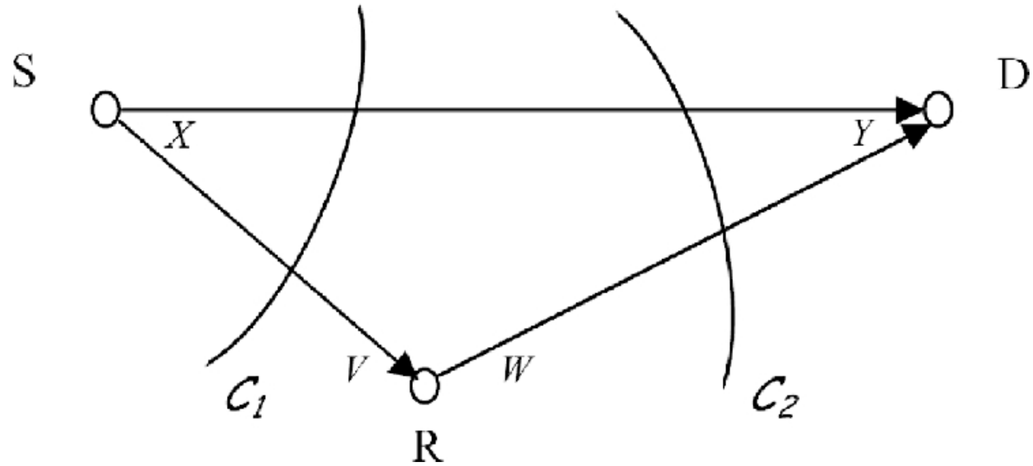


Figure 1.4: The relay channel with three nodes[16].

signal from the relay is W, and the received signal at the destination is Y . Several notions of relaying exist in the literature. The prominent ones are listed in this section.

Conceptually, information is relayed in two phases or modes: first, when S transmits and (R, D) receive, commonly called the broadcast (BC) mode, and second when (S, R) transmit and D receive, also known as the multiple-access (MAC) mode. Note that this differentiation is only conceptual since it is possible for communication in both modes to take place simultaneously. Now, four models of relaying that can be classified based on the above two modes are enumerated:

1. $S \to (R, D); (S, R) \to D$ (most general form of relaying);
2. $S \to R; (S, R) \to D$ (D ignores signal from S in first mode);
3. $S \to (R, D); R \to D$ **(S does not transmit in second mode)**;
4. $S \to R; R \to D$ (multi-hop communication).

Of these, the first model is the most general, and most early results on relaying were based on the first model. The second and the third are simplified models introduced mainly for analytical tractability. For example, they simplify the analysis of outage probabilities and the design of space-time codes for fading relay channels in [31, 22].

Within this dissertation, the source S and relay R are mobile terminals, whereas the destination D is a base station BS, and the selected one-hop communication model was the third one. The choice was motivated by practical reasons. In this model, the transmission occurs in such a way that the orthogonality between relay and source data is provided by time. In contrast, in the second phase of the first and second models, the BS will be receiving information from the source and relay (simultaneously). The underlying problem is how to separate the source and relay data, and that means that the orthogonality must be achieved by the usage of orthogonal codes, which adds complexity to the system.

The last model of relaying is much older as well as simpler than the other three, and is commonly known as multi-hop communication. Unlike the other three models, multi-hop communication does not yield diversity benefits, and it is primarily used to combat signal attenuation in long-range communication. In wireless communication, usually there is severe attenuation of signal power with distance. This attenuation is characterized by a channel attenuation exponent γ. In other words, if the transmitted power is P, then the received power at a distance d is $\frac{P}{d\gamma}$. The value of γ lies in the range of 2 to 6 for most wireless channels. This attenuation makes long-range communication virtually impossible. The simplest solution to this problem is to replace a single long-range link with a chain of short-range links by placing a series of nodes in between the source and the destination. A distinguishing feature of multi-hopping is that each node in this chain communicates only with the one before and the one after in the chain, or nodes that are one "hop" away. In a wireless environment, it may be possible for a node to receive or transmit its signal to other nodes that are several hops away, but such capability is ignored in multi-hopping, making it a simple and extremely popular, but suboptimal mode of user-cooperation.

Of all the modes of user-cooperation discussed so far, multi-hopping is the only one that is widely implemented today.

1.2.4 Half-duplex versus Full-duplex Relaying

A relay is said to be half-duplex when it cannot simultaneously transmit and receive in the same band. In other words, the transmission and reception channels must be orthogonal. Orthogonality between transmitted and received signals can be in time-domain, in frequency domain, or using any set of signals that are orthogonal over the time frequency plane. If a relay tries to transmit and receive simultaneously in the same band, then the transmitted signal interferes with the received signal. In theory, it is possible for the relay to cancel out interference due to the transmitted signal because it knows the transmitted signal. In practice, however, any error in interference cancelation (due to inaccurate knowledge of device characteristics or due to the effects of quantization and finite-precision processing) can be catastrophic because the transmitted signal is typically 100-150dB stronger than the received signal as noted in [31]. Due to the difficulty of accurate interference cancelation, full-duplex radios are not commonly used.

Although early literature on information theoretic relaying was based almost entirely on full-duplex relaying [38, 12], in recent years a lot of research, and especially research directed towards practical protocols, has been based on the premise of half-duplex relaying [31].

1.2.5 Relay Protocols

The capacity of the general relay channel of Figure 1.4 is not known even today, over thirty years after the channel was first proposed. Moreover, there is no single cooperation strategy known that works best for the general relay channel. As it will be discussed in chapter 5, there are at least two fundamental ideas based on which the source and relay nodes can share their resources to achieve the highest throughput possible for any known coding scheme. The cooperation strategies based on these different ideas have come to be known as relay protocols. The scope of this dissertation includes the decode-and-forward and the amplify-and-forward ideas.

Decode-and-forward protocol

The first idea involves decoding of the source transmission at the relay. The relay then retransmits the decoded signal after possibly compressing or adding redundancy. This strategy is known as the decode-and-forward (DF) protocol, named after the fact that the relay can and does decode the source transmission. The decode-and-forward protocol is close to optimal when the source-relay channel is excellent, which practically happens when the source and relay are physically near each other. When the source-relay channel becomes perfect, the relay channel becomes a 2 x 1 multiple-antenna system. Following the naming convention of [12], some authors use the term cooperation to strictly mean the decode-and-forward type of cooperation.

The second idea, sometimes called observation, is important when the source-relay and the source-destination channels are comparable, and the relay-destination link is good. In this situation, the relay may not be able to decode the source signal, but nonetheless it has an independent observation of the source signal that can aid in decoding at the destination. Therefore, the relay sends an estimate of the source transmission to the destination. This strategy is known as the estimate-and-forward (also known as compress-and-forward or quantize-and-forward) protocol.

Amplify-and-forward protocol

The amplify-and-forward (also sometimes called scale-and-forward) protocol is a simple cooperative signaling in which each user receives a noisy version of the signal transmitted by its partner. As the name implies, the user then amplifies and retransmits this noisy version to the destination i.e., the base station(BS). The BS combines the information sent by the user and partner, and makes a final decision on the transmitted bit. Although noise is amplified by cooperation, the base station receives two independently faded versions of the signal and can make better decisions on the detection of information. This method was proposed and analyzed by Laneman et al [29].

1.2.6 Scope of this Dissertation

Despite the straightforwardness of the cooperative diversity concept, its implementation in a practical UL scenario hides many challenging questions and make it an evolving research topic.

One of such questions is *what kind of processing should the relay perform?* In fact, the signal processing might be as simple as just amplifying and forwarding, but it might be more complex and involve demodulation and decoding. It might also require channel estimation. Since the quality of the transmission channels, it is important to investigate which relaying modes are more suitable for the different scenarios.

It should be borne in mind that cooperating requires the MT's that act as relays to sacrifice bandwidth. Therefore the MTs are not expected to "choose" for cooperation unless they really need to do so, ie, when the cooperation gains are significant. So another underlying question is *in which UL communication scenario does cooperation modes outperform the non-cooperative one?* or, stated another way, *how bad must the direct link channel be in comparison with the relay one to motivate cooperative behavior?* It is important to mention that the results are taken from a network perspective and the gains are measured using system capacity and bit error rate (BER) metrics.

The work that was developed in this thesis is within the scope the European CODIV project (FP7-ICT-2007-215477). A significant part of the work is devoted to evaluating the cooperative schemes performance by simulating different transmission chains where the MTs cooperate, and comparing the results with the classical situation where they do not cooperate. It is worth mentioning that, besides CODIV [11], there is already an IEEE working group (IEEE 802.16j) which is responsible for addressing this kind of questions [24].

1.3 Organization

This dissertation is organized as follows. Until the end of this chapter, we revise the telecommunications history until the current situation and discuss future tendencies, namely LTE (Long Term Evolution). A brief introduction to cooperative diversity from a historical perspective is also provided.

Within chapter 2, a brief overview over fundamental wireless communication concepts is given, hoping that it will raise the reader's awareness to the main channel impairments that affect the communication systems. Here, diversity techniques are also presented as a powerful means to increase the channel reliability.

Chapter 3 discusses multiantenna techniques and emphasizes on mathematical framework for the capacity determination of MIMO systems.

Chapter 4 discusses the main ideas behind the multicarrier techniques that were implemented in this thesis, namely orthogonal frequency division multiplexing (OFDM) and its multiple access version, ie, OFDMA. This chapter concludes with a brief overview of a wireless system whose physical layer is based on OFDMA, that is, Mobile WiMAX.

Chapter 5 is the core of this thesis. Here, the proposed cooperative diversity schemes for the uplink communication are described and also compared and contrasted to the classical non-cooperative ones. This chapter begins with a section in which a brief introduction to cooperative diversity evolution is provided. Then it presents and discusses the simulation results that we got for the proposed cooperative schemes.

Finally, chapter 6 concludes and an appendix follows.

Chapter 2

THE COMMUNICATIONS CHANNEL

The first and most fundamental challenge for wireless communication comes from the transmission medium itself. Wireless systems must rely on complex radio wave propagation mechanisms for traversing the intervening space. The requirements of most broadband wireless services are such that signals have to travel under challenging NLOS conditions. Several large and small obstructions, terrain undulations, relative motion between the transmitter and the receiver, interference from other signals, noise, and various other complicating factors together weaken, delay and distort the transmitted signal in an unpredictable and time-varying fashion.

This turns the design of digital communication systems into a challenging task, especially when the service requirements include very high data rates and high-speed mobility.

The channel non-ideality also implies that the first step for developing a proper understanding of state-of-art solutions or for designing effective solutions for future broadband wireless systems is getting a proper insight on how the wireless channel distorts signals.

In this remaining of this chapter, there is a discussion of statistical channel models for describing the channel, such as Rayleigh and Rice models, and an introduction to parameters for describing the channel distortive behavior such as coherence time and bandwidth. Finally, *diversity techniques* are presented as a powerful "set of tools" that designers have in order to increase the channels reliability. Certain diversity combining techniques like selection diversity, maximum ratio combining and equal gain combining are examined.

2.1 Statistical Models

The most simplistic channel that one can think of is an addictive white Gaussian noise (AWGN) channel. As its name suggests, the free-space medium would just pollute the signal by adding some white noise to it. It is related to the thermal noise picked up by a receiver and is proportional to the bandwidth. The higher noise floor, along with the larger pathloss, reduces the coverage range of broadband systems. Only one propagation parameter (two-sided power spectral density $No/2$ (watts/Hz)) would have to be estimated to project suitable transceivers and electronics would be simple.

Unfortunately, except for some satellite-earth link LOS situations, the AWGN model is not an accurate model for describing the channel environment conveniently.

In practical situations, it is often necessary to consider other channel impairments such as the ones that are listed below:

Distance-dependent decay of signal power: In NLOS environments, the received signal power typically decays with distance at a rate much faster than in LOS conditions. This path loss also has an inverse-square relationship with carrier frequency.

Blockage due to large obstructions: Large obstructions, such as buildings, cause localized blockage of signals. Radio waves propagate around such blockages via diffraction but incur severe loss of power in the process. This loss, referred to as shadowing, is in addition to the distance-dependent decay and is a further challenge to overcome.

Large variations in received signal envelope: The presence of several reflecting and scattering objects in the channel causes the transmitted signal to propagate to the receiver via multiple paths. This leads to the phenomenon of multipath fading, which is characterized by large (tens of dBs) variations in the amplitude of the received radio signal over very small distances or small durations. Broadband wireless systems need to be designed to cope with these large and rapid variations in received signal strength. This is usually done through the use of one or more diversity techniques, some of which are covered in more detail in subsequent chapters.

Intersymbol interference due to time dispersion: In a multipath environment, when the time delay between the various signal paths is a significant fraction of the transmitted signal's symbol period, a transmitted symbol may arrive at the receiver during the next symbol period and cause intersymbol interference (ISI). At higher data rates, the symbol time is shorter; hence, it takes only a smaller delay to cause ISI. This makes ISI a bigger concern for broadband wireless and mitigating it more challenging. Equalization is the conventional method for dealing with ISI but at high data rates requires too much processing power. OFDM has become the solution of choice for mitigating ISI in broadband systems, including Fixed WiMAX [7].

Frequency dispersion due to motion: The relative motion between the transmitter and the receiver causes carrier frequency dispersion called Doppler spread. Doppler spread is directly related to vehicle speed and carrier frequency. For broadband systems, Doppler spread typically leads to loss of signal-to-noise ratio (SNR) and can make carrier recovery and synchronization more difficult. Doppler spread is of particular concern for OFDM systems, since it can corrupt the orthogonality of the OFDM subcarriers.

Interference: Limitations in the amount of available spectrum dictate that users share the available bandwidth. This sharing can cause signals from different users to interfere with one another. In capacity-driven networks, interference typically poses a larger impairment than noise and hence needs to be addressed.

There are channel models that are be classified as *physical models*, as they take into account the exact physics of the propagation environment, including reflecting buildings, trees and such, diffracting surfaces and *scatterers*. They are the most accurate models and can be suitable to describe propagation within a campus or delimited region. However, they are computationally intensive and difficult to put in practise, specially when we want to describe big areas. For these cases, the simpler and less accurate *statistical models* are preferred. They are based on measured statistics for a particular class of environments like topography, propagation distance, etc. As explained in [20] and depicted in Fig. 2.1, it is assumed that channel distortion can be decomposed into 3 independent phenomena: median path loss, motion over large areas and small changes in position. Motion over large areas are described by a lognormal distribution whereas the rapid variations in the signal strength are described by distributions like Rayleigh or Rice. The sum of those losses gives the statistical approximation for the terrain losses.

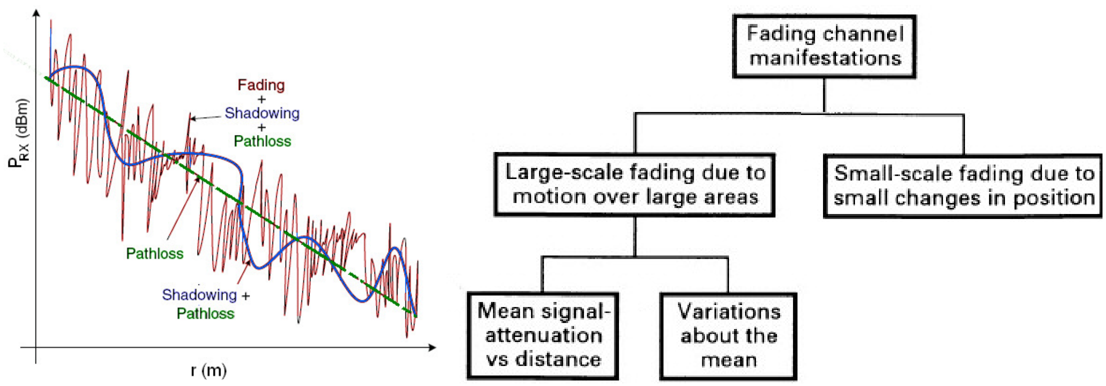


Figure 2.1: The channel distortion can be decomposed into 3 independent phenomena [7].

In the remaining section, there are more details over channel fading manifestations. More information regarding these issues can be found in [26, 46, 20].

2.2 Median Path Loss

The median path loss can be regarded as a generalization of the LOS transmission mode. Propagation distance is taken into account by the inclusion of the propagation path length parameter R and the influence of the medium comes as the path loss exponent n. The transmitted power P_T relates to the received one P_R by

$$\frac{P_R}{P_T} = \frac{\beta_0}{R^n} \tag{2.1}$$

Should n be made 2, the free-space ideal case would be recovered. However, if the medium becomes more hostile, then we penalize 2.1 by increasing n. Recommended values for n for a rural region are about 3-3.5, while for a urban area go up to 4-4.5. Obstructions due to buildings provoke n from 4 to 6.

It is common practise to express the median path loss in dB form:

$$L_P = \beta_0 + 10n \log \frac{r}{r_0} \tag{2.2}$$

where β_0 is the measured path loss at the reference distance r_0.

A number of propagation models are available to predict path loss over irregular terrain. They differ in complexity and can be optimized for indoor or outdoor environments. As an example, we can mention the well-known Okumara and Hata models([42, 19]).

2.3 Shadowing

Shadowing is due to the presence of buildings and vegetation in the cell area or slow motion of the terminal with respect to distant objects. It is fairly well described by *lognormal* models.

These models take into account the existence of several LOS paths and their relative contribution to the overall received signal.

A *lognormal* model suits very well when there is LOS and the reflective paths are unimportant. In particular, if μ is the median value of the path loss (in dB) at a specified distance R from the transmitter, then the distribution x_{dB} of the observed path losses at this distance have the PDF(*probability density function*)

$$f_R(x_{dB}) = \frac{1}{\sqrt{2\pi}\sigma_{dB}} e^{-(x_{dB}-\mu)/2\sigma^2} \tag{2.3}$$

Typical values for the standard deviation σ_{dB} range from 5 to 12 dB. The integration of Eq. 2.3 yields a CDF(cumulative density function) representation form. Figure 2.2 depicts the CDF for various σ_{dB}. From this representation, it is easy to find out the extra power margin that must be included in the link budget to ensure a certain probability of outage.

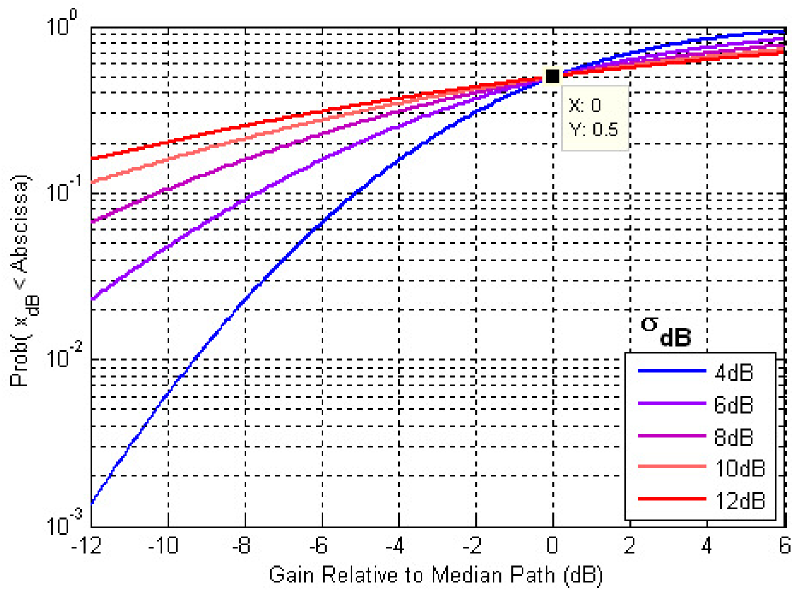


Figure 2.2: A lognormal distribution can be used to model shadowing.

However, in most wireless scenarios, there are NLOS contributions and we are interested in including the effect of reflected waves, namely where there is relative motion of nearby objects. This kind of effect is commonly referred in the literature as fading and will be the topic of discussion of section 2.4.

2.4 Fading

Fading is caused by constructive and destructive interference of multipath waves and occurs when there is relative motion. It can be due to 2 situations. One of them has to do with multipath and appears when there is relative motion of local reflecting objects(example: clouds) or motion of the terminal relative to these local objects. As a matter of fact, at a relatively large distance away from the transmitter, the received multipath amplitudes do not vary significantly. However, when the wavelengths are small compared to the distance(as is often the case), phase variations are highly sensitive to small position variations. Recalling that the received signal is the sum of all multipath components, we can understand that it can be easily distorted. In the vehicular radio scenario, sometimes it is enough to move just a few meters to notice deep signal attenuation or reinforcement. The smallest position variation can have a significant impact on the resulting sum of multipath components. The second situation is a direct consequence of Doppler effect. Such phenomena is also referred to in the literature as *fast fading*.

The power fluctuations of the received signal which are due to median path loss and shadowing can be easily corrected by a power control mechanism. Generally speaking, in the DL, the BS adjusts the transmitted power for each MT is such a way as to make up for those variations in the mean average power of the signal. That adjustment requires feedback from the MT since the BS must know the quality of the received signal. In this thesis it is assumed that the propagation losses due to median path loss and shadowing are perfectly compensated by the power control mechanisms. Thus, we shall simplify the mobile wireless channel model by considering just its *fast fading* distortive effect.

This section proceeds with further details on *fast fading*. We continue by presenting two statistical distributions for *fast fading*, namely the Rayleigh and Rice distributions, the difference being the relative contribution of LOS and NLOS contributions to the final received signal. Then a series of channel parameters such as coherence bandwidth and time are presented, hoping that they simplify the channel description for systems design purposes. These parameters are obtained by the amplitude correlation between the amplitude of the received signal (in time or frequency domains).

2.4.1 Statistical Distributions for fast fading

When there is no direct LOS, the complex envelope of N signal rays (reflections) is given by a sum of independent and identically distributed (i.i.d.) complex random variables:

$$\tilde{E} = \sum_{n=1}^{N} E_n e^{j\theta_n} \tag{2.4}$$

Relative phases θ_n are assumed to be statistically independent and uniformly distributed over $[0, 2\pi]$. Developing this expression yields the *Rayleigh probability density function*:

$$f_R(r) = \frac{r}{\sigma^2} e^{-r^2/2\sigma^2} \tag{2.5}$$

where σ^2 is half the variance power in the complex envelope.

Rayleigh-fading model is well-suited for non line-of-sight (NLOS) situations because all paths (E_n's) are relatively equal. However there is still an important case that must be discussed, which is when there is LOS and reflections are relevant. The complex envelope translates the problem by regarding the complex received wave as the sum of the direct wave E_0 and N reflections:

$$\tilde{E} = E_0 + \sum_{n=1}^{N} E_n e^{j\theta_n} \tag{2.6}$$

Mathematically speaking, the amplitude of the complex envelope is said to be *Rician distributed*:

$$f_R(r) = \frac{r}{\sigma^2} e^{-(r^2+s^2)/2\sigma^2} I_0(\frac{rs}{\sigma^2}) \text{ with } r \geq 0$$

where $s^2 = |E_0|^2$ is the power in the direct path and $I_0(.)$ is the modified Bessel function of zeroth order.

A key factor in the analysis is the ratio of the power in the direct path to the power in the reflected paths is referred to as *Rician K-factor* and is defined as the ratio of the power in the direct path to the power in the reflected paths:

$$K = \frac{s^2}{\sum_{n=1}^{N} |E_n|^2} \tag{2.7}$$

Rician K-factor can be regarded as a link for generalizing of the the Rayleigh and Gauss distributions. Should $s^2 \to 0$,that is, $K \to 0$, Rician reduces to Rayleigh distribution. If, on the other side, $\sum_{n=1}^{N} |E_n|^2 \to \infty$ then $K \to \infty$ and we recover the gaussian distribution.

In the mobile communications context we are interested in providing a certain quality of service and that implies adding a *fading margin* to the link budget, so that the signal overcomes local losses for at less given percentage of time. Therefore the most practical graphics are the ones which represent amplitude distributions in cumulative probability distribution form, that is

$$Pr(r < R) = \int_0^R f_R(r)dr \tag{2.8}$$

It requires design parameters such as *required availability.*

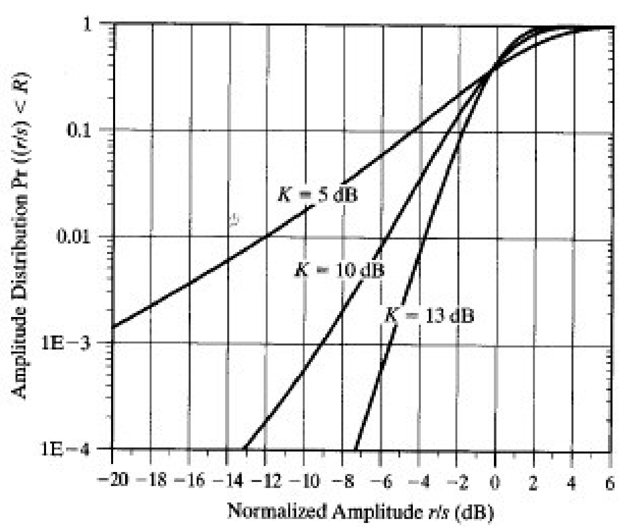


Figure 2.3: Amplitude distributions for a Rician Channel [20].

Fig. 2.3 makes it apparent that the probability of deep fades (which causes *burst* errors) diminishes as the K factor increases and is less common in Gauss channels that in Rayleigh ones.

2.4.2 Delay Spread, Coherence bandwidth and Frequency Selectivity

The correct description of the fading phenomena is very important, since it is the first step for projecting efficient mitigating techniques. For that it is useful to define parameters such as delay spread, coherence bandwidth and frequency selectivity.

We start by defining the autocorrelation function of the channel impulse response as [8]

$$R(\tau_1, \tau_2; \Delta t) = \frac{1}{2} E\left\{h(t, \tau_1) h^*(t + \Delta t, \tau_2)\right\} \tag{2.9}$$

where $()^*$ and $E.$ denote complex conjugation and average, respectively. It it common practise to assume that there is no amplitude or phase correlation between individual replicas whose delays are τ_1 and τ_2. That is the case when echoes travel through paths which are not correlated. This scenario is referred to as *uncorrelated scattering* (US) and its autocorrelation function is a simplification of Eq. 2.9:

$$R(\tau_1, \tau_2; \Delta t) = \rho(\Delta t, \tau_1)\delta(\tau_1 - \tau_2) \tag{2.10}$$

where $\rho(\Delta t, \tau)$ represents the power spectral density of the delay [8]. Fortunately, most mobile radio channels are *wide sense stationary* (WSS) with respect to the time variable and, simultaneously, of *uncorrelated spreading* in the delay variable. The combination of these 2 properties results in a class known as *wide sense stationary uncorrelated scattering*(WSSUS) channels. If we take the Fourier Transform of $\rho(\Delta t, \tau)$ with respect to the Δt, we end up with a function that describes the channel both in delay and Doppler frequency shift domains. That *scattering function* can be expressed as

$$S(\tau_1, f_D) = \int_{-\infty}^{\infty} \rho(\Delta t, \tau) e^{-j2\pi f_D \Delta t} d\Delta t \tag{2.11}$$

Eq. 2.11 is real and can be regarded as a measure for the average power per unit frequency, in the f_D domain, as a function of the delay at the output of the channel. We can get further insight by performing its integration with respect to the Doppler frequency shift:

$$\rho(\tau) = \int_{-\infty}^{\infty} S(\tau, f_D) df_D \tag{2.12}$$

Eq. 2.12 is the delay power spectrum (PDS) and reduces to $\rho(\Delta t, \tau)$ when $\Delta t = 0$. The PDS means the average power at the output of the channel as a function of the delay and can also be regarded as the mean of Eq.2.11 over all Doppler frequency shifts.

From 2.12 we can define two important design parameters: mean delay spread $\overline{\tau}$ and rms delay spread, σ_τ. The mean delay spread is given by

$$\overline{\tau} = \frac{\int_0^\infty \tau \rho(\tau) d\tau}{\int_0^\infty \rho(\tau) d\tau} \tag{2.13}$$

while the rms delay spread is defined as

$$\sigma_\tau = \sqrt{\frac{\int_0^\infty (\tau - \overline{\tau})^2 \rho(\tau) d\tau}{\int_0^\infty \rho(\tau) d\tau}} \tag{2.14}$$

If the power delay profile of the channel is discrete and consists of L_p distinct components, equations 2.13 and 2.14 can be rewritten as 2.15 and 2.16, respectively.

$$\overline{\tau} = \frac{\sum_{p=1}^{L_p} \tau_p \eta_p}{\sum_{p=1}^{L_p} \eta_p} \tag{2.15}$$

$$\sigma_\tau = \sqrt{\frac{\sum_{p=1}^{L_p} \eta_p (\tau_p - \overline{\tau})^2}{\sum_{p=1}^{L_p} \eta_p}} \tag{2.16}$$

where η_p is the normalized received power of path p, whose delay is τ_p. $\overline{\tau}$ relates to the phase error range and σ_τ is an indication of the possible inter symbol interference that limits the communication systems performance. The symbol duration T_s impacts on the channel classification. Should $T_s \gg \tau_{max}$, the signal will "perceive" the channel as being narrowband(NB) and the ISI will be small. But, if $T_s \ll \tau_{max}$ the channel looks wideband(WB). In this case the ISI can be substantial. This is the scenario for the current and prospective communication systems. The solution for this issue is *guard interval* insertion between consecutive symbols. In the frequency domain, the distinction between NB and WB channels relates to the *coherence bandwidth* parameter (B_c), which is the minimum frequency separation between 2 consecutive decorrelated frequencies. As a rule of thumb, the coherence bandwidth of the channel corresponds to the frequency separation that ensures a correlation factor of approximately 0.5 and can be given by either $B_c \approx \frac{1}{5\sigma_\tau}$ [46] or $B_c \approx \frac{1}{\sigma_\tau}$ [44].

The spectrum of a broadband signals crosses several coherence bandwidths ($B \gg B_c$). For a frequency separation superior to the channel coherence bandwidth, fading is uncorrelated and the channel is *frequency selective*. On the other side, in a narrowband channel, the bandwidth is typically much less than the coherence bandwidth of the channel, ie,$B \ll B_c$. This channel is non selective in the frequency domain (*flat fading*). B_c is an important measure. In the multicarrier systems context, the frequency diversity is explored, ie, different copies of the data symbols are transmitted in subcarriers whose frequency separation exceeds the coherence bandwidth of the channel.

Despite the fact that the literature often classifies the channel as NB or WB, it should be pointed out that this procedure is not accurate. Indeed, it is the coherence bandwidth of the signal that should be classified as either NB or WB.

2.4.3 Doppler Spread, Coherence Time and Time Selectivity

Time selectivity is determined by the mobile terminal motion and surrounding objects. Since we are interested in fast fading, we only care about the variations of signal due to Doppler shifts. Each propagation path is associated with a Doppler frequency that depends on the angle between it and at direction of the MT. The maximum Doppler frequency is given by

$$f_{D,p,l} = \frac{\nu}{\lambda_c} \cos(\varphi_{p,l}) \tag{2.17}$$

where ν,λ_c and $\varphi_{p,l} \in [0; \pi]$ represent the velocity of the MT, carrier wavelength and the angle that each subpath makes with the direction of the MT, respectively.

Depending on the propagation scenario, the MT can be surrounded by several objects, which results in a wide variety of angles and different Doppler shifts. The consequence is that NB signals that cross those kind of channels will end up being spread in the frequency domain by the different Doppler shifts.

The multipath channel can be described with by the Doppler power spectral density (PSD) $\rho_D(f_D)$ and Doppler Spread σ_D. Those measures depend on the incidence angle distribution of the different subpath's [10].

If the number of subpath's are assumed to be very high and its incidence angles are uniformly distributed in the $[0; 2\pi]$ range, it is relatively straightforward to find the Doppler PSD for each subpath p or for a frequency flat channel. The autocorrelation for each path p is given by

$$R(\Delta t) = \frac{1}{2} E\left\{h(t, \tau_p) h^*(t + \Delta t, \tau_p)\right\} \tag{2.18}$$

The normalized function can be expressed as [43]

$$R_{norm}(\Delta t) = J_0(2\pi f_{D_{max}} \Delta t) \tag{2.19}$$

where $J_0(.)$ is the modified Bessel function of zeroth order and first kind. R_{norm} relates to the Doppler PSD since it is its Fourier Transform. It is found to be

$$\rho_D(f_D) = \begin{cases} \frac{1}{\pi f_{D_{max}} \sqrt{1-(f_D/f_{D_{max}})^2}} & \forall f_D \in \,]-f_{D_{max}}, f_{D_{max}}[\\ 0 & \text{otherwise} \end{cases} \tag{2.20}$$

In the literature, this spectra is often called Jakes PSD [26]. Doppler Spread is defined by inequality 2.21

$$\sigma_D \leq 2\,|f_{D_{max}}| \tag{2.21}$$

The Coherence Time (T_C) is defined as the minimum separation in time between two consecutive uncorrelated samples. But it practise it is found as the time separation that makes the correlation factor equal to 0.5: $T_c \approx \frac{1}{2 f_{D_{max}}}$ [46] or $T_c \approx \frac{9}{16\pi f_{D_{max}}}$ [50].

In case the duration of the transmitted symbol exceeds the coherence time, the channel is classified as *time selective* or *fast fading*. Otherwise it is *slow fading*.

Diagram 2.4 evidences the duality that exists between time and Doppler spreading of the signal. It shows that coherence time relates to time dispersion in the same way as coherence bandwidth relates to Coherence Bandwidth. Table 2.1 summarizes the results for the channel classification based on those 2 parameters.

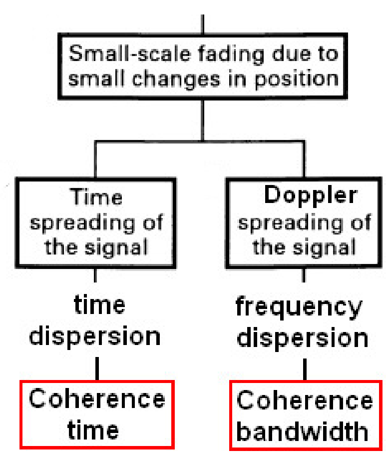


Figure 2.4: Period and Coherence bandwidth relate to small-scale fading.

Condition	True	False
$1/bitrate < T_c$	slow fading	fast fading
$BW_{sinal} < B_c$	freq. flat	freq. selective

Table 2.1: Channel behavior characterization based on period and Coherence bandwidth.

The coherence time of the channel is a very important channel parameter, specially if we want to explore the temporal diversity of the channel. It is achieved by using *interleaving* and channel coding techniques. For that several copies of a given data symbol are transmitted at different instants, which must be spaced apart by more than T_c seconds. Also in the design of cooperative diversity schemes, we are particularly concerned with the coherence time, because that dictates the minimum time difference between the samples which traveled through the direct path and the ones that crossed the relay. Thus, it impacts on the maximum datarate.

In the broadband wireless context, the channel is likely to be fast fading and also frequency selective. Therefore, the ideal would be to design systems that explore both temporal and spatial diversity.

2.5 Channel Models proposed by HIPERLAN/2

Here we describe the channel model that we used in the simulations of this dissertation. We chose this model because of its low implementation complexity, carrier frequency (5 GHz) and availability of several propagation scenarios. This model was used in the specification of HIPERLAN/2 and is based on measurements taken in different propagation scenarios, with and without line-of-sight. Depending on the scenarios, there are five kinds of channels: A,B,C,D and E.

In this dissertation we used model E and the relevant measurement settings were:

- outdoor 90x90 m^2 open area environment;
- the site taken surrounded by buildings;
- no line-of-sight;
- measurements taken for SISO systems.

$$h(t,\tau) = \sum_{p=1}^{L_p} \alpha_p(t) e^{j\varphi(t)} \delta(\tau - \tau_p) \tag{2.22}$$

It is assumed that each path contains an high number of subpaths and that the amplitude $\alpha_p(t)$ is Rayleigh distributed and its variance is η_p. The phase $\varphi(t)$ is uniformly distributed in $[0, 2\pi]$. The Doppler Power Spectrum is given by 2.20. The Power delay profiles of channels A and E can be found in the appendix (section C). Table 2.5 summarizes the main HIPERLAN/2 channel model parameters, where the Coherence Bandwidth was calculated according to the definition proposed in [44].

One of the main limitations of this model is the fact that it was designed for SISO systems. In order to extend this model for the MIMO case we assumed that the channels impulse response are independent with respect to each other, ie, if we use an antenna array such as an uniform linear array (ULA), its elements are assumed to be sufficiently spaced. By doing so, the fading

BRAN Model	Delay Spread(ns)	Maximum delay(τ_{max})	Coherence BW(MHz)
A	50	0.39	2.56
B	100	0.73	1.37
C	150	1.05	0.95
D	140	1.05	0.95
E	250	1.76	0.57

Table 2.2: Some HIPERLAN/2 channel model parameters [2].

at each antenna element can be assumed to be uncorrelated and we are allowed to generate M independent channels for each user.

It is worth mentioning that despite the ease of implementation of this model, it does not take into account the angular distribution with respect to the power delay spectrum, since the angle of computation is made independent of the power of each path.

The 3GPP/3GPP2 *Spatial Channel Model*(SCM) *Ah-Hoc Group*(AHG) proposed an alternative model that ensures space and time consistency, and can be used for MIMO systems. In contrast with the HIPERLAN/2 models, this one modulates the subpaths explicitly, i.e., the amplitudes, phases and coherence angles of each path are randomly generated using statistical distributions that are chosen according to the selected propagation scenario. This model was adopted by european projects such as MATRICE and 4MORE [2, 35]. It was adapted for 5GHz propagation scenarios. Unfortunately, this model is difficult to implement and is computationally intensive.

Chapter 3

MULTIANTENNA TECHNIQUES

The use of multiple antennas allows several channels to be created in space and is one of the most interesting and promising areas of recent innovation in wireless communications. The focus of this chapter is spatial diversity. In contrast to time and frequency diversity, which may require additional bandwidth, this kind of diversity only "costs" extra hardware and computational complexity.

In addition to providing spatial diversity, antenna arrays can be used to focus energy (beamforming) or create multiple parallel channels for carrying unique data streams (spatial multiplexing). When multiple antennas are used at both the transmitter and receiver, these three approaches are often collectively referred to as multiple input / multiple output (MIMO) communication.

This chapter begins with an introductory section where we introduce terms such as Array and Diversity Gain and develop a mathematical framework to model MIMO systems and compute their capacity. We proceed with MIMO System Capacity determinations for situations (channel knowledge at the transmitter and determinism/randomness of the channels). Here we present Ergodic Capacity and Outage Capacity as figures-of-merit to evaluate systems. Finally, we conclude by discussing multiple antenna schemes (open and close loop).

3.1 Preliminaries

In this section we introduce preliminary concepts such as *diversity gain*, *outage and ergodic capacities*, *array gain* and a *MIMO system model*.

3.1.1 Diversity Gain

Up to now, we have emphasized the multipath fading phenomenon as an inherent characteristic of the wireless medium channel. Given this physical reality, how do we make the communication process across the wireless channel into a reliable operation? The answer to this fundamental question lies in the use of *diversity*, which may be viewed as a form of redundancy. In particular, if several replicas of the information-bearing signal can be transmitted simultaneously over independently fading channels, then there is a good likelihood that at least one of the received signals will not be severely degraded by the channel fading. There are several methods for making such a provision. In the context of the material covered in this thesis, we may identify three approaches to diversity:

- Frequency diversity;
- Time (signal-repetition) diversity;
- Space diversity.

In frequency diversity, the information-bearing signal is transmitted by means of several carriers that are spaced sufficiently apart from each other to provide independent fading versions of the signal. This may be accomplished by choosing a frequency spacing equal to or larger than the coherence bandwidth of the channel.

In time diversity, the the information-bearing signal is transmitted in different time slots, with the interval between successive time slots being equal to or greater than the coherence time of the channel. If the interval is less than the coherence time of the channel, we can still get some diversity, but at the expense of performance. In any event, time diversity may be likened to the use of a repetition code for error control coding.

In space diversity, multiple transmit or receive antennas, or both, are used with the space between adjacent antennas being chosen so as to ensure the independence of possible fading events occurring in the channel. Space diversity is the topic that will deserve more attention in this chapter.

We point out that the efficiency of the approaches lays in the independency of the received samples.

3.1.2 Ergodic and Outage Capacities as figures of merit

It is common to use Ergodic and Outage Capacities to compare and contrast different systems.

Ergodic Capacity

The Ergodic Capacity is an important figure of merit to study communication systems. It is the ensemble average information rate over the distribution of the elements of the matrix H or ,stated another way, it is the capacity of the channel when every channel matrix H is an independent realization. This implies that it is the result of infinitely long measurements.

Outage Capacity

The outage Capacity is the capacity that is guaranteed with a certain level of reliability. We define $p\%$ outage capacity as the information rate that is guaranteed for $(100-p)\%$ of the channel realizations, ie, Prob$[C \leq C_{out}] = p\%$

Both the Outage and Ergodic Capacities can also be defined with respect to the cumulative density function (CDF) of the capacity for the system, as a function of the bit rate. The physical meaning of this representation relates to the fact that the channel response is a random variable. Sometimes it can be "good" whereas in the remaining ones, it can be distortive. As a result, the maximum attainable capacity for the system, which obviously depends on the channel response for that realization, will also be a random variable.

The CDF takes into account this fact and is created with a very large number of channel realizations, where each of them corresponds to a maximum bitrate per Hz. It establishes the correspondence between a given system capacity and the probability for which the system capacity for a particular channel realization will be less than that value. Figure 3.1 is an example of a CDF for a SISO and 1x2 SIMO, where the channels were considered to be Rayleigh distributed with zero mean, $E_b/N_o = 15$ dB and Path Loss is 6 dB. The code that was used to generate this figure can be found in the appendix A.

By definition, the $p\%$ outage capacity is the capacity for which the CDF is $p\%$ and the ergodic capacity is the capacity for which the CDF is 50%.

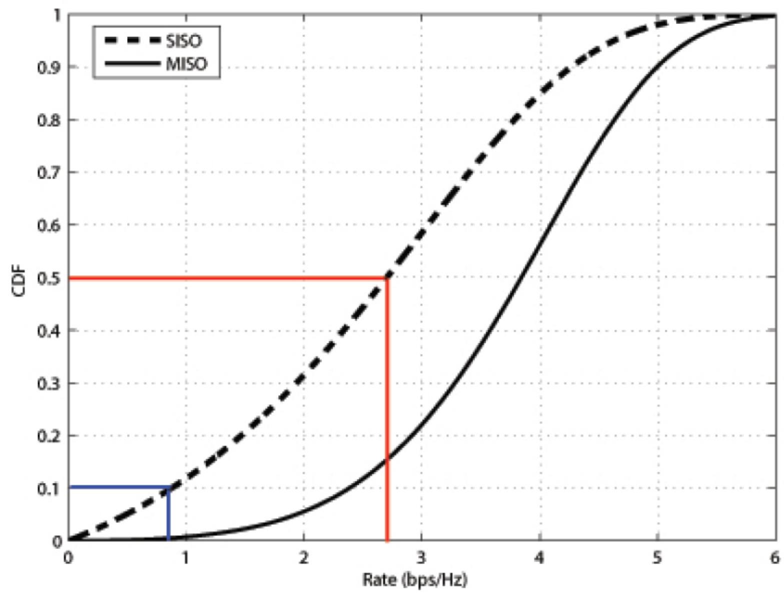


Figure 3.1: CDF representation.

Figure 3.1 makes it apparent that, for these channel conditions, the SIMO system perform better than the SISO one, since the ergodic and outage capacities of SISO are lower than SIMO.

3.1.3 Array Gain

Array Gain is the average increase in the signal-to-noise ratio (SNR) at the receiver that arises from coherent combining effect of multiple antennas at the receiver or transmitter or both. If the channel is known to the multiple antenna transmitter, the transmitter will weight the transmission with weights, depending on the channel coefficients, so that there is coherent combining at the single antenna receiver (SIMO case). The array gain in this case is called transmitter array gain. Alternatively, if we have only one antenna at the transmitter and no knowledge of the channel and a multiple antenna receiver, which has perfect knowledge of the channel, then the receiver can suitably weight the incoming signals so that they coherently add up at the output (combining), thereby enhancing the signal.

3.1.4 MIMO System Model

In order to develop a mathematical framework for the MIMO systems, we start by assuming that E_s is the transmit energy per bit, and that the channel is described by the matrix is $\mathbf{H}$. The dimension of $\mathbf{H}$ is a M_R x M_T matrix, where M_R and M_T are the number of receive and transmit antennas, respectively. Its rows are the channel frequency response between output i and input j, where $i = 1...M_T$ and $j = 1...M_T$ for each user k. $h_{k,ij}$ translate the attenuation and multipath fading phenomena.

With this, if we assume that $M_T = M_R = 2$, we write

$$\mathbf{H} = \begin{bmatrix} h_{k,11} & h_{k,12} \\ h_{k,21} & h_{k,22} \end{bmatrix}. \tag{3.1}$$

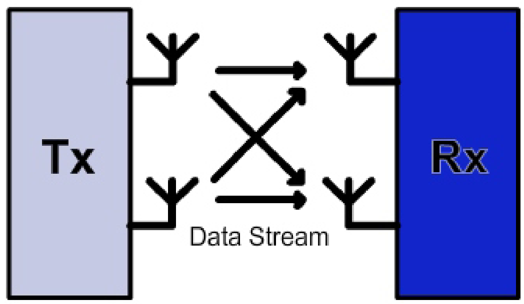


Figure 3.2: MIMO channel for $M_T = M_R = 2$.

The covariance matrix for transmit signal can be expressed as

$$\mathbf{R}_{ss} = \frac{E_s}{M_T}\mathbf{I}_{M_T} \tag{3.2}$$

where the channel impulse response is constrained by

$$M_T = \sum_{j=1}^{M_T} |h_{ij}|^2 \tag{3.3}$$

In **??**, the covariance matrix for receiver noise is found to be given by $\mathbf{R}_{nn} = \mathrm{E}\left\{\mathbf{n}\,\mathbf{n}^H\right\}$, where $(\cdot)^H$ denotes transpose and complex conjugation. When the noise is AWGN, $\mathbf{R}_{nn}$ simplifies to $N_0\,\mathbf{I}_{M_R}$.

If we assume maximum likelihood detection over M_R receive antennas, the received vector is

$$\mathbf{r = H\ s + n} \tag{3.4}$$

and the covariance matrix for the received signal is

$$\begin{aligned} \mathrm{E}\left\{\mathbf{r}\,\mathbf{r}^H\right\} &= \mathrm{E}\left\{(\mathbf{Hs}+\mathbf{n})\,(\mathbf{Hs}+\mathbf{n})^H\right\} && (3.5)\\ &= \mathrm{E}\left\{(\mathbf{Hs}+\mathbf{n})\left((\mathbf{sH})^H+\mathbf{n}^H\right)\right\} && (3.6)\\ &= \mathbf{H}^H\mathrm{E}\left\{\mathbf{s}\,\mathbf{s}^H\right\}\mathbf{H}^H+\mathrm{E}\left\{\mathbf{n}\,\mathbf{n}^H\right\} && (3.7)\\ &= \mathbf{R}_{rr}+\mathbf{R}_{nn} && (3.8) \end{aligned}$$

3.2 MIMO System Capacity

The system Capacity is defined as the maximum possible transmission rate such that the probability of error is arbitrarily small. Shannon's Theorem states that the capacity relates to the transmitted signal, the number of transmit antennas and the channel quality **??**:

$$C = \max_{\mathrm{Tr}[\mathbf{R}_{ss}]=M_T} \log_2 \det\left(\mathbf{I}_{M_R} + \frac{E_S}{M_T N_0}\mathbf{H}\ \mathbf{R}\left\{\mathbf{s}\,\mathbf{s}^H\right\}\mathbf{H}^H\right)\ bps/Hz \tag{3.9}$$

The Capacity C is also called error-free spectral efficiency or data rate per unit bandwidth(BW) that can be sustained reliably over the MIMO link. Thus if our bandwidth is W Hz, the maximum achievable data rate over this BW using MIMO techniques is WC bits/s.

3.2.1 Channel Unknown to the transmitter

When the channel is unknown to the transmitter, Eq.3.9 simplifies to

$$C = \sum_{i=1}^{r} \log_2\left(1 + \frac{E_S}{M_T N_0}\lambda_i\right) \tag{3.10}$$

where r is the channel rank and λ_i is the eigenvalue of $\mathbf{H}\ \mathbf{H}^H$ and is given by $\lambda_i = \sum_{j=1}^{M_T} |h_{i,j}|^2$.

When power distribution is uniform and the channel matrix $\mathbf{H}$ is orthogonal, it can be shown that the capacity is maximized for $\lambda_i = \lambda_j = \beta/M,\ i,j = 1,2...M$, where $M = M_T = M_R$ and $\beta = \sum_{i=1}^{M_R} \lambda_i$. This capacity is

$$\begin{aligned} C &= M\log_2(1+\beta\tfrac{E_S}{N_0 M^2}) \\ &= M\log_2(1+\tfrac{E_S}{N_0}),\ \text{ if } \mathbf{H} \text{ is diagonal and unitary.} \end{aligned} \tag{3.11}$$

It can be shown that the capacity of an orthogonal MIMO channel is $\min\{M_T, M_R\}$ times larger than the capacity of a SISO channel[54].

3.2.2 Channel Known to the transmitter

It is possible by various means to learn the channel state information (CSI) at the transmitter. The channel knowledge can be used to distribute the energy or power across space (antennas) and frequency (subchannels) so as to maximize spectral efficiency. In such an event the capacity can be increased by resorting to the so-called "water filling principle", by assigning various levels of transmitted power to various transmitting antennas. This power is assigned on the basis that the better the channel gets, the more power it gets and vice versa.

Since water filling is applicable only to purely orthogonal channels, it becomes necessary to convert a frequency selective channel into a set of parallel frequency flat channels, which are orthogonal to each other. This is an optimal allocation algorithm.

3.2.3 Deterministic Channels

SIMO Channel Capacity

In a SIMO channel, $M_T = 1$ and there are M_R receive antennas. When the channel is unknown to the transmitter, the capacity is given by

$$C = \log_2\left(1 + \sum_{i=1}^{M_R} |h_i|^2 \frac{E_S}{N_0}\right) \tag{3.12}$$

If it is further assumed that the channel elements are all normalized to 1, then Eq. 3.12 simplifies to

$$C = \log_2\left(1 + M_R \frac{E_S}{N_0}\right) \tag{3.13}$$

This is a remarkable result, since it shows that antenna arrays are benefic, even when there is no spatial diversity. From Eq. 3.12, we conclude that the system achieves a spatial diversity gain of M_R relative to the SISO case. If $M_R = 4$ and $SNR = 10dB$, $C_{SIMO} = 5.528$ bit/s/Hz.

MISO Channel Capacity

In MISO channels, when there is not channel knowledge at the transmitter (as it happens with the systems that use Alamuti codes), $M_R = 1$ and there are M_T transmit antennas, we obtain

$$C = \log_2\left(1 + \sum_{j=1}^{M_T} |h_j|^2 \frac{E_s}{M_T N_0}\right) \tag{3.14}$$

If the channel coefficients are equal and normalized as $\sum_{j=1}^{M_T} |h_j|^2 = M_T$, then the capacity for the MISO case becomes the same as the one for the SISO case, as is presented in Eq. 3.15

$$C = \log_2\left(1 + \frac{E_s}{N_0}\right) \tag{3.15}$$

In contrast, when the receiver has CSI and weights the transmission with weights depending on the channel coefficients, coherent combining will take place at the receiver and the capacity will be similar to Eq. 3.12 (the only difference being the fact that $\mathbf{H}\,\mathbf{H}^H$ is a summation from 1 to M_R and not to M_T).

If the channel coefficients are equal and normalized as $\sum_{j=1}^{M_T} |h_j|^2 = M_T$, the capacity becomes

$$C = \log_2\left(1 + M_T \frac{E_s}{N_0}\right) \tag{3.16}$$

Despite SIMO and MISO achieve equal gains for $M = M_T = M_R$, MISO is more common in practise. The reason is that it allows the mobile terminals to be simple.

3.3 Multiple antenna schemes

Here we will focus our attention on spatial diversity. The underlying idea is to achieve diversity gains by increasing the number of transmit and/or receive antennas and without using any additional bandwidth or transmit power. Spatial diversity improves reliability by a factor of 10-100 [7]. Depending on which end of the wireless link is equipped with multiple antennas, we may identify three forms of space diversity:

1. **Receive diversity**, which involves the use of a single transmit antenna and multiple receive antennas. It relates to the SIMO (*single-input, multiple-output*) channel.

2. **Transmit diversity**, which involves the use of multiple transmit antennas and a single receive antenna. It relates to the MISO (*multiple-input, single-output*) channel.

3. **Diversity on both transmit and receive**, which combines the use multiple antennas at both the transmitter and receiver. This third form of space diversity includes transmit and receive diversity as special cases. In the literature, a wireless channel using multiple antennas at both ends is commonly referred to as a *multiple-input, multiple-output (MIMO) channel.*

The main purposes of using MIMO are:

1. **Increase maximum attainable bit rate and capacity.** Given fixed values of transmit power and channel bandwidth, MIMO technology offers a sophisticated approach to exchange increased system complexity for *boosting* the channel capacity(i.e,. the spectral efficiency of the channel, measured in bits per second per hertz) up to a value significantly higher than that attainable by the SISO channel. More specifically, when the wireless communication environment is endowed with rich Rayleigh scattering, the MIMO channel capacity is roughly proportional to the number of transmit or receive antennas, whichever is smaller. That is to say, we have a spectacular increase in spectral efficiency, with the channel capacity being roughly doubled by doubling the number of antennas at both ends of the link.

2. **Increase the system reliability (thereby decreasing BER);**

3. **Increase the coverage area;**

4. **Decrease the required transmit power.**

However, these four desirable attributes may be antagonist; for example, an increase in the bit rate implies an increase in the transmit power or BER. The type and amount of antennas that are chosen reflects the importance that the systems designer gave to each of these aspects, as well as to cost and space considerations.

Despite the cost associated with extra antennas and higher RF chain complexity, the array gains are so huge that the importance of arrays in future broadband systems in unquestionable.

Apart from providing spatial diversity, the antenna arrays can focus energy (beamforming) or create several parallel "streams" to carry individual information (spatial multiplexing).

Until the end of this section, we discuss two main groups of multiantenna techniques (see Fig. 3.3): open-loop and closed-loop. With Open Loop MIMO, the communications channel does not utilize explicit information regarding the propagation channel. Common Open Loop MIMO techniques include pure spatial diversity schemes (transmit and receive diversity, in sections 3.3.2 and 3.3.1, respectively) and Spatial Multiplexing (section 3.3.3). With Closed Loop MIMO, the transmitter collects information regarding the channel to optimize communications to the intended receiver. Closed Loop MIMO typically utilizes digital signal processing techniques to electrically focus the beam pattern, leading to the shorthand name for this approach - beamforming (section 3.3.2).

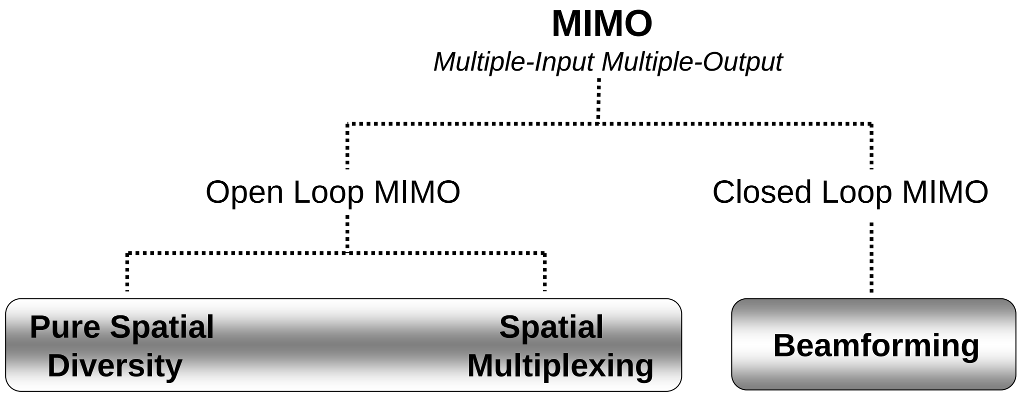


Figure 3.3: Multiantenna technology organization chart.

3.3.1 Receive diversity

Pure spatial diversity schemes yield no direct increase of the transmitted bit rate. Here, the transmission of M symbols requires $M_T \geq M$ consecutive channel uses.

In this section we will discuss three kinds of diversity combining methods that take place at the receiver side. They differ in complexity and also performance. As depicted in fig.3.4, a SIMO receiver can be thought of as a cascade of two functional blocks; the first one consists of N_R SISO parallel receivers, which receive the input signals $x_1(t)...x_{N_R}(t)$. Those signals are assumed to be uncorrelated and its SNR is $\gamma_1...\gamma_{N_R}$. The second stage is a combining circuit. Its purpose is to combine the received signals so as to maximize the SNR of the output signal.

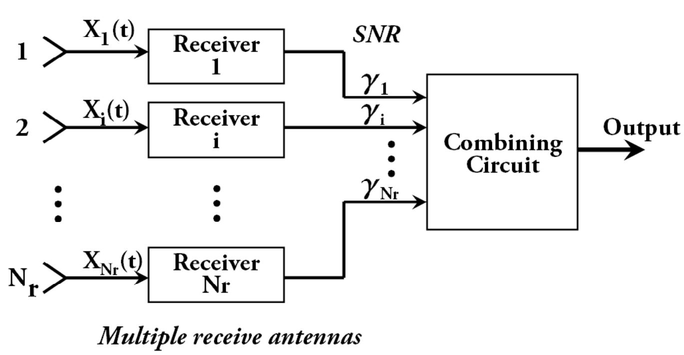


Figure 3.4: Block Diagram of a SIMO Receiver.

As discussed in [20, 28], there are three main diversity combining methods – selection combining, maximal-ratio combining, and equal-gain combining. They differ in complexity and performance.

Selection Combining

In conceptual terms, selection combining is the simplest combining method. Given N_R receiver outputs produced by a common transmitted signal, the logic circuit *selects* the particular receiver output with the largest signal-to-noise ratio as the received signal. It can be shown that the frequency-flat, slowly fading Rayleigh channel is modified through the use of of selection combining into a Gaussian channel, provided that the number N_r of diversity channels is sufficiently large. Realizing that a Gaussian channel ia a "digital communication theorist's dream", we can now see the practical benefit of using selection combining.

The selection combining procedure just described is relatively straightforward to implement. However, from a performance point of view, it is not optimum, in that it ignores the information available from all the diversity branches except for the particular branch that produces the largest instantaneous power of its own demodulated signal. This limitation is mitigated by the *maximal-ratio combiner.*

Maximum Ratio Combining

In maximal ratio combining (MRC), the signals from all the N_R branches are weighted according to their individual SNR's and then summed. Here the individual signals need to be brought into phase alignment before summing. If the signals are $\mathbf{r}_i$ from each branch, and each branch i has a gain G_i, then

$$\mathbf{r}_{M_R} = \sum_{i=1}^{N_R} G_i \mathbf{r}_i \tag{3.17}$$

where $r_i = h_i s_i + \nu_i$, $s_i = 2E_s$ being the transmitted signal, ν_i is the noise in each branch with a power spectral density of $2N_0$ and h_i is the channel coefficient.

Therefore, from 3.17 we get

$$\mathbf{r}_{M_R} = \sum_{i=1}^{N_R} G_i h_i s_i + \sum_{i=1}^{N_R} G_i \nu_i \tag{3.18}$$

We obtain, if $G_i = h_i^*$ for all i (perfect channel knowledge)

$$\gamma_{M_R} = \frac{E_s}{N_0} \sum_{i=1}^{N_R} |G_i|^2 \tag{3.19}$$

Note that $\gamma_{M_R} = \frac{E_s}{N_0}|G_i|^2$ is the SNR per antenna, Eq.3.19 is nothing more than the sum of the SNR's of each antenna, which means that γ_{M_R} can be large even if the individual SNR's are small. This makes MRC is a powerful technique to improve signal reception in SIMO channels, specially when we have perfect channel knowledge.

The transmission chains that were implemented in the scope of this thesis employ MRC at the BS side. For further information, the interested reader is referred to section 5.4.

Equal-Gain Combining

It is the same as MRC but with equal weighting for all branches. Hence, in this sense, it is sub-optimal. The performance is marginally inferior to MRC, but the complexities of implementation are much less.

Thus, MRC is the optimum scheme in the absence of interference. The drawback of combining the signals received over all antennas is that they have to be demodulated in parallel. For this reason, antenna selection based on a rough estimate of the signal powers at the antennas or equal gain combining can reduce the complexity of the receiver considerably.

3.3.2 Transmit diversity

Very efficient transmit diversity schemes are obtained by *Space-Time Coding (STC)*. The idea is to encode an original data stream so that redundant information is transmitted over the different antenna branches and in consecutive symbol periods. The receiver is not necessarily equipped with an antenna array, since STC allow decoding in serial (i.e., in the time dimension). It makes them attractive for small mobile receivers. It can be shown that the STC schemes achieve the same capacity as the MISO schemes [17].

Because of complexity and efficiency reasons, the most common STC schemes are *Space-Time Block Coding (STBC)* with Alamuti codes to distinguish the antennas [17]. STBCs are designed as pure diversity schemes and provide no coding gain. Yet, they are much simpler to implement. A well known scheme was proposed by Alamuti in [5]. This scheme allows to achieve a diversity gain of 2 for $M_R = 1$ with rate 1. Thanks to the orthogonality of this STBC, the receiver can separate the two symbols by a simple linear combiner. From a practical point of view, this scheme is very robust as it transmits the whole information even when one of the two branches is inactive. When this happens, the scheme simply falls back to SISO transmission. Another advantage of Alamuti codes is the fact that they do not require channel knowledge at the transmitter side.

Beamforming

The *beamforming* techniques are an alternative to increase the systems reliability without increasing the transmission energy, and they do so by suppressing or canceling interfering signals. Fig. 3.5 illustrates the usage of antenna elements to direct the beampattern to the desired destination while avoiding interferers.

In contrast to transmit diversity, here the available antenna elements are used to adjust the power of the incoming signals. It is achieved based on computations that take into account the physics of the incoming waves (direction of arrival, DOA) or in mathematical sense (eigenbeamforming). Beamforming based on DOA focus the energy by regulating the weights of each antenna element according to a given criteria (commonly SNR maximization or minimization of minimum square errors). DOA techniques are effective in LOS scenarios. In the remaining ones, eigenbeamforming achieve better results [45].

3.3.3 Spatial Multiplexing

Spatial Multiplexing consists of splitting the input signal (high bit rate) into N_t independent signals whose bit rate is lower.

Figure 3.6 depicts a spatial multiplexing scheme. Let the bit rate of the incoming signal be $R_{in} = R\ \min(N_t, N_r)$ and assume that the number of transmit and receive antennas is equal, i.e., $N_t = N_r$. The incoming bits (high bit rate) are demultiplexed by a S/P converter and the bit rate is reduced by a factor of $\min(N_t, N_r)$ (rate per stream is R). Then these $\min(N_t, N_r)$ signals cross the channel and are received by N_t antennas. These antennas are wired to a DSP unit that

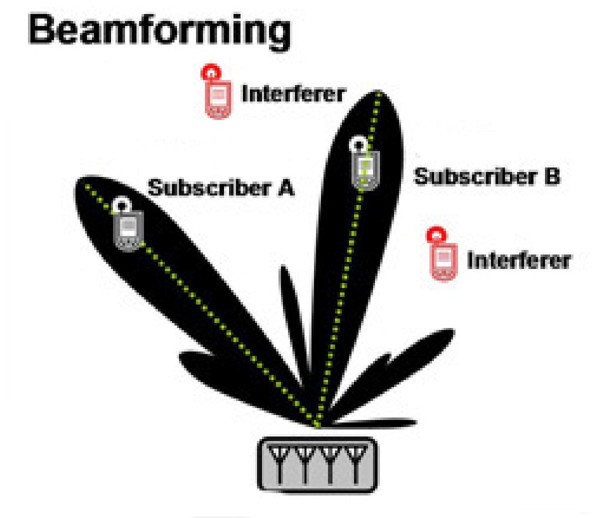


Figure 3.5: Usage of antenna elements to direct the beampattern and avoid interferers[58].

puts them "working as a whole"; this is the only way to differentiate the signals at each transmit antenna and to suppress interferes.

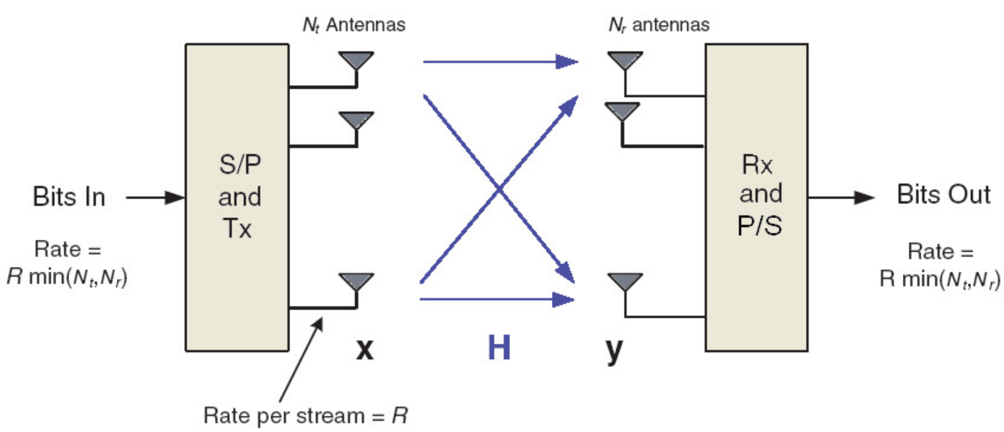


Figure 3.6: A spatial multiplexing scheme MIMO has high capacity because it transmits the signals that result from multiplexing the incoming signal.[7]

This is the most interesting kind of MIMO for achieving high bit rates. Assuming that these signals are decoded correctly, the spectral efficiency is increased by a factor of $\min(N_t, N_r)$.

One could easily think that the addition of antennas would increase the reliability of the channel for broadband access indefinitely, but that is not true. Some of the reasons are:

- Extra coding: the existence of multiple antennas, requires extra coding, which means a decrease in coding efficiency;
- Interference between transmit antennas;
- Channel estimation errors.

In the optimum case of space multiplexing, the capacity (or maximum bit rate) grows in proportion to $\min(N_t, N_r)\log(1+SNR)$ when the SNR is high. Otherwise, the best strategy is to send the same signal at all antennas and use precoding. In this case, the capacity is much smaller, but it continues to grow in proportion to $\alpha\ \min(N_t, N_r)$. In any case, MIMO performance exceeds space-time coding, in which the bit rate grows according to N_r (most optimistic case). The most famous example for spatial multiplexing is the BLAST (Bell Labs Layered Space-Time) architecture proposed in [15].

Chapter 4

MULTICARRIER SYSTEMS

Orthogonal frequency division multiplexing is an example of a multicarrier system and achieves frequency diversity through the use of multicarrier modulation. OFDM systems transmit information data in many subcarriers, where sub-carriers are orthogonal to each other so that the spectrum efficiency may be enhanced. OFDM can be easily implemented by the IFFT (inverse fast Fourier transform) and FFT (fast Fourier Transform) process in digital domain, and has properties such as high-speed broadband transmission, robustness to multipath interference and frequency selective fading and high spectral efficiency. It is also worth mentioning that the OFDM modulation scheme can be used to make a multiple access technique, resulting in orthogonal frequency division multiple access (OFDMA).

Such advantages justify the belief that, in the same way as CDMA enabled 3G, orthogonal frequency division techniques such as OFDM and OFDMA will be among the key technologies behind the physical layer of 4G.

This chapter is organized as follows: section 4.1 discusses the OFDM modulation principles and then section 4.2 continues with its adaptation to a multiple access technique (OFDMA). Finally, section 4.3 concludes with a brief description of the physical layer of a commercial application of OFMA, which is the Mobile WiMAX. The reader that wants to broaden its knowledge with respect to these topics is referred to [20] and [7].

4.1 Orthogonal Frequency Division Multiplexing

As the bandwidth increases and systems move towards more time-dispersive environments (e.g. offices), solutions that rely on equalizers or a rake receivers to overcome the channel influence become highly complex. For such cases, the use of *multiple carriers systems*, such as *Orthogonal Frequency Division Multiplexing* (OFDM) was proposed, whose idea is to split the broadband communication channel into several narrowband orthogonal ones. Its major strengths include efficient spectrum utilization and capacity to handle high bit rates.

The concept of using the discrete Fourier transform (DFT) as a part of the digital modulation/demodulation in the transmitter and receiver part of the wireless system to achieve parallel data transmission is not new. As a matter of fact it was proposed at roughly 3 decades ago ([56]), but by then the technology wasn't mature enough to face the associated complexity and cost issues. Some of its advantages are high spectral efficiency and robustness against multipath distortion.

Nowadays it is the basis for various wireless LAN and PAN physical layers; IEEE 802.11a, 11g, 11n, DVB, DAB, WiMax and UWB (MB-OFDM). Its future is promising and studies envision OFDM for 4th generation cellular networks.

For a comprehensive description of orthogonal frequency division multiplexing (OFDM) and its applications in different wireless systems, the reader is referred to [23, 20, 41] and [7]. Hereto we will treat the basic multicarrier concepts.

Latter on this dissertation we will apply a multicarrier technique (OFDMA) for establishing communication between mobile users and a base station.

OFDM as a solution for high bit rates

OFDM is meant to provide wireless data links supporting high rates (up to 54 megabits/s) to link workstations, laptops, printers and personal digital assistants to a network access node without the expense of cabling and *with* the thread of multipath.

To develop insight into the underlying communication-theoric operations carried out in the OFDM system, consider Fig. 4.1, which is a functional diagram of an OFDM encoder.

The sequence of operations is as follows:

- FEC decoding (optional);
- M-ary demodulation;
- P/S conversion;
- FFT with L-points;
- S/P conversion;
- A/D conversion;
- RF up-conversion.

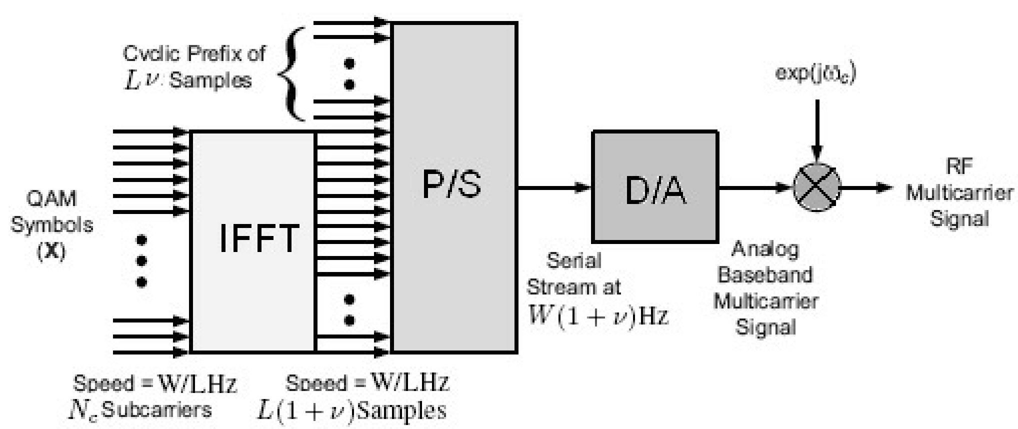


Figure 4.1: Simplified implementation of a OFDM transmitter.[7]

In this example, the maximum bit rate is $R = \frac{B}{N_c}\frac{N_c \log_2(M)}{1+\nu}$.
All subcarriers(N_c) have bandwidth W/L has $\log_2(M)$ data bits. The cyclic prefix extension is translated by the inclusion of a $1+\nu$ penalty factor.

Forward error correction and digital modulation

OFDM starts with forward error correction (FEC) coding. Its function is to insert redundance in the signal so that it becomes more robust against noise. Typical FEC algorithms include CRC (cyclic redundancy check) or Viterbi codes. FEC trades bandwidth efficiency for robustness and is optional.

The next step is M–ary digital modulation. In a typical OFDM scenario the transmission bandwidth is smaller than half the target throughput. As a result M is often made 4 (16–QAM), like in well established standards such as the 802.11a [24]. With 16–QAM the in-phase and quadrature channels are independently modulated with a four-level signalling derived from the incoming binary data stream. Nominally, the levels are ± 1 and ± 3. Since the points are equally likely to be selected, $\log_2 M = 4$ bits are transmitted at each symbol time. When the bit rates are not so demanding, BPSK can be preferred(M= 2). Other values for M might be difficult to achieve due to hardware complexity.

Multicarrier modulation

Now comes the point that distinguishes OFDM from the conventional systems: it is a for of *multicarrier modulation*. The whole system bandwidth is subdivided into several parallel narrow N_c subbands. In particular, instead of sending a very high bit rate over one carrier, it is preferred to send several lower bit rates R/N_c over each of the N_c distinct *subcarriers*.

This strategy is an effective measure to overcome multipath. By demultiplexing the data set into N_c parallel subcarriers, each of them will run N_c slower than the incoming data. Figure 4.2 illustrates a possible scenario for the single carrier case. At high bit rates, the largest channel delay is comparable to the symbol time of the system. In this way, echoes can have severe effects such as distortion and even signal cancellation. However, as it can also be seen in Fig. 4.2, if subcarriers are used, the symbol period is allowed to increase and the channel will only corrupt a limited portion at each symbol period.

Stated another way, the coherence bandwidth for each subband is made large in comparison to the system bandwidth.

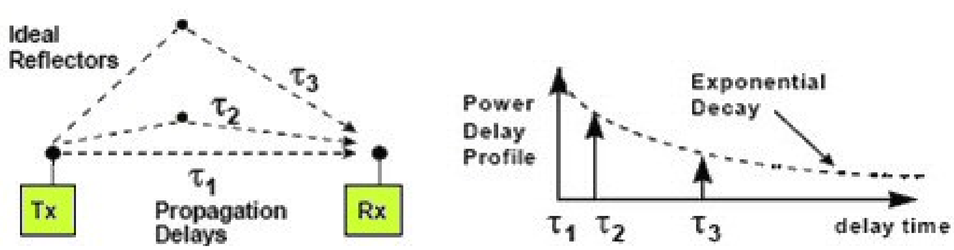


Figure 4.2: Channel dispertion due to multipath.

Digital Mudulation owns its proctical and efficient implementation to the availability of *discrete Fourier transform*(DFT) blocks.

Due to the properties of the IDFT, the subchannels are shaped like $\sin x/x$. An example of spectrum of three OFDM subcarriers is shown in Fig. 4.3, which shows that spectra are partly overlapping, significantly increasing the spectral efficiency as compared to conventional non-overlapping multi-carrier systems. It is also clear that the separation of the different carriers cannot be carried out by bandpass filtering. Therefore, baseband processing is applied which exploits the *orthogonality* property of the subcarriers. This property is apparent from Fig. 4.3, where at the maximum of one subcarrier all other carriers have a zero amplitude.

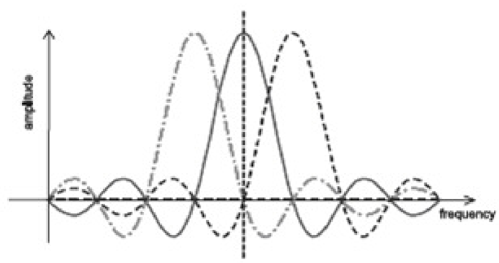


Figure 4.3: 3 subcarrier OFDM symbol in the spectral domain.[7]

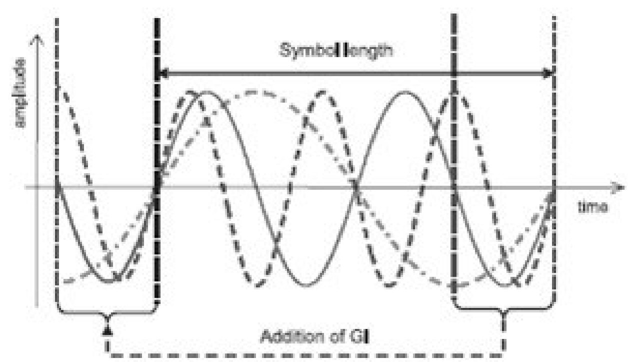


Figure 4.4: OFDM symbol in the time domain, showing the inclusion of a GI prefix.[7]

P/S conversion and cyclic prefix inclusion

The following step is P/S conversion, where a *cyclic prefix* is normally added to the signal(Fig. 4.4). To increase the robustness of the OFDM system against ISI caused by multipath propagation, *guard intervals* are included in the serial data stream of the OFDM transmitter so as to overcome the effect of intersymbol interference(ISI) and inter-channel interference (ICI) produced by signal transmission over the wireless channel and impairments.

The pertinent OFDM symbol is cyclically extended in each guard interval. Specifically, the *cyclic extension* of an OFDM symbol is the periodic extension of the DFT output, as shown by

$$s(-k) = s(N_c - k) \text{ for } k = 1, 2, \ldots, \nu \tag{4.1}$$

where N_c is the number of subchannels in the OFDM system and ν is the duration of the baseband impulse response of the wireless channel. The condition described in Eq. 4.1 is called a *cyclic prefix*. Clearly, inclusion of cyclic prefixes increases the bandwidth of OFDM transmission bandwidth. Since the addition of a GI decreases the effective data rate of the system, the ratio between the number of carriers Nc, which is equal to the symbol length in samples, and the GI length ν is an important design parameter. It must be chosen in a tradeoff between ISI robustness and effective data rate.

Having stuffed the guard intervals in the P/S converter in the transmitter with cyclic prefixes, the guard intervals (and with them, cyclic prefixes) are removed in the S/P converter in the receiver *before* the conversion takes place. Then the output of the S/P converter is in the correct form for discrete Fourier transformation.

D/A conversion and up–convertion

Finally D/A takes place and the incoming data stream is up–converted to RF (frequency f_{RF}) over the wireless (multipath) channel.

OFDM–receiver

The OFDM–receiver follows a sequence of operations in the reverse order of these performed in the transmitter of Fig. 4.1. Specifically, to recover the original input binary data stream, the received data stream, the received signal is passed through the following processors:

- RF down-conversion;
- A/D converter;
- S/P converter;
- L-point FFT algorithm;
- P/S converter;
- M–ary demodulator;
- FEC decoder.

In particular, focusing on the transmission part we may make 2 statements:

1. *The subcarriers constitute an orthogonal set.*
2. *The complex modulated (heterodyned) signals are multiplexed in the frequency domain.*

Put together, these two statements therefore justify referring to the communication system of Fig. 4.1 as an *orthogonal frequency-division (OFDM) system.*

Issues concerning OFDM implementation

OFDM techniques also face several issues. First, there is the problem associated with OFDM signals having a high peak-to-average ratio that causes nonlinearities and clipping distortion([21]). This can lead to power inefficiencies that need to be countered. Second, OFDM signals are very susceptible to phase noise and frequency dispersion, and the design must mitigate these imperfections. This also makes it critical to have accurate synchronization. It involves two kinds of synchronism:

- **Time Synchronization** – The temporal offset of the OFDM symbols is not very harmful. Thus the time synchronization requirements are relaxed.

- **Frequency Synchronization** – The requirements for frequency synchronization are more strict than for time synchronization, because detection requires that symbols are orthogonal in the frequency domain.

There are other factors that must be considered while designing OFDM systems:

- Available Bandwidth: the bigger the bandwidth, the higher the number of subcarriers that can be used with a reasonable cyclic prefix;

- Bit-rate: The system must be able to unsure the minimum system bit rates;

- Delay Spread: the maximum delay must be known in order to choose an appropriate value for the cyclic-prefix;

- Number of Subcarriers: A high number of sub-carriers is good to combat multipath effectively. A downside, however, is that, as the number of subcarriers increases, the synchronization at the receiver becomes complex.

- Symbol Period and CP length: the systems designer must choose appropriate values in order to avoid wasting bandwidth;

- FEC coding: FEC avoids errors at the expense of inserting redundancy in the message;

4.1.1 Advantages for mobile systems communications

OFDM has countless advantages over other solutions for high-speed transmission, such as:

Reduced computational complexity: OFDM can be easily implemented by using FFT (Fast Fourier Transformation) / IFFT (Inverse Fast Fourier Transformation).

Exploitation of frequency diversity: OFDM facilitates coding and interleaving across subcarriers in the frequency domain, which can provide robustness against burst errors caused by portions of the transmitted spectrum undergoing deep fades.

Use as a multiaccess scheme: OFDM can be used as a multiaccess scheme, where different tones are partitioned among multiple users. This scheme is referred to as *Orthogonal Frequency Division Multiple Access (OFDMA)* and is exploited in mobile WiMAX. This scheme also offers the ability to provide fine granularity in channel allocation. In relatively slow time-varying channels, it is possible to significantly enhance the capacity by adapting the data rate per subscriber according to the signal-to-noise ratio of that particular subcarrier.

Robust against narrowband interference: OFDM is relatively robust against narrowband interference, since such interference affects only a fraction of the subcarriers.

4.1.2 Turning multipath into an advantage

As mentioned above, the influence of the ISI caused by multipath propagation is removed at the receiver, when the guard interval (GI) is chosen long enough. The other effect of multipath, the frequency selective fading, remains. Now, however, since the bandwidth is subdivided into parallel carriers, the subcarrier bandwidth is smaller than the channel coherence bandwidth. That is why the channel can be regarded as frequency flat for a certain carrier.

What is most remarkable about OFDM is that it can be used as a baseline for other schemes. Two examples are coded-OFDM and OFDMA:

- **Channel Coding:** OFDM robustness against frequency selectivity can be improved using channel coding. Here the bits are coded in a block basis and spread across the various subcarriers using interleaving. Because blocks are spread among several subcarriers, the probability that an entire received block is wrong is reduced.

- **OFDMA:** OFDM multicarrier technique can be used for multiple access. In contrast with conventional OFDM, in which all subcarriers are given to a single user, in OFDMA they are distributed among the various users.

4.2 Orthogonal Frequency Division Multiple Access

In contrast to OFDM, where all subcarriers in a given time slot are allocated to a single user, in *Orthogonal frequency division multiple access* the subcarriers are subdivided among the various users. The main advantages that come from employing OFDMA are the possibility of exploring multiuser diversity, adaptive modulation and more freedom with respect to resource allocation.

This section will discuss the main ideas behind OFDMA in the following way: section 4.2.1 explains what it is meant bymultiple access strategies for OFDM; section 4.2.2 discusses subchannel allocation possibilities; and section 4.2.3 mentions some OFDMA implementation issues.

4.2.1 Multiple access strategies for OFDM

There are many possibilities for allocating resources in a multiple access strategy that is based on *Orthogonal frequency division multiplexing*. The applicability of this scheme is depicted in fig. 4.5, were we can see that several users are allowed to receive data simultaneously.

The first and most fundamental question behind a multiple access strategy: *how to to provide orthogonal and interference-free transmission channels, for each active connection?* and there are several possible answers. The most usual way of dividing the available dimensions between multiple users is through multiplexing in the frequency, time or code domains:

- **FDMA** – each user gets just one portion of the total available system bandwidth.

- **TDMA** – each user gets just one temporal slot, whether the slot attribution is *on-demand* or in a fixed rotation fashion. The wireless TDMA systems always end up using FDMA in some way, since it is not always possible to use the whole spectrum.

- **CDMA** – each user shares its bandwidth and slots with other users. The different users's data is separated by pseudo-noise(PN) codes, which are orthogonal to each other.

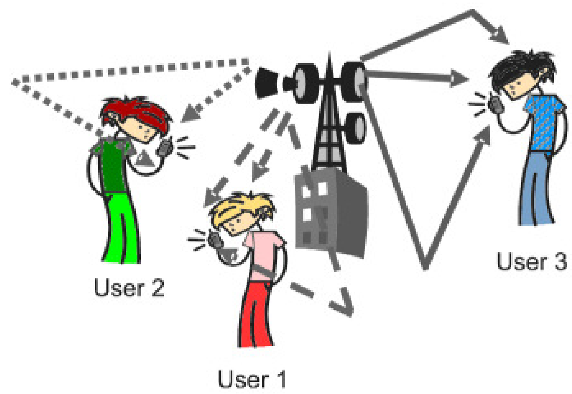


Figure 4.5: Multiuser scenario for an OFDMA system: users communicate simultaneously [48].

OFDMA results from a combination of TDMA-FDMA. Thus, it involves sharing time slots and subcarriers. Fig. 4.6 represents an OFDMA symbol. As it happens in the OFDM case, each temporal slot corresponds to a symbol and each symbol is a combination of subcarriers. Nevertheless, in contrast to OFDM, where each symbol is allocated to a single user, in OFDMA the subcarriers and slots belong to several users.

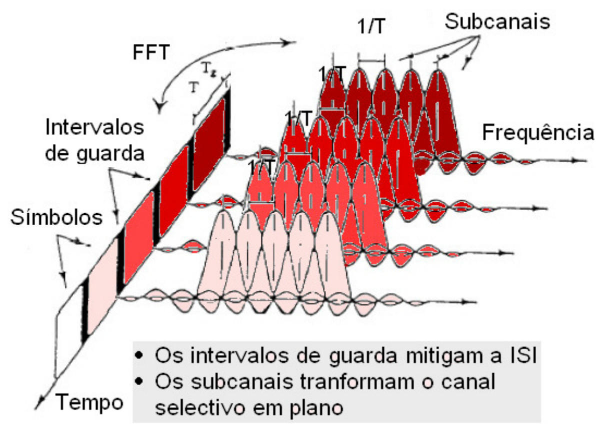


Figure 4.6: OFDMA symbol representation in the frequency and time domains.

4.2.2 Subchannel allocation

Despite the conceptual straightforwardness of OFDMA, its implementation hides issues whose solutions have a definite impact on the final system performance. As an example we can mention the allocation algorithms that are used to distribute the available subcarriers by the users and the decision criteria.

As it can be seen in sections 4.2.2 and 4.2.2, the resource allocation flexibility that OFDMA offers allows the system designers to explore frequency or multiuser diversity, respectively.

Exploring frequency diversity

Figure 4.7 represents this allocation scenario for a 3 user system, where the block of subcarriers are interleaved across the spectrum. Here we can observe that there are subcarriers allocated at regions where the channel frequency response is low, but there are also others where the subcarriers are is better.

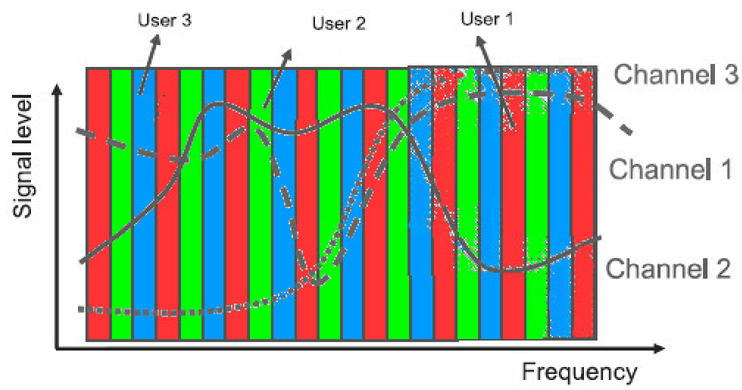


Figure 4.7: An OFDMA scheme that explores frequency diversity.[48]

The subchannels which are spaced apart benefit from frequency diversity, and this is particularly interesting for mobile applications.

Multiuser diversity and adaptive modulation

The allocation schemes which are based on contiguous subcarriers do not explore diversity in the frequency domain, but they make up for that by exploring multiuser diversity instead. Such schemes do so by allocating subchannels based on the channel frequency response.

Since all users have a specific location, the transmitted/received signals by the users will experience different channel responses. For this reason, the impulse responses that correspond to the different users will differ. In OFDMA, we can take this to our advantage by allocating subchannels to the users whose channel transfer function is higher. But that requires channel state information knowledge and also algorithms that regulate the subcarrier attribution (ie, increased hardware complexity). The crucial aspect multiuser diversity gains are the number of users within the OFDMA system. The allocation is based on the channel frequency response that the users experience. So, the more users the system have, the higher the probability that a set of channels is *good*.

A possible OFDMA scheme is illustrated in Fig. 4.8. Here we assume a 3 user system, where the block of subcarriers at the left is given to user 1, the middle ones to user 2 and the remaining ones to user 3.

Note that, despite channel 1 has its maximum at the subcarrier block at the right, we allocate it to user 3 and we do so because we are aiming at system optimality. In fact, channel 3 is weaker at the left block, where channel 1 is still acceptable.

The advantages of this procedure are the facts that several users can receive/transmit simultaneously and the channel transfer function will be optimized for each combination of users. In

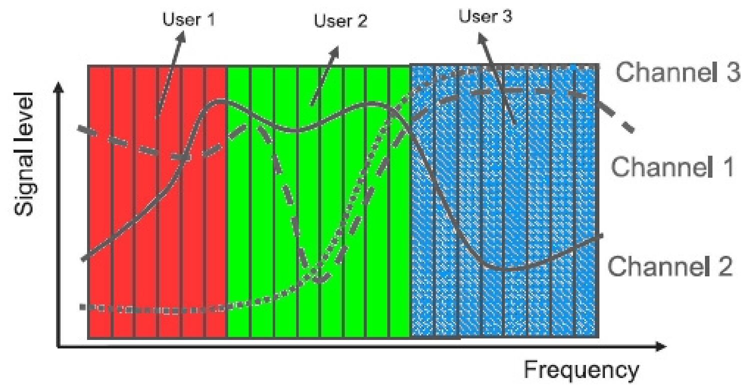


Figure 4.8: An OFDMA scheme that explores multiuser diversity. [48]

fact, if the allocation algorithms prioritize users whose subchannels maximize the received SINR, then multiuser diversity yields considerable gains in the whole system capacity.

And there are several subcarrier allocation algorithms; some try to increase the system capacity by *always* giving priority to the user with the best channel. The reader should note that ,despite its optimality from a system capacity perspective, this family of algorithms are rather "blind" and might lead to *starvation*). In the other extreme, we have the other kind of algorithms that focus on fairness, by minimizing the maximum time that each user has to wait for a subchannel. In practise, is often more desirable to try to find the best compromise between fairness and performance. The WiMAX standard, for example, does not specify the allocation algorithms, in an attempt to stimulate competitiveness between the industry.

Besides the possibility of exploring multiuser diversity, multicarrier modulation also allows the system designers to explore *adaptive modulation and coding* (AMC). As a matter of fact, the allocation algorithms are frequently associated with the selection of the *right* modulation and coding. Given a receiver that only demodulates signals whose SINR is above a certain level, AMC selects the most spectrally efficient modulation strategy at the transmitter side (ex.: BPSK, QPSK, 64-QAM, etc). If the SINR requirements are strict, the modulation will have to be robust at the expense of spectral efficiency. In the opposite side, OFDMA will employ a spectrally efficient modulation scheme such as 64-QAM and will attain a high bit rate. Stated another way, AMC achieves best tradeoff between transmission throughput and modulation robustness, and by doing so, the system is always working at the maximum possible bit rate, for each user. The algorithms that explore AMC are employed in the physical layer of technologies such as WiMAX.

Generally speaking, contiguous subchannels are more suitable for fixed, portable or with low mobility applications.

4.2.3 OFDMA implementation issues

OFDMA explores several forms of diversity, and that translates to diversity gains. The reader, however, should be aware that the overall gain that OFDMA gets from diversity is lower than the value that one could expect by "summing up" the gains that result from the various implicit diversity techniques. The first explanation for this is the fact that each dimension reduces the gains that can be achieved from the other dimensions. Second, and most importantly, we have to consider the interference between neighboring cells, power and synchronization mismatches and the impossibility of keeping orthogonality in dense urban areas.

4.3 Overview of the physical layer of WiMAX

The physical layer is responsible for ensuring the transport os bits through the wireless channel at high bit rates. WiMAX defines three main physical layers, which differ depending on application type and configuration. Table 4.1 summarizes their main characteristics.

	802.16	802.16-2004	802.16e-2005
Status	Completed Dec/2001	Completed June 2004	Completed Dec/2005
Frequency band	10-66GHz	2-11GHz	2-11GHz for fixed; 2-6GHz for mobile applications
Application	Fixed LOS	Fixed NLOS	Fixed and mobile NLOS
MAC architecture	P2M,mesh	P2M,mesh	P2M, mesh
Transmission scheme	Single carrier only	Single carrier, 256 OFDM or 2,048 OFDM	Single carrier, 256 OFDM or scalable OFDM with 128, 512, 1,024, or 2,048 subcarriers
Modulation	QPSK, 16 QAM,64 QAM	QPSK, 16 QAM, 64 QAM	QPSK, 16 QAM, 64 QAM
Gross data rate	32-134.4Mbps	1-75Mbps	1-75Mbps
Multiplexing	Burst TDM/TDMA	Burst TDM/TDMA/ OFDMA	Burst TDM/TDMA/ OFDMA
Duplexing	TDD and FDD	TDD and FDD	TDD and FDD
Channel bandwidths	20MHz, 25MHz, 28MHz	1.75MHz, 3.5MHz, 7MHz, 14MHz, 1.25MHz, 5MHz, 10MHz, 15MHz, 8.75MHz	1.75MHz, 3.5MHz, 7MHz, 14MHz, 1.25MHz, 5MHz, 10MHz, 15MHz, 8.75MHz
Air-interface designation	WirelessMAN-SC	WirelessMAN-SCa WirelessMAN-OFDM WirelessMAN-OFDMA WirelessHUMAN	WirelessMAN-SCa WirelessMAN-OFDM WirelessMAN-OFDMA WirelessHUMAN
WiMAX implementation	None	256-OFDM as Fixed WiMAX	Scalable OFDMA as Mobile WiMAX

Table 4.1: WiMAX Physical layers[7]

One of the aspects that table 4.1 clarifies is the fact that each layer is associated with a frequency band. If we are operating in 10-66 GHz, a LOS path is necessary because the wavelength is small. However, if we are operating in the 2-11 GHz band, then the wavelength is bigger and communication under NLOS is possible.

We proceed this section with a summarization of the salient parameters of WiMAX. Next, we provide a brief discussion of some of the expected technologies that WiMAX uses to enhance performance such as adaptive modulation and coding, subchannel usage and advanced antenna systems. The reader who wishes to learn more about WiMAX is referred to [7, 58, 58].

4.3.1 OFDM parameters in fixed and mobile WiMAX

The fixed and mobile versions of WiMAX have slightly different implementations of the OFDM physical layer. Fixed WiMAX (IEEE 802.16- 2004) uses a 256 FFT-based OFDM physical layer. Mobile WiMAX (802.16e-2005 standard) uses a scalable OFDMA-based physical layer. In the case of mobile WiMAX, the FFT sizes can vary from 128 to 2048 carriers. Table 4.2 shows the OFDM-related parameters for both the OFDM-PHY and the OFDMA-PHY. The parameters are shown here for only a limited set of profiles that are likely to be deployed and do not constitute an exhaustive set of possible values.

WiMAX OFDM-PHY (Fixed)

A fixed WiMAX signal uses OFDM modulation. For this version the FFT size does not depend on the channel bandwidth and is fixed at 256, which 192 subcarriers used for carrying data, 8 used as pilot subcarriers for channel estimation and synchronization purposes, and the rest used as guard band subcarriers. If we make as analogy with an IP network, a pilot would be the header of a data packet. The pilot carriers use BPSK modulation, whereas data is BPSK, QPSK, 16 QAM, or 64 QAM modulated. The modulation of the data subcarriers typically varies with distance according to fig. 4.9, ie, the better the link (that has to do with the propagation distance), the higher the efficiency of the modulation strategy that we use.

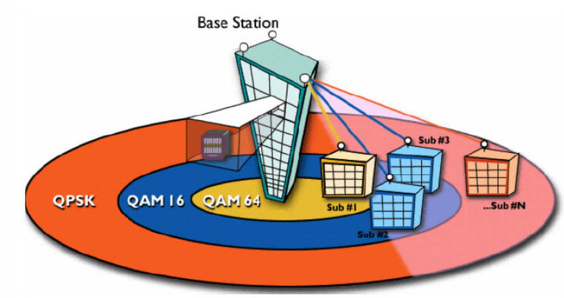


Figure 4.9: Modulation as a function of distance.[57]

Since the FFT size is fixed, the subcarrier spacing varies with channel bandwidth. When larger bandwidths are used, the subcarrier spacing increases, and the symbol time decreases. Decreasing symbol time implies that a larger fraction needs to be allocated as guard time to overcome delay spread. As Tab. 4.2 shows, WiMAX allows a wide range of guard times that allow system designers to make appropriate trade-offs between spectral efficiency and delay spread robustness. For maximum delay spread robustness, a 25 percent guard time can be used, which can accommodate delay spreads up to 16 μs when operating in a 3.5MHz channel and up to 8 μs when operating in a 7MHz channel. In relatively benign multipath channels, the guard time overhead may be reduced.

Although WiMAX can use TDD (Time Division Duplex) or FDD (Frequency Division Duplex), the usual is to employ TDD, since it is more spectrally efficient (it only requires one channel for UL and DL) and simplifies the equipment (which translates to cost advantages). Nevertheless, TDD might cause interferences because the BS and MTs operate at the same frequency. A solution for this issue is to prohibit the BS and the MTs to transmit simultaneously, which is accomplished

Parameter	Fixed WiMAX OFDM-PHY OFDM-PHY		Mobile WiMAX Scalable OFDMA-PHYa		
FFT size	256	128	512	1,024	2,048
Number of used data subcarriers	192	72	360	720	1,440
Number of pilot subcarriers	8	12	60	120	240
Number of null/guardband subcarriers	56	44	92	184	368
Cyclic prefix or guard time (Tg/Tb)	1/32, 1/16, 1/8, 1/4				
Oversampling rate (Fs/BW)	Depends on bandwidth: 7/6 for 256 OFDM, 8/7 for multiples of 1.75MHz, and 28/25 for multiples of 1.25MHz, 1.5MHz, 2MHz, or 2.75MHz.				
Channel bandwidth (MHz)	3.5	1.25	5	10	20
Subcarrier frequency spacing (kHz)	15.625			10.94	
Useful symbol time (μs)	64			91.4	
Guard time assuming 12.5% (μs)	8			11.4	
	15.625			10.94	
OFDM symbol duration (μs)	72			102.9	
Number of OFDM symbols in 5 ms frame	69			48.0	

Table 4.2: OFDM Parameters Used in WiMAX. [7]

by gap insertion (when the BS transmits the MTs are idle and vice-versa).

WiMAX OFDMA-PHY (Mobile)

The first WiMAX specifications use OFDM. Here all subcarriers are allocated to a single user at a time. 802.16e-2005 uses OFDMA, hence allowing users to share subcarriers and time-slots. This, of course, also have downsides: the transmitters needs to get information about all users and the receiver needs to know which subcarriers correspond to each user. OFDMA can accommodate various users with different specifications: speed,QoS, etc. One of the main advantages of preferring OFDMA lays in its flexibility with respect to transmission power reduction and the minimization of the PAPR problem: by dividing the available bandwidth among the various MTs within the cell, each MT only gets a portions of the overall subcarriers, hence it transmits with a smaller PAPR.

In Mobile WiMAX, the FFT size is scalable from 128 to 2,048, thereby justifying the designation of **S**-OFDMA to the OFDMA implementation of the mobile WiMAX. Here, when the available bandwidth increases, the FFT size is also increased such that the subcarrier spacing is always 10.94kHz. This keeps the OFDM symbol duration, which is the basic resource unit, fixed and therefore makes scaling have minimal impact on higher layers. A scalable design also keeps the costs low. The subcarrier spacing of 10.94kHz was chosen as a good balance between satisfying the delay spread and Doppler spread requirements for operating in mixed fixed and mobile environments. This subcarrier spacing can support delay-spread values up to até 20μs and vehicular mobility up to 125 km/h when operating in 3.5GHz. A subcarrier spacing of 10.94kHz implies that 128, 512, 1,024, and 2,048 FFT are used when the channel bandwidth is 1.25MHz, 5MHz, 10MHz, and 20MHz, respectively. As it was the case for the fixed WiMAX, the mobile WiMAX also allows the modulation type to be QPSK, 16 QAM or 64 QAM.This is specially advantageous for the upper layers, because they do not perceive any impact when the channel bandwidth and subcarrier spacing change. Consequently, the implementation cost is reduced.

4.3.2 Frame and slot structure

PHY layer is also responsible for the slot allocation and the transmission of frames. Each slot consists of a subchannel that spans 2,3 or 4 OFDM symbols, depending on the scheme that is adopted to attribute the chosen subchannels. A contiguous series of slots that is given to a user is called *user data region*; the scheduling of the data region is based on criteria such as QoS and channel state.

4.3.3 Modulation and adaptive coding in WiMAX

In WiMAX, the modulation is adaptive, meaning that ,within a region that is covered by a given cell, we can have several kinds of coding and modulation as a function of SNR (Eb/N_0). When the transmission conditions are optimum (ie, LOS transmission and small propagation distance) we can have a modulation type that allows high bit rates, such as 64-QAM. If the propagation conditions are worse (ie, low SINR and/or big propagation distance) then a modulation will be required so as to sacrifice transmission rate for the sake of connection stability (with low error rate).

the literature refers to this concept as adaptive modulation and coding (AMC). Figure 4.10 shows some of the 52 modulation and coding schemes that WiMAX supports.

In the downlink, the usage of QPSK, 16 QAM or 64 QAM is required, both for mobile and fixed WiMAX; 64 QAM is optional in the uplink. FEC (forward error correction) coding, which is achieved with convolucional codes, is compulsory. *Turbo* or *LDPC (low-density parity check)* codes are also optional and it is expected that they will play an important role in the implementations of the most competitive companies.

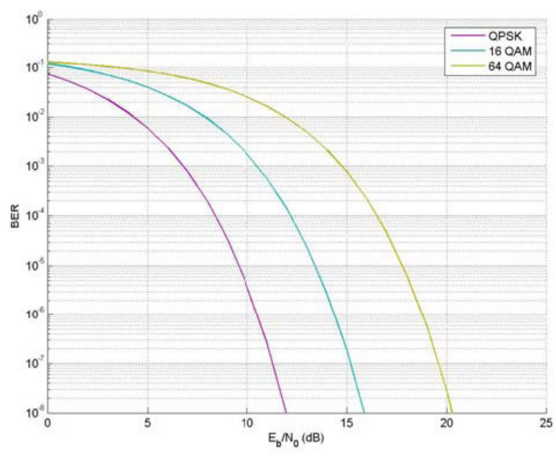


Figure 4.10: SNR vs BER (calculated using matlab (`bertool`)).

Spectral efficiency is defined as the amount of information that can be transmitted on a given bandwidth. It is measured in bits/s/Hz.

Figure 4.11 clearly shows what was mentioned before: modulation such as 64-QAM are only spectrally efficient for high SNR conditions, and despite having a high transmission rate, for low

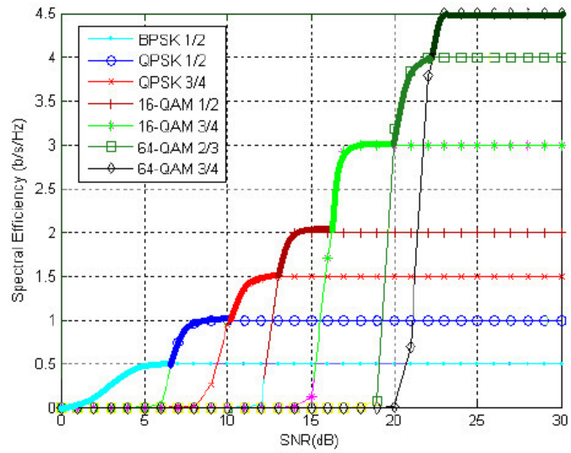


Figure 4.11: Spectral Efficiency vs SNR. AMC "moves" in the bold curve.

SNR values the spectral efficiency is reduced. The less efficient modulation strategies are relatively better for the lower SNR.

There is also a predefined IEEE table with the specifications of the minimum SNR for each modulation. As table 4.3 exemplifies, for a SNR of 10.5dB, the recommended modulation to use is QPSK with a coding rate equal to 1/2.

Modulation	Code Rate	Minimum SNR (dB)
BPSK	1/2	6.4
QPSK	1/2	9.4
	3/4	11.2
16–QAM	1/2	16.4
	3/4	18.2
64-QAM	2/3	22.7
	3/4	24.4

Table 4.3: Minimum SNR values for each modulation type.[58]

4.3.4 Advanced Features for Performance Enhancements

WiMAX defines a number of optional advanced features for improving the performance. Among the more important of these advanced features are support for multiple-antenna techniques, hybrid-ARQ, and enhanced frequency reuse.

Advanced Antenna Systems (AAS)

The WiMAX standard provides extensive support for implementing advanced multiantenna solutions to improve system performance. Significant gains in overall system capacity and spectral efficiency can be achieved by deploying the optional *advanced antenna systems* (AAS) defined in WiMAX. AAS includes support for a variety of multiantenna solutions, including transmit diversity, beamforming, and spatial multiplexing.

Hybrid-ARQ

ARQ (Automatic Repeat Request) systems are those in which the receiver checks the validity of the received data. Upon failure, it asks for a retransmission. The verification is performed with CRC and can be improved if, in addition to CRC,we protect the feedback data with a error correction code. This strategy is referred to as Hybrid-ARQ.

Hybrid-ARQ is, therefore, an ARQ system that is implemented at the physical layer together with FEC, providing improved link performance over traditional ARQ at the cost of increased implementation complexity. The simplest version of H-ARQ is a simple combination of FEC and ARQ, where blocks of data, along with a CRC code, are encoded using an FEC coder before transmission; retransmission is requested if the decoder is unable to correctly decode the received block. When a retransmitted coded block is received, it is combined with the previously detected coded block and fed to the input of the FEC decoder. Combining the two received versions of the code block improves the chances of correctly decoding. This type of H-ARQ is often called type I chase combining.

To further improve the reliability of retransmission, WiMAX also optionally supports type II H-ARQ, which is also called incremental redundancy. Here, unlike in type I H-ARQ, each (re)transmission is coded differently to gain improved performance. Typically, the code rate is effectively decreased every retransmission. That is, additional parity bits are sent every iteration, equivalent to coding across retransmissions.

Improved Frequency Reuse

Although it is possible to operate WiMAX systems with a universal frequency reuse plan, doing so can cause severe outage owing to interference, particularly along the intercell and intersector edges. To mitigate this, WiMAX allows for coordination of subchannel allocation to users at the cell edges such that there is minimal overlap. This allows for a more dynamic frequency allocation across sectors, based on loading and interference conditions, as opposed to traditional fixed frequency planning. Those users under good SINR conditions will have access to the full channel bandwidth and operate under a frequency reuse of 1. Those in poor SINR conditions will be allocated nonoverlapping subchannels such that they operate under a frequency reuse of 2, 3, or 4, depending on the number of nonoverlapping subchannel groups that are allocated to be shared among these users. This type of subchannel allocation leads to the effective reuse factor taking fractional values greater than 1. The variety of subchannelization schemes supported by WiMAX makes it possible to do this in a very flexible manner. Obviously, the downside is that cell edge users cannot have access to the full bandwidth of the channel, and hence their peak rates will be reduced.

Chapter 5

RELAY-ASSISTED COOPERATIVE SCHEMES

5.1 Introduction

Wireless systems are one of the key components for enabling the information society. Thus, it is expected that the demand for wireless services will continue to increase in the near and medium term, calling for more capacity and created the need for cost effective transmission techniques that can exploit scarce spectral resources efficiently. It is anticipated that the broadband mobile component of beyond 3G systems must be able to offer bit rates in excess of 100Mbps in indoor and picocell environments. To achieve such high bit rates, so as to meet the quality of service requirements of future multimedia applications, orthogonal frequency division multiple access (OFDMA) has been adopted in different flavors of broadband wireless systems [32, 23]. The physical layer of WiMAX and LTE for downlink, for example, rely on OFDMA.

It is commonly agreed that the provision of the broadband wireless component will probably rely on the use of multiple antennas at transmitter/receiver side. Multiple input and multiple output (MIMO) is a very promising technique to mitigate the channel fading and thus improve the cellular system capacity. By configuring multiple antennas at both the base station (BS) and mobile terminal (MT), the channel capacity may be improved and be proportional to the minimum number of the antennas at the transmitter or receiver sides [15]. However, using an antenna array at the MTs may not be feasible due to size, cost and hardware limitations. Moreover, if the MTs are equipped with multiple antennas, the spatial separation between antennas must be large enough to guarantee the statistical independence of the faded signals for optimal performance [18]. These devices are usually small and light and thus this spatial separation requirement is difficult to satisfy.

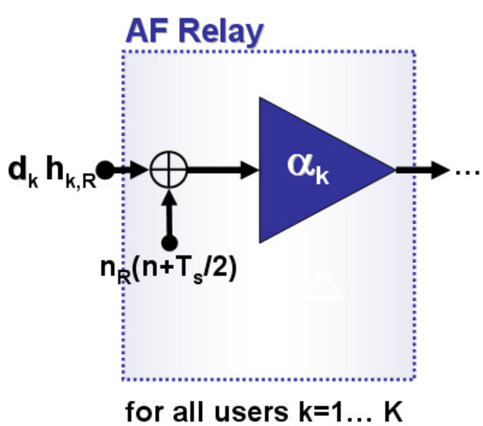
Cooperative communications is a promising solution for wireless systems to overcome the above limitations [16]. It allows single antenna devices to gain some benefits of spatial diversity without the need for physical antenna arrays. Research on cooperative diversity can be traced back to the pioneering papers of Van der Meulen [38] and Cover, El Gamal [12] on the information theoretic properties of the relay channel. Explicit cooperation of neighboring nodes was considered in [49, 13, 31]. In such cooperative transmission scenarios, two or more sources (genuine sources or relays) transmit the same information to a destination, generating a virtual antenna array. More details are provided in section 5.2.2 and [11].

In [13, 31], the use of orthogonal space-time block coding (STBC) in a distributed fashion for practical implementation of user cooperation has been proposed. Several authors have also addressed the search and design of practical distributed space-time codes for cooperative communications [51]. A cooperative scheme for the UL OFDMA has been proposed in [22]. In this scheme each user transmits his partner's and his own data on different subcarriers.

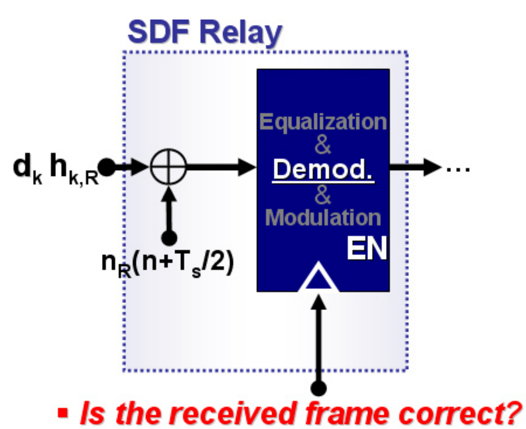
In this dissertation we evaluate the performance of virtual MIMO or relay-assisted cooperative schemes designed for the UL OFDMA based systems. In the scheme proposed in this dissertation there is no explicit cooperation between the different users. The proposed cooperative schemes emulate a MIMO channel with 2 transmit and M receive antennas. Two relaying protocols are considered: amplify-and-forward (AF) and selective decode-and-forward (SDF).

In AF protocols, which are schematized in fig. 5.1**(a)**, the relays only scale the signals received from the source (or from other relays) and forward them to the destination (or to other relays) without other processing. These protocols are easy to implement in practical communication systems, as the computational complexity they introduce at the relay is limited to the scaling and delay operations. The amplify-and-forward protocol was first introduced in [31] for the single-relay, and the orthogonality is ensured by sending data in different time intervals. Another potential challenge is that sampling, amplifying, and retransmitting analog values is technologically nontrivial. Nevertheless, amplify-and-forward is a simple method that lends itself to analysis, and thus has been very useful in furthering the understanding of cooperative communication systems.

DF protocols are the protocols in which the relays operate on the signal they receive from the source (or from other relays) before forwarding it. The processing involves equalization, demodulation and modulation. They are considered selective DF (SDF), as shown in figure 5.1**(b)**, if they use the frame information to check whether the received bit is correct or not and use that knowledge to inhibit transmission if the information is wrong. Figure 5.1 also make it apparent that the relays insert noise into the received signal.



(a) AF protocol.



(b) SDF protocol.

Figure 5.1: Relaying schemes comparison.

We derive the instantaneous normalized capacities and both the outage capacities and bit error rate (BER) are compared against the non-cooperative SISO and SIMO systems, considering different scenarios. We show by simulations that the proposed cooperative schemes increase the system capacity and coverage, mainly in dense urban environment where a high path loss (PL) does exist.

The remaining part of this chapter is organized as follows: in section 5.2, we present and contrast two world wide projects that focus on cooperative diversity. In section 5.3, we start by presenting a general system model for the proposed OFDMA based relay-assisted systems and we derive the instantaneous capacities of the relay-assisted schemes. Then, in 5.4, we assess the performance, in terms of outage capacities and BER, in different scenarios. We investigate their dependence on E_b/N_0 and the path loss. The most important results are summarized in the next chapter and also in [39].

5.2 World Wide Initiatives

At the moment, there are groups and projects whose purpose it to study the standardization of cooperative diversity protocols.

The purpose of this section is exactly to introduce two world wide initiatives that are taking place in order to turn cooperative diversity results into practical and commercial standards; one of them is the IEEE 802.16j(section 5.2.1) and the other one is CODIV (Enhanced Wireless Communication Systems Employing **CO**operative **DIV**ersity), in section 5.2.2. As we shall find out throughout this section, the main difference between these initiatives lies in the fact that CODIV assumes that relays can be mobile, whereas 802.16j assumes they are fixed.

We point out that the scope of this dissertation is related to CODIV, that is managed by the telecommunications institute [55].

5.2.1 IEEE 802.16j Standard Main Characteristics

In the standardization arena, the most advanced project contemplating the use of relaying and some forms of cooperation is the IEEE 802.16j. The IEEE 802.16 working group has created on March 30, 2006 a "Relay Task Group" to develop a draft for the Mobile Multi-hop Relay (MMR) system (802.16j).

As listed in the documents issued within this task group [25], the main motivations for multi-hop relaying were the following:

- Coverage/Range Extension.
- Improved throughput.
- In-building penetration.
- Infrastructure offloading.
- Improved frequency reuse.

Figure 5.2 illustrates the architecture and usage scenarios that are envisioned for IEEE 802.16j.

In terms of main requirements, the objectives of the standard to be developed are that it has to provide backwards compatibility with 802.16e, and in this sense the architecture is not a mesh network but rather a tree as shown in Figure 5.3. The relay stations are not allowed to forward the traffic between two UT nodes and all the traffic must go through the MMR-BS as in a conventional cellular architecture. An ad-hoc capable UT can be a source, destination, or a relay, although dedicated relay stations are considered in most of the documents issued up to now.

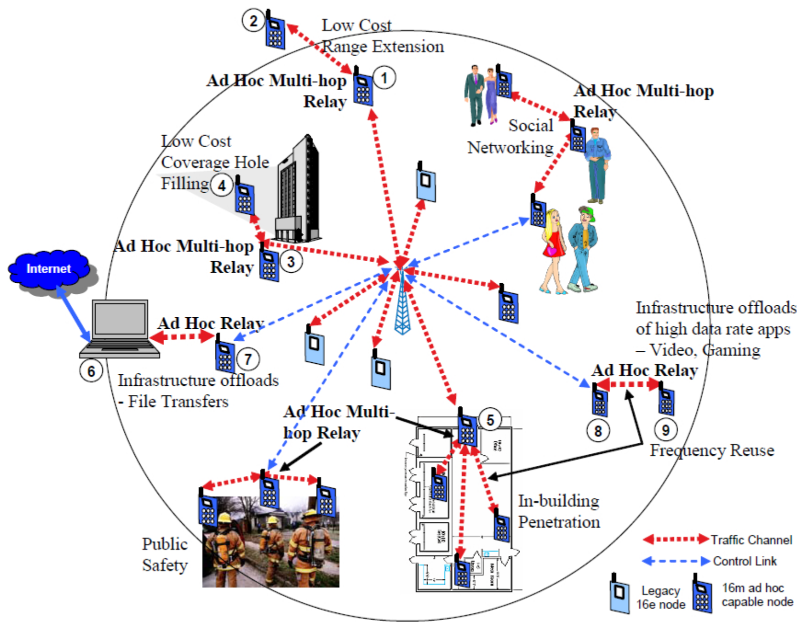


Figure 5.2: Architecture and usage scenarios for IEEE 802.16j.

The main specifications for IEEE 802.16j arise from the required backwards compatibility with 802.16e. The chosen access technique is OFDMA, the duplexing method is TDD (but FDD is optional) and the bandwidth, coding, beam-forming options are the same as for 802.16e.

5.2.2 CODIV cooperation scenarios

The main goal of the CODIV is to investigate and develop technologies that would enhance the performance of wireless communication systems, employing the inherent diversity of radio channels (channel diversity), as well as the cooperation between users (multiuser diversity). In fact, one of the key points of the project is to exploit the cooperative reserves and capabilities of the user-equipments, although incentive policies might be needed to stimulate this cooperation. The expected improvement in the performance of radio systems pursued by CODIV technologies will come in terms of higher bit rates and spectrum efficiency, and decreased (enhanced) power consumption, as well as coverage extension and fairness. Therefore, a first step in the project should be to identify and define the scenarios where these technologies can be validated and evaluated, allowing the estimation of the benefits for the future wireless communications, on one hand, and enabling comparisons with other proposed solutions, on the other hand.

CODIV analyzes the scenarios that are more likely to match the operation of the future wireless communication systems, focusing on the cellular and metropolitan cases, where the technology developed in CODIV are expected to have a relevant influence. So, a preliminary selection of scenarios, in which more benefit can be obtained with the application of CODIV technology, is proposed and such scenarios are defined through the identification of the main top-level technical requirements. These scenarios will constitute the ones used for the evaluation of the different diversity mechanisms developed in the WP3 and WP4 (the workgroups that will study the physical and network layer algorithms), as well as for the assessment of the diversity methods integration

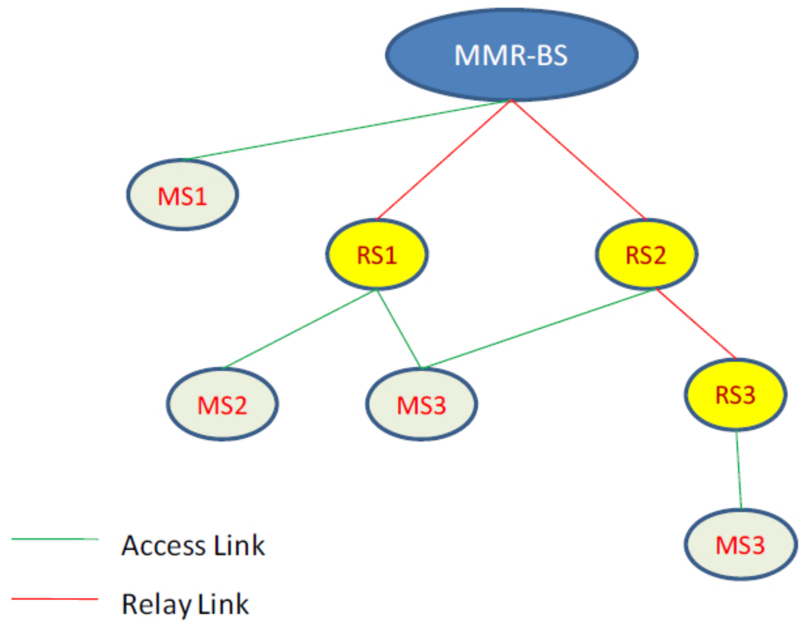


Figure 5.3: Architecture for IEEE 802.16j.

that will be carried out at system level in WP5.

Preliminary Scenarios Definition

According to the CODIV project's scope of work, *we are looking to enhance and elaborate the performance of wireless communication systems with respect to its key performance characteristics such as coverage, fairness, and QoS. In order to do so we need to define scenarios for analysis, simulation and actual demonstration*[11].

When reviewing the work done so far in the field of cooperative diversity in wireless communication it appears that although there are many possibilities for exploiting the principle of cooperative diversity, the vast majority of the concepts and techniques are based on a basic transmitter - relay - receiver scenario as depicted in Fig. 5.4.

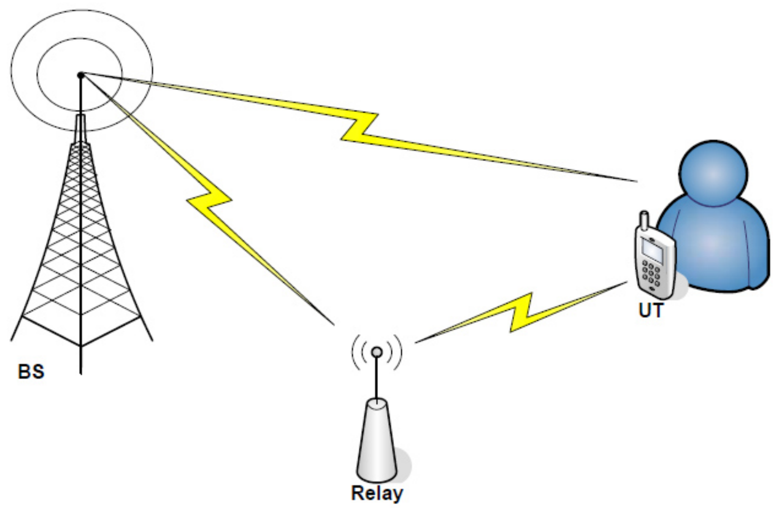


Figure 5.4: The basic scenario of cooperative diversity.

This was the scenario that was explored in the scope of this dissertation. More details follow in sections 5.1 to 5.4.

The greatest advantage of this basic scenario is that by slightly changing it, we can demonstrate a broad and diverse set of techniques for cooperative diversity. The different possibilities for using this basic scenario will be described in section 5.2.2.

One other concept for demonstration is to elaborate existing MIMO techniques in a way which will provide us with better tools for coping with the CODIV challenges. This scenario consists of a BS with a certain number of antennas, and the same number of UTs with single antenna. In this scenario we will use something similar to what is known in the technical literature as the combination of joint detection in the upstream and pre-equalization in the downstream, and it will be a test bed for examining new MIMO techniques as a complementary scenario to the relay based scenario.

Although there is a conceptual difference between both scenarios, there are general requirements with which all scenarios needs to comply. As examples, we can mention the facts that the BS is assumed to be fixed whereas the relay and MTs can be mobile or fixed. Also, with respect to the issue of environmental characteristics, the relay-based scenario will focus on outdoor scenario whereas the multi-antenna scenario will focus, at least in a first phase, on an indoor small office environment.

Relay based scenarios

The basic relay based scenario represents, actually, a group of scenarios which can be exploited in many ways to demonstrate either by simulation or by physical demonstration a large number of techniques and algorithms to be used in a cooperative diversity wireless network the different possible sub-scenarios are here forth described.

- **Diversity enhancement**

Here we can provide a scenario in which the relays are situated significantly apart, thus providing links with minimal correlation which leads to higher diversity.

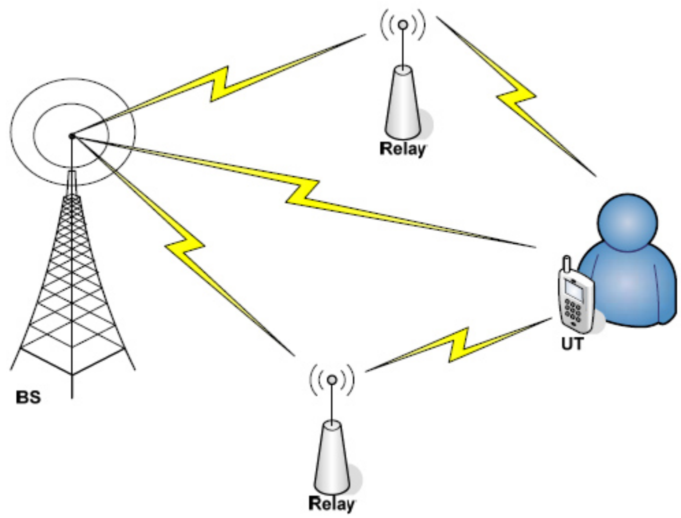


Figure 5.5: Diversity enhancement scenario.

- **Coverage enlargement**

Here we can provide a scenario in which a single UT or a group of UTs is far beyond the BS's realistic transmission range, and a coverage enlargement is reached by using the relays.

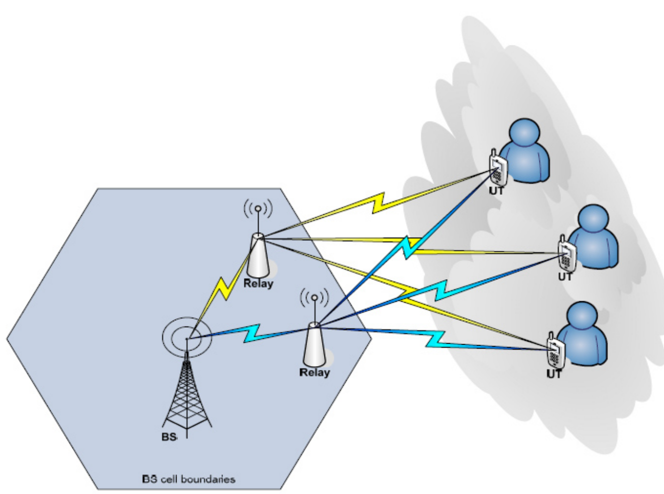


Figure 5.6: Coverage enlargement scenario.

- **Fairness enhancement**

Here we can provide a scenario in which the fairness enhancement is being examined using the relays as obstacle bypass.

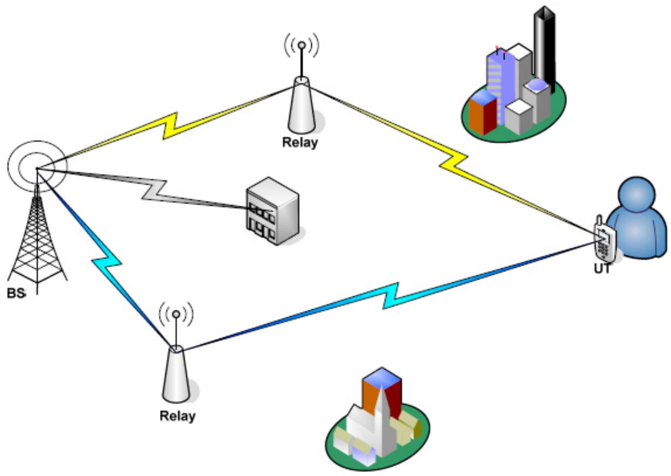


Figure 5.7: Fairness enhancement scenario.

- **Additional possible scenarios**

Using the relay based concept we can form additional scenarios such as a scenario in which the relays are linked between them, forming a sub-net or a cluster. These scenarios are intended to be examined by simulation as the physical demonstration equipment does not necessarily support it.

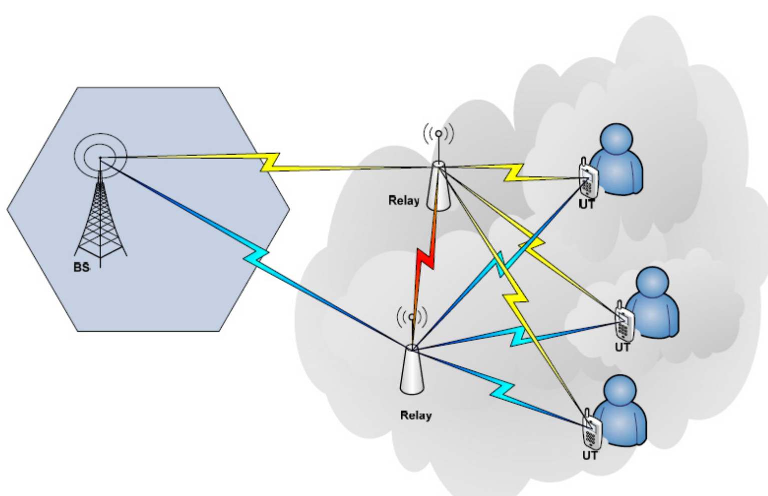


Figure 5.8: Suggested additional scenario.

5.2.3 CODIV and 802.16j main differences and similarities

In essence, the main difference between the CODIV project and the scenarios outlined for 802.16j, is that in the CODIV vision there is *no specific design for dedicated relay stations and the terminals may act as relays.* Therefore, CODIV always require have mobility capabilities. However any user terminal(UT) with cooperation and relaying capabilities can be used as a fixed or nomadic relay and used either by the operator to improve the capabilities of its network or by a customer. By other side, one can imagine that an operator can simply use a UT and make it operate solely as a relay station. Therefore the performance objectives and benefits to the user pointed out in 802.16j still hold for CODIV.

Table 5.1 shows the performance objectives, the benefits to the user and possible use cases associated with the different scenarios, as listed in IEEE 802.16j harmonization document concerning the scenarios. The same holds for CODIV since, as we referred previously, if the terminals have relaying capabilities, any of the scenarios can be fulfilled using a UT as a dedicated relay station, although we could point out that, eventually, when used as fixed or nomadic relay stations, the additional complexity of the mobile relays would be unneeded.

Scenario	Performance Objectives	Benefits to the user	Use cases
Fixed infrastructure with relay stations	Improved coverage, Higher capacity, Extended Range	Ubiquitous access,Higher capacity	Coverage holes (Shadowing from buildings, valleys, tunnels), Cell Edge
In-building	Improved coverage, higher capacity	Ubiquitous access, Higher capacity	Inside building Inside tunnels Under ground
Temporary Coverage	Improved coverage, higher capacity, Extended Range	Ubiquitous access Higher capacity	Emergency disaster, Coverage of temporary event
Coverage On Mobile Vehicle	Improved coverage, Higher capacity,	Ubiquitous access, Higher capacity	Inside buses and taxis
	Extended Range	Mobility	Inside Ferries, Inside Trains

Table 5.1: Performance objectives, benefits and use cases associated with the different scenarios.

5.3 Proposed Relay-Assisted Cooperative Schemes

Within this section, emphasis will be given to the capacity analysis of both non-cooperative systems and two half-duplex virtual MIMO schemes: *amplify-and-forward* and *selective decode-and-forward* based relays. These results can be also found in the article that was proposed for the 4^{th} international wireless conference on mobile communications [39]. For that, we start by presenting a system model for an uplink OFDMA system (section 5.3.1) and a Non-Cooperative MISO system capacity analysis.

5.3.1 System Model

Figure 5.9 depicts the proposed uplink OFDMA system, which consists of K MTs, a relay and a BS. In OFDMA, we can pre allocate or dynamically assign the subcarriers to different users. We assume, for simplicity, a static allocation scheme in which the total available subcarriers, N_c, are equally distributed among the K users so that each of them occupies N_c/K subcarriers. All the MTs and the relay are equipped with single antenna whereas the BS uses an antenna array.

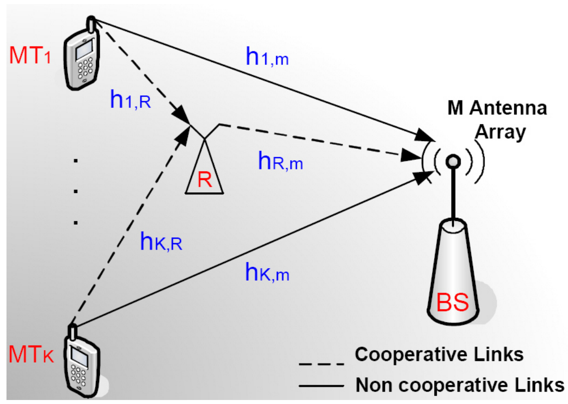


Figure 5.9: Virtual MIMO Scheme for OFDMA based systems.

Here the signal that comes from each mobile user k arrives at the BS by a non-cooperative direct link and a non-cooperative link, which is created by the relay.

The cooperative cycle requires 2 phases (Fig. 5.10). In the first phase all users broadcast their own information, $d_1...d_k$, to the half duplex relay and BS. The relay does not transmit data during this time. In the second one the MTs do not transmit data, and the relay transmits the received information to the BS, which can now decode the information.

Next, we analyze both non-cooperative systems and two half-duplex virtual MIMO schemes: *amplify-and-forward* and *selective decode-and-forward* based relays.

5.3.2 Non-Cooperative MISO system

The classical system is included to act as a reference for comparison of the advantages and disadvantages offered by the relay-assisted schemes. Without loss of generality, we focus our analysis on a generic user k. We also assume that the number of subcarriers is equal to the number of active users, i.e. $N_c = K$.

Thus, the received signal at the BS, at instant $n + T_s$ and antenna m is given by

$$y_{BS,k,m}(n + T_s) = d_k h_{k,m} + n_m(n + T_s) \tag{5.1}$$

where d_k is the data symbol of the k^{th} user, $h_{k,m}$ represents the complex flat Rayleigh fading non-cooperative channel of the k^{th} user and of the m^{th} antenna, and $n_m(n + T_s)$ are the zero mean complex additive white Gaussian noise (AWGN) samples on antenna m at instant $n + T_s$. In this classical systems the signal received on each antenna are combined by using maximum ratio combining (MRC). Thus, the soft decision variable of the k^{th} user may be expressed as

$$\hat{d}_k = \underbrace{d_k \sum_{m=1}^{M} |h_{k,m}|^2}_{Desired\ Signal} + \underbrace{\sum_{m=1}^{M} h^*_{k,m} n_m(n + T_s)}_{BS\ Noise} \tag{5.2}$$

The normalized capacity for the classical (non-cooperative) case can be written as

$$C_{k,non-coop.} = \log_2\left(1 + \sum_{m=1}^{M} \left|h_{k,m}\right|^2 SNR\right) \tag{5.3}$$

5.3.3 Amplify and Forward

In this scheme (Fig. 5.10), during the first phase the MTs transmit at full power for $T_s/2$ of the time. During the second phase the relay must also transmit at full power for $T_s/2$ to make up the full time slot.

Thus, the received signal at the BS, at instant $n + Ts/2$ and antenna m is given by

$$y_{BS,k,m}(n + T_s/2) = d_k h_{k,m} + n_m(n + T_s/2) \tag{5.4}$$

The received signal at the BS and at instant $n + T_s$ is given by

$$y_{BS,k,m}(n + T_s) = \alpha_k \left(d_k h_{k,R} + n_R(n + T_s/2)\right) h_{R,m} + n_m(n + T_s) \tag{5.5}$$

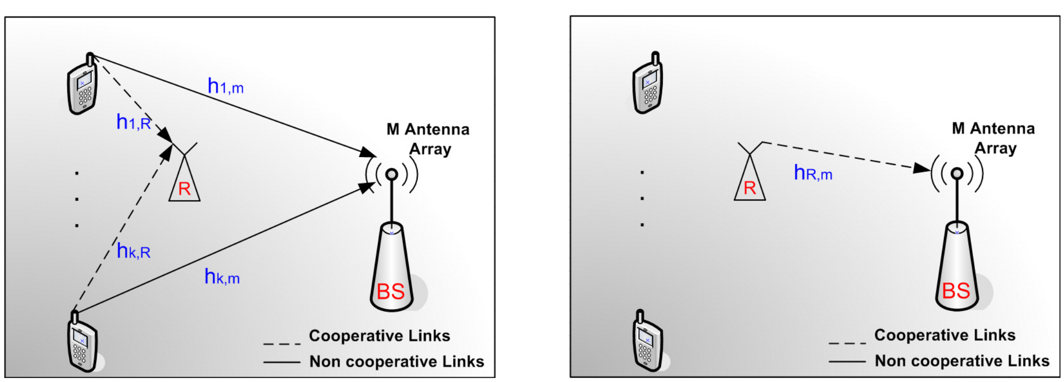
where $n_R(n + T_s/2)$ are the complex AWGN samples on relay, with zero mean and variance σ^2. $h_{R,m}$ and $h_{k,R}$ represent the complex flat Rayleigh fading cooperative channels from the relay to the BS and from user k to the relay, respectively. The constant α_k is used to constrain the transmit relay power to one and is given by

$$\alpha_k = \frac{1}{\sqrt{\left|h_{k,R}\right|^2 + \sigma^2}} \tag{5.6}$$

After some mathematical manipulations and considering that $\overline{h}_{k,R,m} = \alpha_k h_{k,R} h_{R,m}$, we can write

$$y_{BS,k,m}(n + T_s) = d_k \overline{h}_{k,R,m} + \alpha_k h_{R,m} n_R(n + T_s/2) + n_m(n + T_s) \tag{5.7}$$

At the BS, the received signals at instants $n + T_s/2$ and $n + T_s$ are combined by the MRC criterion. The received signals are multiplied by coefficients $g_{k,m}(n')$, which are given by Eq. 5.8.



(a) First phase: $0 \rightarrow T_s/2$

(b) Second phase: $T_s/2 \rightarrow T_s$

Figure 5.10: Cooperation schemes assume a single hop relay system.

$$g_{k,m}(n') = \begin{cases} h^*_{k,m}/\sigma^2, & n' = n + T_s/2; \\ \overline{h}^*_{k,R,m}/\sigma^2_{k,m}, & n' = n + T_s. \end{cases} \tag{5.8}$$

Here $(\cdot)^*$ denotes complex conjugate operator and $\sigma^2_{k,m}$ represents the total power noise received on antenna m at instant $n + T_s$. It may be related to σ^2 as follows:

$$\sigma^2_{k,m} = \sigma^2 \left(\alpha_k^2 |h_{R,m}|^2 + 1\right) = \sigma^2 \beta_{k,m} \tag{5.9}$$

Thus the soft decision variable of the k^{th} user may be expressed as

$$\begin{aligned} \hat{d_k} = & \underbrace{\frac{d_k}{\sigma^2} \sum_{m=1}^{M} \left(|h_{k,m}|^2 + \left|\bar{h}_{k,R,m}\right|^2 / \beta_{k,m}\right) +}_{Desired\ Signal} \\ & \underbrace{+ \frac{\alpha_k}{\sigma^2} \sum_{m=1}^{M} \left(h_{R,m} \bar{h}^*_{k,R,m} / \beta_{k,m}\right) n_R(n + T_s/2)}_{Relay\ Noise} \\ & \underbrace{+ \frac{1}{\sigma^2} \sum_{m=1}^{M} \left(h^*_{k,m} n_m(n + T_s/2) + \bar{h}^*_{k,R,m} n_m(n + T_s)\right)}_{BS\ Noise} \end{aligned} \tag{5.10}$$

From Eq. 5.10, it is easy to see that the normalized capacity is given by

$$C_{k,AF} = \frac{1}{2} \log_2 \left(1 + \eta_{AF_k} SNR_k\right) \tag{5.11}$$

where the factor η_{AF_k} is given by Eq. 5.12. The factor $1/2$ is used since the bandwidth is increased by a factor of 2.

$$\eta_{AF_k} = \frac{\left|\sum_{m=1}^{M} \left(|h_{k,m}|^2 + \frac{|\bar{h}_{k,R,m}|^2}{\beta_{k,m}}\right)\right|^2}{\alpha_k^2 \left|\sum_{m=1}^{M} h_{R,m} \frac{\bar{h}^*_{k,R,m}}{\beta_{k,m}}\right|^2 + \sum_{m=1}^{M} \left|h^*_{k,m}\right|^2 + \sum_{m=1}^{M} \left|\frac{\bar{h}^*_{k,R,m}}{\beta_{k,m}}\right|^2} \tag{5.12}$$

The signal noise ratio (SNR) can be represented as

$$SNR_k = \frac{P_{t,k}}{\sigma^2} \tag{5.13}$$

where $P_{t,k}$ represents the overall transmitted power of the k^{th} user and σ^2 is the variance of the additive noise at both the BS and relay. The overall transmit power is constrained to one for all K active users, i.e., $P_{t,k} = 1,\ k = 1, \ldots, K$.

5.3.4 Suboptimum Amplify and Forward

Section 5.3.3 presented an amplify and forward scheme in which the BS needs to estimate a relatively large amount of variables, such as: $h_{k,m}$, σ^2, $\overline{h}_{k,R,m}$ and $\sigma^2_{k,m}$ in order to perform MRC. However, in practise, this might not be feasible due to difficulties at estimating $\sigma^2_{k,m}$. For this reason, there are situations in which a *suboptimum* but *simpler* scheme which *only* requires estimations for $h_{k,m}$ and $\overline{h}_{k,R,m}$ might be a better solution.

The proposed *Suboptimum Amplify and Forward* scheme only differs from the *Optimum Amplify and Forward* one at the processing that takes place at the BS. Whereas the optimum AF algorithm requires the BS to perform MRC, the suboptimum AF requires a *suboptimum MRC* combining that only requires the knowledge of $h_{k,m}$ and $\overline{h}_{k,R,m} = \alpha_k h_{k,R} h_{R,m}$.

For this scheme, we can express the received signals at instants $n + T_s/2$ and $n + T_s$ as:

$$y_{BS,k,m}(n+T_s) = d_k \overline{h}_{k,R,m} + \alpha_k h_{R,m} n_R(n+T_s/2) + n_m(n+T_s) \tag{5.14}$$

In this case, the received signals are multiplied by coefficients $g_{k,m}(n')$, which are given by Eq. 5.15.

$$g_{k,m}(n') = \begin{cases} h^*_{k,m}, & n' = n + T_s/2; \\ \overline{h}^*_{k,R,m}, & n' = n + T_s. \end{cases} \tag{5.15}$$

Note that 5.15 differs from 5.8 in the fact that the total power noise received on antenna m at instant $n+T_s/2$,σ^2, and the total power noise received on antenna m at instant $n+T_s$, $\sigma^2_{k,m}$, are no longer needed.

The soft decision variable of the k^{th} user may be expressed as

$$\begin{aligned} \hat{d_k} = & \underbrace{d_k \sum_{m=1}^{M} \left(|h_{k,m}|^2 + \left|\bar{h}_{k,R,m}\right|^2 \right) +}_{Desired\ Signal} \\ & \underbrace{+ \alpha_k \sum_{m=1}^{M} \left(h_{R,m} \bar{h}^*_{k,R,m} \right) n_R(n+T_s/2)}_{Relay\ Noise} \\ & \underbrace{+ \sum_{m=1}^{M} \left(h^*_{k,m} n_m(n+T_s/2) + \bar{h}^*_{k,R,m} n_m(n+T_s) \right)}_{BS\ Noise} \end{aligned} \tag{5.16}$$

From Eq. 5.16, the normalized capacity is now given by

$$C_{k,sub-opt-AF} = \frac{1}{2} \log_2 \left(1 + \eta_{AF_k} SNR_k\right) \tag{5.17}$$

where the factor η_{AF_k} is given by 5.18.

$$\eta_{sub-opt-AF_k} = \frac{\left| \sum_{m=1}^{M} \left(|h_m|^2 + \left|\bar{h}_{k,R,m}\right|^2 \right) \right|^2}{\alpha_k^2 \left| \sum_{m=1}^{M} h_{R,m} \bar{h}^*_{k,R,m} \right|^2 + \sum_{m=1}^{M} \left| h^*_{k,m} \right|^2 + \sum_{m=1}^{M} \left| \bar{h}^*_{k,R,m} \right|^2} \tag{5.18}$$

As usual, $P_{t,k}$ represents the overall transmitted power of the k^{th} user and σ^2 is the variance of the additive noise at both the BS and relay. The overall transmit power is again constrained to one for all K active users, i.e., $P_{t,k} = 1,\ k = 1, \ldots, K$.

5.3.5 Selective Decode and Forward

It has been shown that fixed DF transmission does not offer diversity gains for large SNR, because requiring the relay to fully decode the information transmitted by MTs limits the performance of this scheme to that of non-cooperative systems [31]. A selective DF scheme can be used to overcome the fixed DF shortcomings.

In this scheme, during the first phase the MTs transmit at full power for $T_s/2$ of the time, and the received signal at the BS, at instant $n+Ts/2$ and antenna m is also given by Eq. 5.4. During the second phase, the relay first demodulates and decodes the received signals. Upon success, it re-encodes the data and forwards to the BS. Thus, the received signal on antenna m at instant $n+T_s$ can be written by

$$y_{BS,k,m}(n+T_s) = d_k h_{k,m} + n_m(n+T_s) \tag{5.19}$$

At the BS, the received signals given by Eq. 5.4 and Eq. 5.19 are combined using the MRC criterion and the k^{th} user's resulting soft decision variable is

$$\begin{aligned} \hat{d_k} = & \underbrace{d_k \sum_{m=1}^{M} \left(|h_{k,m}|^2 + |h_{R,m}|^2 \right) +}_{Desired\ Signal} \\ & + \underbrace{\sum_{m=1}^{M} \left(h_{k,m}^* n_m(n+T_s/2) + \bar{h}_{R,m}^* n_m(n+T_s) \right)}_{BS\ Noise} \end{aligned} \tag{5.20}$$

As can be seen from 5.20, , in case of full decoding of the information transmitted by MTs, this cooperative scheme yields a diversity gain of $2.M$ under perfect conditions with respect to a non-diversity scheme and a gain of 2 with respect to a non-cooperative SIMO system, i.e., a cellular system with a single antenna at MT and a BS equipped with M antennas, where the signals of each antenna are combined using the MRC.

In the outage case where the relay fails to decode the data correctly, it cannot help the MTs for the current cooperation round. In this case the BS only uses the signal received directly from the MTs. Therefore, the soft decision variable is given by Eq. 5.2.

The normalized capacity of this scheme can be given by

$$C_{k,SDF} = \begin{cases} \frac{1}{2}\log_2\left(1+\sum_{m=1}^{M} |h_{k,m}|^2 SNR_k\right) & \text{if} \frac{|h_{k,R}|^2}{N_R} < \frac{\sum_{m=1}^{M}|h_{k,m}|^2}{N_{BS}} \\ \frac{1}{2}\log_2\left(1+\sum_{m=1}^{M} \left(|h_{k,m}|^2 + |h_{R,m}|^2\right) SNR_k\right) & \text{if} \frac{|h_{k,R}|^2}{N_R} \geq \frac{\sum_{m=1}^{M}|h_{k,m}|^2}{N_{BS}} \end{cases} \tag{5.21}$$

where N_{BS} and N_R are the noise power at BS and relay, respectively. Note that the quality of the channel between MTs and the relay plays a key role. When this channel is worse than the direct one the maximum average capacity is given by the first term of 5.21. Otherwise, the maximum average capacity is given by the second term, assuming the relay can fully decode the information transmitted by the MT.

5.4 Numerical Results

In this section, we present and discuss the main simulation results that we used to assess the proposed cooperative schemes performance. Two metrics are used: outage capacity and BER.

5.4.1 Outage Capacity

We computed instantaneous capacity by generating several realizations of the channels according to a Rayleigh fading distribution, with several means, in a quasi-static channel. After that, we computed the outage capacities metrics. We define $\xi\%$ outage capacity as the information rate that is guaranteed for $(100 - \xi)\%$ of the channel realizations, that is, $P(C \leq C_{outage}) = \xi\%$ [9]. We assume perfect channels knowledge at the BS and the relay (for all cases). We compare the proposed cooperative schemes against the non-cooperative SISO and 1x2 SIMO systems, considering different PL values between the MTs and the BS. We assume for all the presented results a nonexistent PL between both MTs-Relay and Relay-BS,i.e., $E|h_{k,R}|^2 = 1$ and $E|h_{R,m}|^2 = 1$.

We refer to the cooperative relay-assisted schemes as VMISO AF or SDF when the MT, relay and the BS are equipped with single antenna. For the case when the BS is equipped with an antenna array with two elements and both the MT and relay with single antenna we refer to the same schemes as VMIMO AF or SDF. We assume that the antenna elements are sufficiently far apart to assume M independent channels, i.e., independent fading processes. Furthermore, we normalize the overall transmitted power to one for all the considered schemes.

This metric is more realistic than ergodic capacity, since some wireless services have minimum requirements on the supported data rates, below which the service is unsustainable. Therefore, we observe an outage if the achievable random rates fall below a certain level [49]. It should be emphasized that the outage capacity of the half-duplex cooperative schemes is expected to increase when compared to the non-cooperative ones. However, it would also be expected that the ergodic capacity of the cooperative schemes would drop significantly since redundant information is being transmitted.

Dependance on E_b/N_0

Figure 5.11 shows the performance results of the non and cooperative schemes in terms of the 5% outage capacity of the different schemes as a function of E_b/N_0, where E_b is the transmitted energy per bit, No/2 is the bilateral power spectral density of the noise and when the PL is 10dB.

For the case when all terminals are equipped with a single antenna the performance of the cooperative relay-assisted schemes outperforms the non-cooperative SISO system for all values of SNR, because in the former schemes extra information is transmitted from relay and thus outage is much lower. Comparing the proposed relay-assisted schemes, we can see that the VMISO SDF outperforms VMISO AF. For the VMISO SDF scheme, in the outage case where the relay fails to decode the data correctly, the BS only uses the signal received directly from the MTs. However, as SNR increases, the performance of all the relay-assisted schemes tends to be the same. When the BS is equipped with an antenna array the cooperative relay-assisted only outperforms the non-cooperative SIMO system for low values of SNR, up to 15 and 20 dB for VMIMO AF and SDF, respectively. This is due to the fact that at high SNR the probability of outage in non-cooperative SIMO transmission is low enough to not suffer serious signal degradation. From this figure, we can see that for 95% of the channel realizations and for a SNR=15dB, VMISO SDF and VMIMO SDF

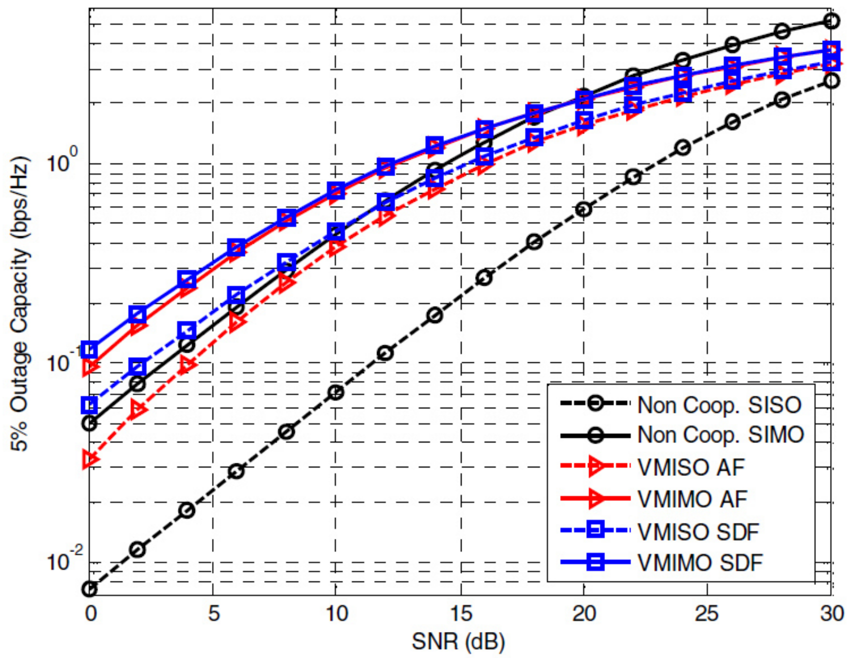


Figure 5.11: Outage capacity as function of SNR for a PL=10dB and considering one and two antennas at the BS.

achieve a capacity of approximately 0.9 and 1.3 bps/Hz while non-cooperative SISO and MISO systems only achieve approximately 0.2 and 1.0 bps/Hz, respectively.

Dependance on the Path Loss

The results leading with the Figure 5.12 show the 5% outage capacity, for the same schemes, but as function of the path loss for a $SNR = 8dB$.

We note that as PL increases, the difference between the performance of the cooperative schemes and non-cooperative ones dramatically increases, mainly for the case when all terminals are equipped with single antenna. When the BS is equipped with 2 antennas, the outage capacity of the cooperative schemes VMIMO SDF and VMIMO AF is higher than non-cooperative SIMO for a PL up to 4 and 6 dB, respectively.

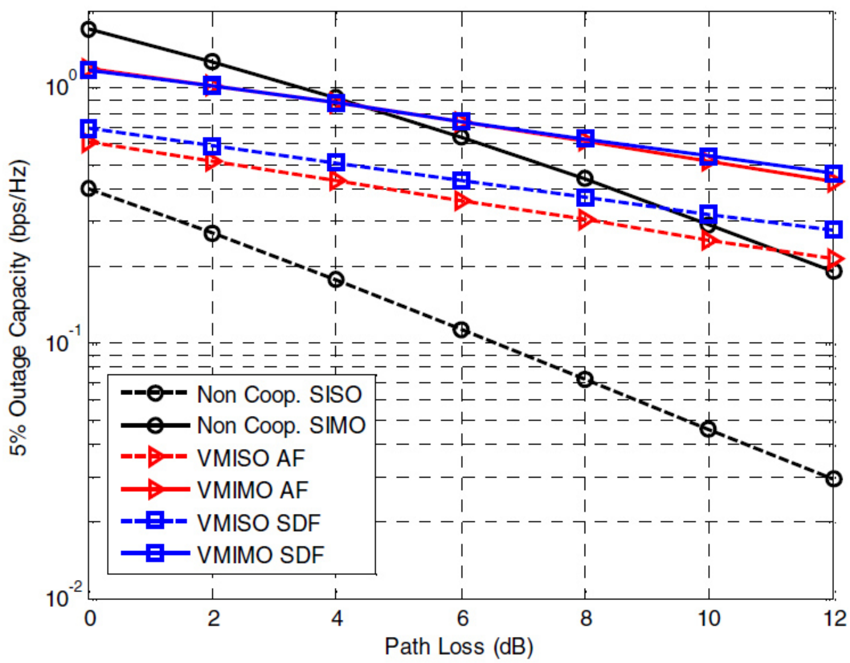


Figure 5.12: Outage capacity as function of path loss for SNR=8dB and considering one and two antennas at the BS.

5.4.2 Bit Error Rate

In order to assess the cooperative schemes in realistic scenarios we used a Rayleigh fading channel, whose system parameters are derived from the European BRAN Hiperlan/2 standardization project [36]. We extended these time models to space-time, assuming that the distance between antenna elements is far apart to assume M independent channels for each user.

Bit Error Rate simulations were done in the Cocentric System Studio (CCSS) environment and the main parameters used in the simulations are the ones that we present in table 5.2. It includes the number of carriers (N_c); Guard Period specification (GP); number of users (K) ; channel bandwidth; OFDM symbol duration and the used channel profiles. It is assumed that the receivers (BS and relay) have perfect knowledge of the channel. The transmitter power is normalized to one in all presented schemes.

To have the same spectral efficiency in all systems, the modulation scheme is QPSK for the cooperative based systems and BPSK for the non-cooperative systems. In fact, cooperation schemes require a 2-phase communication cycle, and that makes them two times less spectrally efficient than the classical ones. On the other side, QPSK is two times more efficient than BPSK. Thus, if cooperative schemes employ BPSK and the non-cooperative uses use QPSK, they all end up with the same spectral efficiency.

It is also worth mentioning that channel BRAN E model was extended to a space time model.

Also importantly, throughout this chapter, we refer to “path losses” of 0dB, 10dB and so forth. It is important to note that those are *relative* path loss values, and the channels which are 0dB are the best ones. In the same way, “path gains” are also defined with respect the the channels which are 0dB.

In the appendix of this dissertation, chapter B is devoted to provide more details regarding the simulation tool CCSS.

Number of carriers (N_c)	1024
Guard Period	256 samples / 3.2 μs
Number of Users (K)	32
Bandwidth	64MHz
OFDM symbol duration(T_s)	16μs
Modulation Scheme	BPSK (non-coop.) QPSK (coop.)
Channel Profile	ETSI BRAN E
Number of Taps	18
Maximum Delay	1.76 μs

Table 5.2: Main Simulation Parameters.

Dependance on E_b/N_0

Figure 5.13 shows the performance results of the non and cooperative schemes in terms of the average bit error rate as function of E_b/N_0, where E_b is the transmitted energy per bit (normalized to one) and $N_0/2$ the bilateral power spectral density of the noise. These results were obtained considering that the path losses of the different channels do not exist.

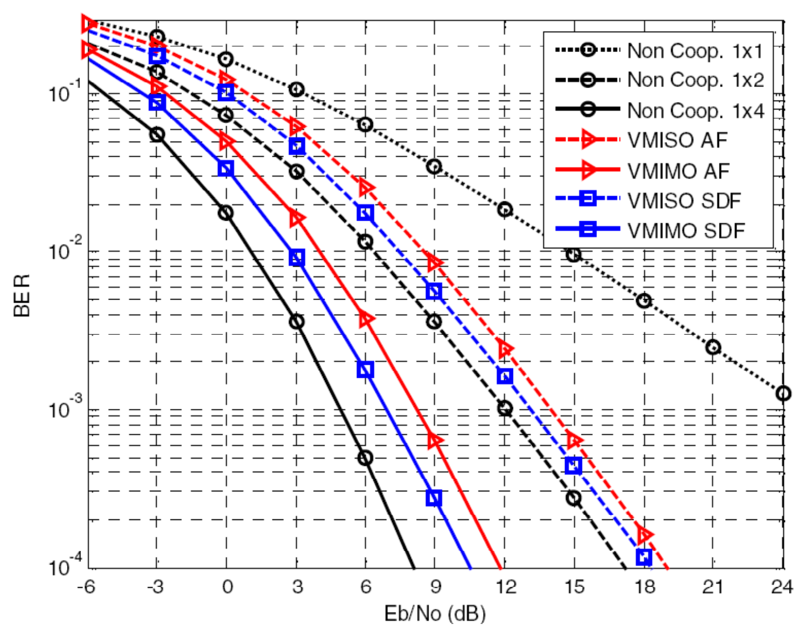


Figure 5.13: Performance comparison of non and cooperative schemes for a $PL = 0$ dB.

This figure suggests the following observations:

- VMISO AF and SDF outperform the non-cooperative 1x1 SISO system, but they are not as good as 1x2 SIMO.

- VMIMO AF and SDF outperform the non-cooperative 1x2 SIMO system. As a matter of fact, their performance is between that of 1x2 SIMO and 1x4 SIMO.
- The performance improvement of VMISO cooperative schemes against non-cooperative is lower than in the antenna array case.
- The selective DF cooperative scheme outperforms the other cooperative schemes, as it only retransmits to the BS the information that was successfully detected at the relay. This means that if the channels between MTs and relay are in outage, the relay does not take part as the decoding set in phase two and thus it does not increase the overall system performance. From the figure we can see, for a BER target of 1.0e-3, a gain of about 12 dB and 5 dB of the VMISO SDF and VMIMO SDF against non-cooperative 1x1 SISO and 1x2 SIMO, respectively.

The most remarkable conclusion that can be taken from Fig. 5.13 is that, even when the path losses in the MTk-BS link are not severe, cooperating still bring diversity gains.

Figure 5.14 shows the performance of the same schemes but now considering that the path loss between MTs and BS is 10 dB.

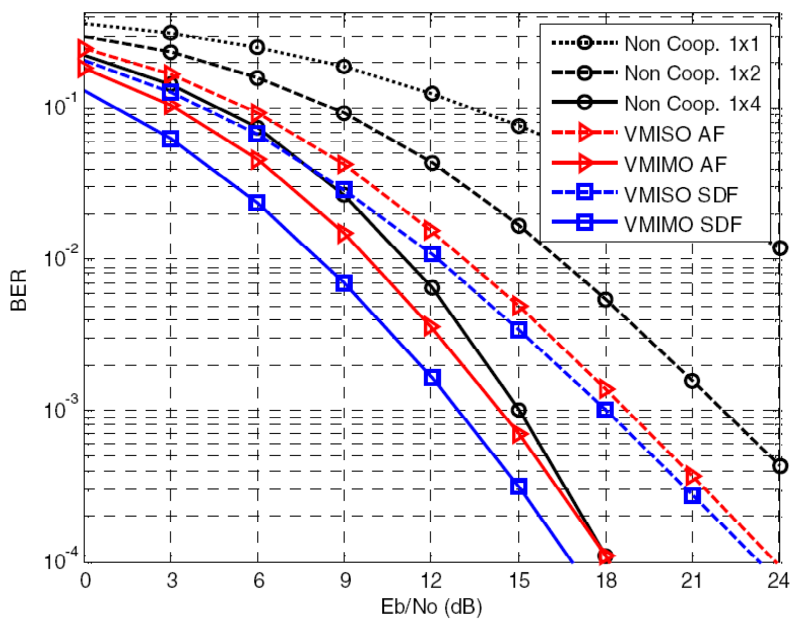


Figure 5.14: Performance comparison of non and cooperative schemes for $PL = 10$ dB.

The main observations are:

- As it happened in Fig. 5.13, SDF outperforms AF.
- For E_b/N_0 values up to 18dB, VMIMO outperforms the non-cooperative 1x4 SIMO system. This means that when the direct link is in outage the use of cooperative based schemes dramatically increases the system performance.
- VMISO's performance is between that of 1x2 SIMO and 1x4 SIMO.

The scenarios in which the relative path loss in the MTk-BS link is high evidence the performance improvement of cooperation based schemes. In fig. 5.14, for the VMISO SDF and VMIMO SDF and a BER target of 1.0e-2, we obtained a gain of approximately 12 dB and 8 dB against non-cooperative 1x1 SISO and 1x2 SIMO, respectively.

Cooperation schemes can be proposed as a "mode", which is turned on when the path loss in the direct link is really high and turned off when it is not so relevant.

Figure 5.15 shows the performance of the same schemes presented in Figure 5.13 with the difference being the fact that the channels between MTs and relay are 10dB better that the other channels. The purpose of this set of simulations was to investigate the impact that the quality of the MTk-Relay link has on the overall cooperative systems performance.

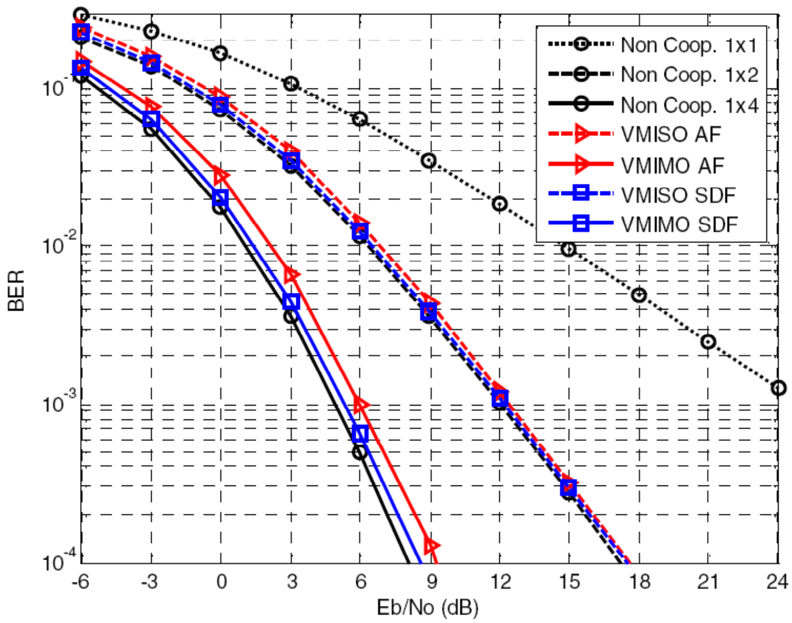


Figure 5.15: Performance comparison of non and cooperative schemes when the channel between MTs and relay is 10 dB better that the other channels.

From this figure we can observe that the performance of all cooperative schemes is improved in comparison with the previous scenario. The performance of the VMISO and VMIMO schemes is very close to the ones obtained by the non-cooperative 1x2 SIMO and 1x4 SIMO. Here, the kind of processing that is taking place at the relay (AF or SDF) does not seem to have a severe impact on the overall scheme performance. The explanation for this is that when the MTs-Relay channels are good, the probability of outage decreases and almost all of the data are successfully decoded at the relay (when SDF is being used) or amplified (when the relays performs AF).

Dependance on the Path Loss

Figure 5.16 aims at evaluating out the robustness of the proposed schemes against path loss between MTs and BS for SNR= 8dB. Once again, this scenario evidences the performance improvement of cooperation based schemes.

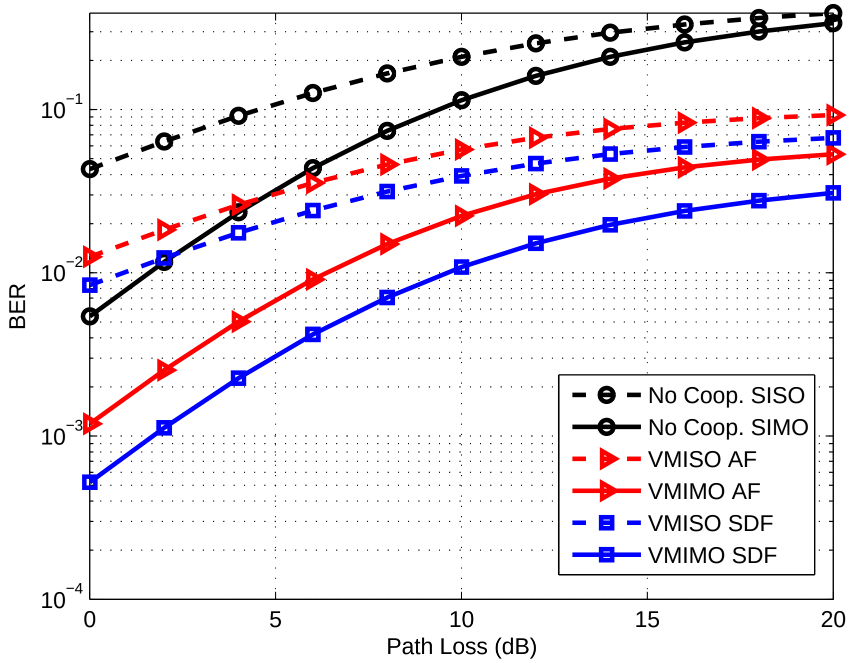


Figure 5.16: Performance evaluation of the robustness of cooperative schemes against path loss between MTs and BS (SNR=8dB).

For low path loss values, the non-cooperative schemes outperform all VMISO cooperative systems. However, as the path loss increases, the SDF and AF have increasingly higher BER, but not as much as the non-cooperative SIMO and MIMO. As expected, VMIMO schemes yield better BER results than VMISO. VMIMO AF and SDF are superior to SISO and 1x2 SIMO in the evaluated path loss range (0 to 20 dB's).

But the main point is that, despite cooperative schemes have a BER which increases with the path loss, that increase is not as high as the one that the non cooperative schemes experience. Stated another way, cooperative schemes are more robust against degradation in the MTk-BS link.

Figure 5.17 aims at evaluating the impact that the MTs-Relay channels have on BER.

We observe that when this channel is good, that is, as the path gain increases, all VMISO schemes emulate 1x2 SIMO perfectly, and all VMIMO schemes emulate 1x4 SIMO.

This is a very important result, since it evidences the impact that quality of the MTs-Relay channel has on he overall performance of the cooperative schemes. Indeed, when this channel is good, the cooperation strategy (AF or SDF) that is being used is not the most important factor for the schemes performance. What is relevant is the MTk which is chosen to act as relay, since it will impact on the quality of the MTk-Relay channel.

Therefore, the results suggest that the algorithms that are used to select the MT's that will act as relays might be more important for the success of the cooperation strategy than the processing that takes place at the relay. Such observations agree with [37].

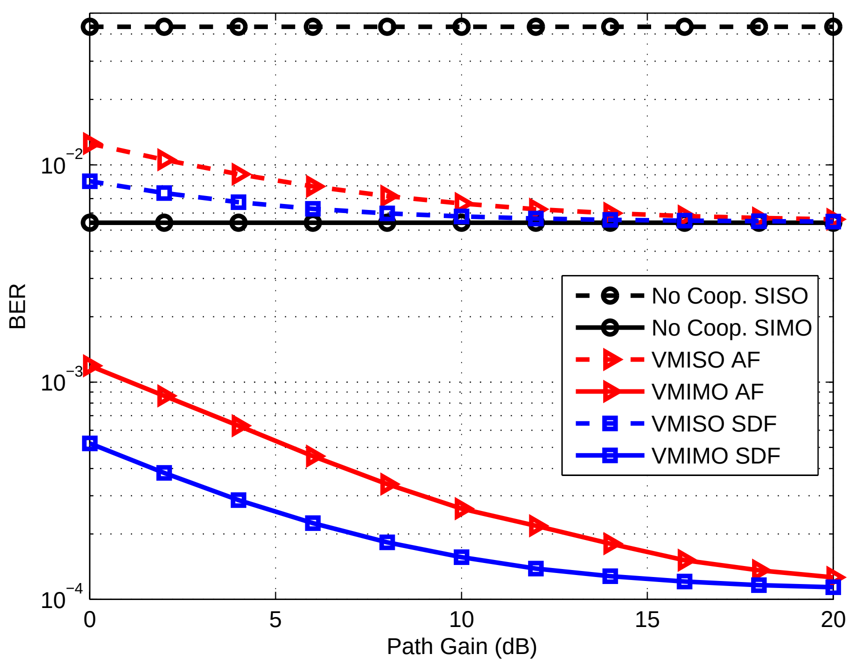


Figure 5.17: Performance evaluation of the robustness of cooperative schemes against path loss between MTs and Relay (SNR=8dB).

BER penalty due to Non-Optimum processing at the BS

Figures 5.18 and 5.19 shows the BER for the optimum and non-optimum AF schemes as function of E_b/N_o for a Path Loss in the MT$_k$–BS channel of $PL = 0dB$ and $PL = 10dB$. In short, it evidences that the non-optimum processing reduces the BER, but the BER is still comparable or inferior to the one that can be achieved without cooperation at all.

Fig. 5.18 suggests that when there are no path losses, the sub-optimal virtual MISO and MIMO AF schemes perform worse than the optimal VMISO and VMIMO AF ones, respectively. And what is most remarkable is that the performance differences increase as the transmit power increases. For low values of E_b/N_0 (below 5 dB), VMISO Sub-Opt. AF and VMISO AF perform similarly, but as E_b/N_0 increases, the performance differences between VMIMO Sub-Opt. AF and VMIMO Opt. AF become higher.

In this scenario, the sub-optimal AF schemes still bring performance enhancements over the non-cooperative 1x1 systems, but not over 1x2 systems.

Fig. 5.19 evidences that when the MT_k–BS channel is 10dB worse than the remaining ones, the sub-optimal virtual MISO and MIMO AF schemes perform worse than the VMISO and VMIMO AF ones, respectively. The exception is VMISO for $E_b/N_0 < 13dB$. Again, the performance differences increase as the transmit power increases.

In this scenario, the sub-optimal AF schemes still yield significant performance enhancements over the non-cooperative 1x1 systems and over 1x2 systems.

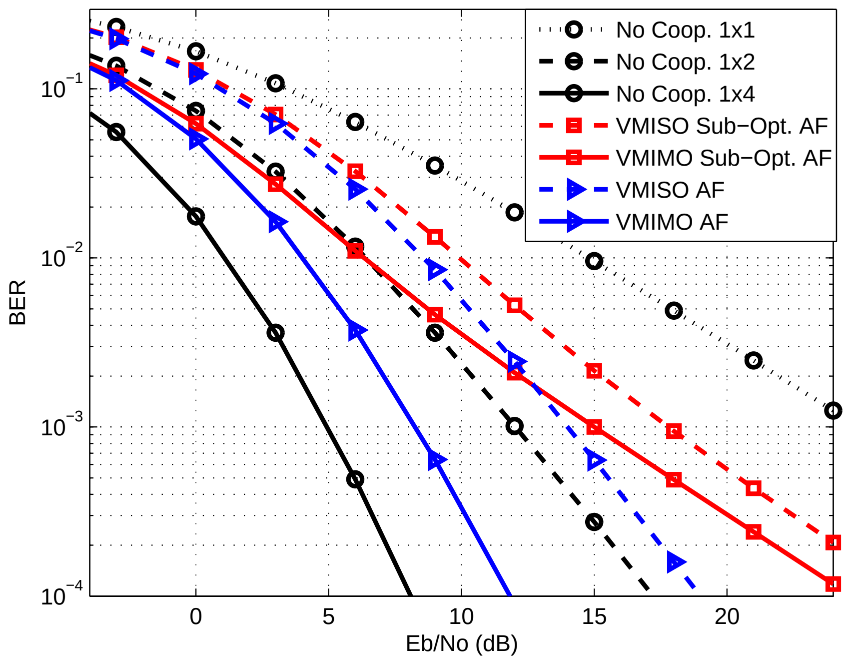


Figure 5.18: AF ft Sub-Opt. AF based on BER for $PL_{km} = 0$ dB

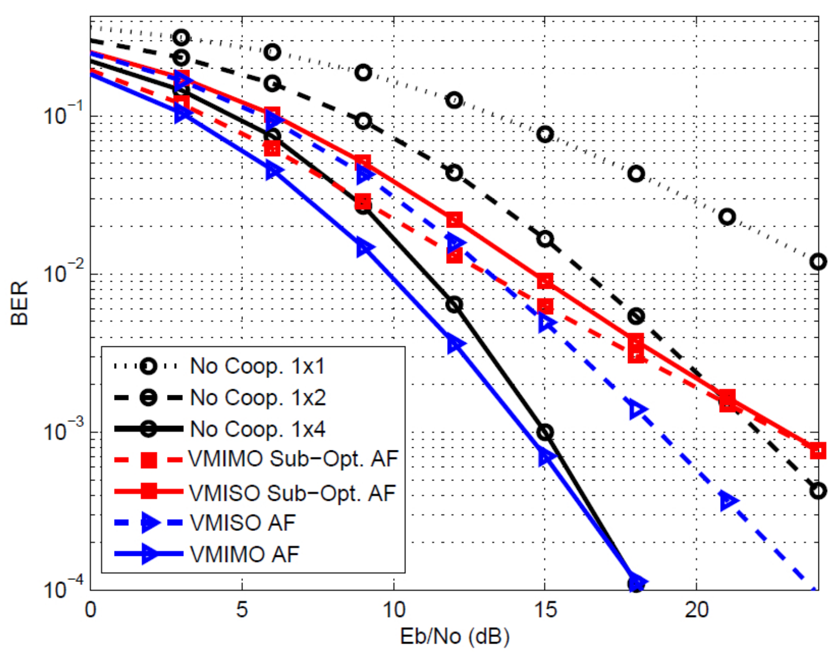


Figure 5.19: AF ft Sub-Opt. AF based on BER for $PL_{km} = 10$ dB.

Chapter 6

CONCLUSIONS

The main goal stated at the beginning of this dissertation was to study, implement and evaluate the performance of low-complexity cooperative diversity schemes projected for mobile communication systems.

First, a framework was defined for the scenario of several users that transmit to a BS. Through a revision of the state of the art and general considerations of both diversity techniques and cooperative procedures, AF and SDF cooperative schemes were mapped into simulation models based on OFDMA and were fully simulated in the CoCentric System Studio environment. The obtained results showed that the proposed cooperative schemes for the uplink communication mitigate fading induced by multipath propagation, thereby increasing the capacity and coverage of wireless systems. Cooperation gains were particularly high when multipath losses are considerable, as is the case for dense urban regions.

This chapter includes a section that contrasts the proposed objectives and the achieved results (section 6.1). Then it proceeds with section 6.2 which discusses possible extensions of the work developed in this dissertation.

6.1 Achieved Results

We proposed and evaluated virtual MIMO schemes designed for the UL OFDMA based systems. Two types of relays were analyzed: amplify and forward and selective decode and forward. We derived the instantaneous normalized capacities and both the outage capacities and BER were compared against the non-cooperative SISO and SIMO systems, considering different scenarios:

Similar path losses in all links Here the interuser(MTs-Relay), direct(MTs-BS) and relay-BS links have the same quality. The purpose of this scenario is to check if there are cooperation gains when the direct link quality is comparable to the one that is created by the relay.

Path loss in the direct link Here the path gain of the interuser channel was made 10 dB with respect to the remaining ones. The purpose was to test the impact of using cooperation when the direct link is poor on the overall system performance.

Path gain in the MT$_k$–relay link Here the path gain of the interuser channel was made 10 dB with respect to the remaining ones. The purpose was to test the impact of the quality of the interuser channel on the overall system performance.

The outage capacities and BER analysis were made as a function of E_b/N_0 and also as a function of the path losses/gains of the direct/interuser channels. The idea behind investigating the impact of the path losses/gains on the outage capacities and BER was to study the robustness of these systems against degraded communication conditions.

The results have shown that:

- The performance of the proposed cooperative schemes dramatically increases as compared with the non-cooperative SISO and SIMO systems;
- SDF based relay schemes outperforms the AF ones in all the studied scenarios;
- A sub-optimum AF scheme can be designed in such a way that the MRC at the BS does not need to estimate the cooperation link variance. This scheme brings computational complexity benefits in comparison with VMISO/VMIMO AF at the expense of BER penalties. Still, these schemes still emultate MIMO when path losses are severe;
- All cooperative schemes are more robust against path losses in the direct link than the classical schemes (P2P).

It is clear from the presented results that the proposed cooperative schemes, mainly the SDF scheme, allow a significant improvement in user capacity and coverage than non-cooperative systems *above all* in dense urban scenarios where a severe path loss does exist.

6.2 Extensions and continuing work

While some key results for cooperative communication have already been obtained, there are many more issues that remain to be addressed.

Doppler Effect Consideration

Since CODIV's vision for 4G is that the MTs and Relays can be mobile, it is of prime importance to take the Doppler effect into consideration. In fact, 4G mobile users will demand broadband internet access anywhere (cars, trains, airplanes included) and, therefore, future systems will have to designed/tested under these "worst case" conditions. Therefore, the channel models that were used in the simulations of these thesis would be more realistic if they could model motion in the MTs and Relay.

Channel Coding

The simulation results of this dissertation would be more close to reality if channel coding was used.

For the coded cooperation method, a natural issue is the possibility of designing a better coding scheme. In [27], convolutional codes are applied to the coded cooperation framework. These coding schemes were originally developed for noncooperative systems. An interesting open problem is the development of design criteria specifically for codes that optimize the performance of coded cooperation.[3]

Convolutional and Turbo codes decrease the transmission efficiency but bring coding gains. They can be worthy in poor communication conditions.

The Potential Advantages of Using an Antenna Array at the Relay

Within this dissertation, the UL communication scenario included a Relay, whose role was performed by a mobile terminal. Here it was assumed that mobile terminal was likely to be a small mobile phone, and that it was not likely to have an antenna array. But it can also happen that the MT is a laptop or even a PDA. In this case, the availability of antenna arrays is more likely. This case suggests that it would also be pertinent to design and simulate transmission chains where the relay is equipped with an antenna array (for example, a two element antenna array).

Another UL cooperation scenarios where the relays are likely to have antenna arrays are the ones that include dedicated relays. These are important scenarios, since they are the ones that the IEEE 802.16j working group is exploring[25].

One-Hop Cooperation Models

Recall that in the introduction of this dissertation (section 1.2.3) it was explained that there are four major possibilities for the one-hop cooperation scheme:

1. $MT_k \rightarrow (Relay, BS); (MT_k, Relay) \rightarrow BS$ (most general form of relaying);
2. $MT_k \rightarrow Relay; (MT_k, Relay) \rightarrow BS$ (BS ignores signal from MT_k in first mode);
3. $MT_k \rightarrow (Relay, BS); Relay \rightarrow BS$ **(**MT_k **does not transmit in second mode)**;
4. $MT_k \rightarrow Relay; Relay \rightarrow BS$ (multi-hop communication).

Nevertheless, there are scenarios where the other models would be interesting. For example, the first model allows the BS to get 2 copies of the transmitted symbol by the direct MT_k-BS link. Even if this channel is bad, this is still an advantage in comparison with the third model, as it includes an extra symbol to perform MRC at the BS. In fact, this is a way of exploring time diversity (the MT_k-BS link used twice to transmit the same information) and cooperative diversity at the same time. The disadvantage is the fact that the MT_k must transmit in the full communication cycle and use channel code to provide orthogonality between the MT_k and relay.

Neighbor Selection Strategies

An important question is how partners are assigned and managed in multi-user networks. In other words, how is it determined which users cooperate with each other, and how often are partners reassigned? Systems such as cellular, in which the users communicate with a central base station, offer the possibility of a centralized mechanism. Assuming that the base station has some knowledge of the all the channels between users, partners could be assigned to optimize a given performance criterion, such as the average frame error rate for all users in the network. In contrast, systems such as ad hoc networks and sensor networks typically do not have any centralized control. Such systems therefore require a distributed cooperative protocol, in which users are able to independently decide with whom to cooperate at any given time. A related issue is the extension of the proposed cooperative methods to allow a user to have multiple partners. The challenge here is to develop a scheme that treats all users fairly, does not require significant additional system resources, and can be implemented feasibly in conjunction with the system's multiple access protocol. Laneman and Wornell [30] have done some initial work related to distributed partner assignment and multiple partners, and additional work by others is ongoing.[3]

On the Field Testing

It would be very interesting to see the actual results of cooperation on the field. We bear in mind that the results that were obtained with simulations often differ from the practical ones. The reasons for this are manyfold; firstly, the used channel model might not translate all the distortive effect of the channel, user interference, used equipment and modulation lack of robustness. It is important to see if the cooperative gains are large enough to make up for its “implementation costs”.

One of CODIV goals is to implement some of these cooperative algorithms in a prototype.

CONTRIBUTIONS TO CONFERENCES

The main results obtained within the scope of this dissertation were resumed into a paper which is entitled: "Performance Evaluation of Virtual MIMO Schemes for the UL OFDMA Based Systems" [39]. It was submitted and accepted for the The Fourth International Conf. on Wireless and Mobile Communications, which took place in Athens, Greece from 27 July to 1 August 2008.

FUTURE APPLICATION OF THE WORK DEVELOPED IN THE SCOPE OF THIS THESIS

The insight developed in the scope of this thesis with respect to user-cooperation justify the belief that cooperative diversity has potential for implementation in mobile handheld devices, but here fair sharing of resources must be ensured by a suitable protocol. It can be anticipated that the current understanding of the principles of user-cooperation, together with advances in technology will enable cooperative communication networks in future.

Appendix A

Matlab code for plotting CDF's of SISO/SIMO systems

Hereto we include the Matlab code that was used for plotting CDF's of SISO/SIMO systems. The interested reader is challenged to verify that A.1 can be easily extended to plot CDF's of cooperative systems.

Listing A.1: cdf_comparison.m: CDF graphical comparison of SISO/SIMO systems

```
function [ CDFsiso , x_siso , CDFsimo , x_simo ]= cdf_comparison
%CDF_COMPARISON this script performs a CDF comparison between :
%-> SISO
%-> SIMO with M=2
% Syntax : [ CDFsiso , x_siso , CDFsimo , x_simo ]= CDF_COMPARISON
%%%%%%%%%%%%%%%%%% Initial Parameters %%%%%%%%%%%%%%%%%%%%%%%%%%%
N=1e6 ; Nhist=1e5 ; %Number of generated bits
Eb_No=15; %Eb_No (dB)
PL=6; ; %h_km Path Loss (dB)
SNR=10.^(Eb_No/10) ; Var=1./SNR; %Assuming received power=1
%%% Channel simulation and Capacities computation for each PL and SNR %%%
alpha1=10^(-PL/10);
[ P1_1 , h1_1 ]= myRayleigh (alpha1 , N) ; %Channel between MT1 and
% antena 1 of the BS
[ P1_2 , h1_2 ]= myRayleigh (alpha1 , N) ; %Channel between MT1 and
% antena 2 of the BS
%Channel Modeling and Capacity computation ( random variables )
% non-cooperative siso and simo
Csiso = log2( 1+( P1_1 )*SNR ) ; %BS has 1 receiving antenna (M=1)
Csimo = log2( 1+( P1_1+P1_2 )*SNR ) ; %BS has 2 receiving antenna (M=2)
%.......................................
%Capacity histograms
[hist_siso , x_siso]=hist(Csiso , Nhist) ;
[hist_simo , x_simo]=hist(Csimo , Nhist) ;
```

```
%Cumulative Density Functions
CDF_siso=cumsum(hist_siso)/N;
CDF_simo=cumsum(hist_simo)/N;
%.........................................
%Graphics : CDF as function of BitRate ( bps/Hz )
figure
plot ( x_siso , CDFsiso , 'b.-' ,...
xsimo , CDFsimo , 'k-*' )
grid on
legend('SISO','MISO','Location','Best')
ylabel('CDF') , xlabel ( 'Bit Rate (bps/Hz)' )
%.........................................
% SUBFUNCTIONS
%.........................................
function [ power , h ]= myRayleigh (alpha , N)
%MYRAYLEIGH Computes a Rayleigh fading channel impulse response
% and its power delay profile .
% Syntax : [POWER, H]= MYRAYLEIGH(ALPHA, N)
%
% Input paramaters : ALPHA: path attenuation factor [ 0 , 1 ]
% N : number of generated bits
% Output parameters : POWER: H's power delay profile (Nx1 vector)
% H : channel impulse response (Nx1 vector)
% $Revision : 1.0 $ $Date : 2008/02/02 17:05:10 $
h=sqrt(alpha/2)*( randn(N,1) + j*randn(N, 1) );
power=abs(h).^2;
```

Appendix B

Cocentric System Studio Environment

B.1 Cocentric System Studio

B.2 Overview

`System Studio` is the high performance model-based algorithm design and analysis tool, combining simulation performance and modeling efficiency, plus industry's integration into the implementation design and verification flow. Algorithm design is an essential task in signal processing applications such as wireless telephony. According to [53], more than 50% of all mobile phones worldwide rely on algorithms designed with System Studio, making it the clear market leader. `System Studio` offers an additional advantage to the systems designer, which is being easy to work with. Fig. B.1 shows that System Studio's model-based design concept provides highest design efficiency by providing a development environment with both graphical and language abstractions that capture the entire system in a hierarchical fashion.

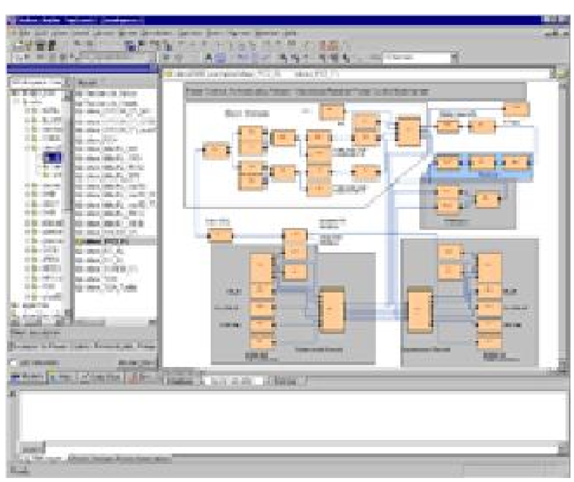


Figure B.1: System Studio's model-based design environment.

B.3 System Level Solution

`System Studio`(CCSS) addresses this design challenge by providing a model-based electronic system-level design creation, simulation and analysis environment. `System Studio` provides extensive support for the design and analysis of complex signal processing functions such as multi-antenna receiver algorithms, multimedia processing, and communication standards compliance. It offers a rich set of analysis functions that designers use to meet the requirements for high-quality user experience with speech, multimedia, and Internet connectivity given the imperfections introduced by transmission, non-ideal analog components, and fixed-point digital implementation.

Within the scope of this thesis, CCSS is used to implement transmission chains in an hierarchical fashion and then simulate it. Each transmission chain is built using functional blocks, that can be created by the user of provided by CCSS by default. And each of these blocks can be implemented in a lower hierarchical level by simpler blocks or in C language. In this way, the designer can manage complexity by working in a *divide-and-conquer* fashion.

Figure B.2 shows an example of a simple chain that was implemented in CCSS.

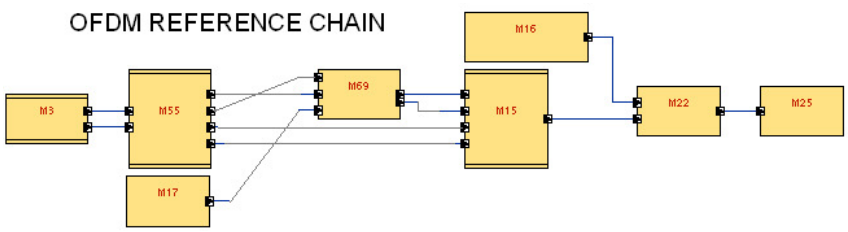


Figure B.2: Within the scope of this thesis, CCSS is used to implement and simulate transmission chains in such a way that complexity is "hidden" in lower implementation layers.

The meaning of the blocks is as follows:

M3 OFDM symbol transmitter;

M55 Rayleigh Channel simulator;

M69 AWGN noise simulator;

M15 Channel Equalization and demodulation;

M16 and M22 Bit error test;

M25 BER statistics.

Its dataflow simulation engine provides simulation performance necessary to explore the design space in an acceptable amount of time. According to [53], its fixed point simulation acceleration concepts allow it to achieve a speed improvement of 10x for typical mixed floating-point/fixed-point simulation and up to 200x for fixed-point only simulation. System Studio's intuitive graphical user interface and extensive model libraries and reference design kits jump-start commercial design efforts in the areas of advanced wireless, multimedia and telecom technical standards.

`System Studio`'s fast performance on single simulation runs is enhanced on compute clusters, taking advantage of multicore architectures, by its native capability to distribute simulation iteration runs to multiple processing elements, then merging the simulation results into a report. Nevertheless, we did not explore the parallel processing possibilities that `Cocentric System Studio` offers and ran it on a single core Linux machine instead.

The interested reader is referred to [52], where the `Cocentric System Studio` capabilities are discussed in more detail.

Appendix C

Power Delay Profiles of HIPERLAN/2 models

C.1 Channel Model A

Delay [ns]	Variance [dB]
0	0
10	-0.9
20	- 1.7
30	- 2.6
40	- 3.7
50	- 4.3
60	- 5.2
70	- 6.1
80	- 6.9
90	- 7.8
110	- 4.7
140	- 7.3
170	- 9.9
200	- 12.5
240	- 13.7
290	- 18.0
340	- 22.4
390	- 26.7

Table C.1: Power delay profile of channel A [36].

C.2 Channel Model E

Delay [ns]	Variance [dB]
0	- 4.9
10	- 5.1
20	- 5.2
40	- 0.8
70	- 1.3
100	- 1.9
140	- 0.3
190	- 1.2
240	- 2.1
320	- 0
430	- 1.9
560	- 2.8
710	- 5.4
880	- 7.3
1070	- 10.6
1280	- 13.4
1510	- 17.4
1760	- 20.9

Table C.2: Power delay profile of channel E [36].

Bibliography

[1] Web Site for the 3rd Generation Partnership Project (3GPP). Available at: http://www.3gpp.org/. Accessed September 19, 2008.

[2] Web Site for the European IST project 4MORE.Available at: http://www.ist-4more.org/. Accessed October 5, 2008.

[3] Aria Nosratinia, Todd E. Hunter, Ahmadreza Hedayat, "Cooperative Communication in Wireless Networks",IEEE Communications Magazine , October 2004.

[4] Ahmad R. Bahai and Burton R. Saltzberg. *Multi-Carrier Digital Communications: Theory and Applications of Ofdm.* Plenum Publishing Co.,1999.

[5] Alamuti, S. "A Simple Transmit Diversity Technique for Wireless Communications," IEEE JSAC, 16(8), October 1998.

[6] Peter Almers, Fredrik Tufvesson, ,Andreas F. Molisch, "Keyhole Effect in MIMO Wireless Channels: Measurements and Theory," IEEE Trans. on Wireless Communications, Vol. 5, NO. 12, Dec. 2006.

[7] Jeffrey G. Andrews, Arunabha Ghosh, Rias Muhamed, *Fundamentals of WiMAX: Understanding Broadband Wireless Networking*, Feb 27, 2007, Prentice Hall.

[8] P. A. Bello, "Characterization of randomly Time-Variant Linear Channels," IEEE Trans. on Vehicular Tech., Vol. 11, Dec. 1963.

[9] E. Biglieri, J. Proakis and S. Shami, "Fading Channels: Information Theoretic and Communications Aspects," IEEE Trans. Inf. Theory, Vol. 44, No. 6, Oct. 1998.

[10] R. H. Clark, "A Statistical Theory of Mobile-Radio Reception," Bell Systems Technologies Journal, Vol. 47, 1968.

[11] Web Site for CODIV – Enhanced Wireless Communication Systems Employing **CO**operative **DIV**ersity Group. Available at: https://projects.av.it.pt/. Accessed October 9, 2008.

[12] T. M. Cover, A. A. E. Gamal, "Capacity Theorems for the Relay Channel," IEEE Transactions on Information Theory, vol. 25, no. 5, pp. 572-584, Sep. 1979.

[13] M. Dohler, Virtual Antenna Arrays, Ph.D. Thesis, King's College London, London, UK, Nov. 2003.

[14] European Telecommunications Standards Institute. Available at: http://portal.etsi.org. Accessed June 5, 2008.

[15] G. J. Foschini and M. J. Gans, "On limits of wireless communications in a fading environment when using multiple antennas," Wireless Personal Communications Magazine, Vol. 6, No. 3, Mar. 1998.

[16] Frank H. P. Fitzek and Marcos D. Katz. *Cooperation in Wireless Networks: Principles and Applications*, Springer, 2006.

[17] D. Gesbert, M. Shafi, D.S. Shiu, P. J. Smith, A. Naguib, "From Theory to Practice: An Overview of MIMO Space-time Coded Wireless Systems," IEEE JSAC, Special Issue on MIMO Systems Part I, March 2003.

[18] A. Goldsmith. *Wireless Communications*, Cambridge University Press, New York, NY, USA, 2005.

[19] M. Hata, "Empirical Formula for Propagation Loss in Land Mobile Radio Services," IEEE Trans. on Vehicular Tech., August 1980, pp. 317-325.

[20] S. Haykin and M. Moher. *Modern Wireless Communication.* Prentice-Hall, Inc., Upper Saddle River, NJ, USA,2004.

[21] Heung-Gyoon Ryu, Sang Burm Ryu, Seon-Ae Kim, "Design and Performance Evaluation of the MIMO SFBC CI-OFDM Communication System," IWCMC '08. The Fourth International Conf. on Wireless and Mobile Communications, Athens, Greece, 27 July - 1 August 2008.

[22] Ho-Jung and Hyoung-Kyu Song, "Cooperative Communication in SIMO Systems with Multiuser-OFDM," IEEE Transactions on Consumer Electronics, Vol. 53, No 2, May 2007.

[23] Hui Liu and Guoqing Li, *OFDM-Based Broadband Wireless Networks*, John Wiley & Sons, Inc., 2005.

[24] Web Site for IEEE 802 LAN/MAN Standards Committee. Available at: `http://ieee802.org/index.html`, `http://grouper.ieee.org/groups/802/16/`.Accessed October 1, 2008.

[25] Web Site for IEEE 802.16j. Available at: `http://wirelessman.org/relay`. Accessed October 22, 2008.

[26] C.J.Jakes. *Microwave Mobile Communications.* IEEE Press,USA,1994.

[27] M. Janani et al., "Coded Cooperation in Wireless Communications: Space-Time Transmission and Iterative Decoding," IEEE Trans. Sig. Proc., vol. 52, no. 2, Feb. 2004, pp. 362-71.

[28] M. Jankiraman. *Space-Time Codes and MIMO Systems.* Artech House Boston Inc., 2004.

[29] J. N. Laneman, G. W. Wornell, and D. N. C. Tse, "An Efficient Protocol for Realizing Cooperative Diversity in Wireless Networks," Proc. IEEE ISIT, Washington, DC, June 2001, p. 294.

[30] J. N. Laneman, G. W. Wornell, "Distributed Space- Time-Coded Protocols for Exploiting Cooperative Diversity In Wireless Networks," IEEE Trans. Info. Theory, vol. 49, no. 10, Oct. 2003, pp. 2415-25.

[31] J. N. Laneman, D. N. C. Tse, and G. W. Wornell, "Cooperative diversity in wireless networks: Efficient protocols and outage behaviour," IEEE Trans. Inform. Theory, Vol. 50, No 12, pages: 3062-3080, Dec. 2004.

[32] R. Laroi, S. Uppala and J. Li, "Designing a Mobile Broadband Wireless Access Network," IEEE Signal Processing Magazine, Vol. 25, Sep. 2004.

[33] L. Litwin and M. Pugel, "The principles of OFDM," Thomson Multimedia, 2001.

[34] Matlab Central File Exchange Website. `http://www.mathworks.com/matlabcentral/index.html`. Accessed October 7, 2008.

[35] MATRICE – MC-CDMA Transmission Techniques for Integrated Broadband Cellular Systems. Available at: `http://matrice.av.it.pt/`. Accessed October 6, 2008.

[36] J.Medbo, P.Schramm, 'Channel Models for HIPERLAN/2 in different indoor scenarios', ETSI BRAN *doc. 3ERI085b*,1998.

[37] Meng Yu, Jing Li, "Is amplify-and-forward practically better than decode-and-forward or vice versa?," Acoustics, Speech, and Signal Processing, 2005. Proceedings. (ICASSP '05). IEEE International Conference on 18-23 March 2005, On page(s): iii/365- iii/368 Vol. 3.

[38] E. C. van der Meulen, "Three-terminal communication channels," Advances in Applied Probability, vol. 3, 1971, pp. 120-154

[39] A. V. Moço, S. Teodoro, A. Silva and A. Gameiro, "Performance Evaluation of Virtual MIMO Schemes for the UL OFDMA Based Systems," IWCMC '08. The Fourth International Conf. on Wireless and Mobile Communications, Athens, Greece, 27 July - 1 August 2008.

[40] Nabar, R.U.; Bolcskei, H.; Kneubuhler, F.W., "Fading relay channels: performance limits and space-time signal design," Selected Areas in Communications, IEEE Journal on Volume 22, Issue 6, Aug. 2004, pp. 1099 - 1109.

[41] R.D.J. van Nee and R. Prasad. *OFDM for Wireless Multimedia Communications.* Artech House Publishers, 1999.

[42] Y. Okumura et al., "Field Strength and Its Variability in VHF and UHF Land-Mobile Radio Service," Rev. of the Electrical Comm. Lab. (Japan), Sept./Oct. 1968, pp. 825-873.

[43] M. Patzold. *Mobile Fading Channels.* John Wiley & Sons, 2002.

[44] J.Proakis. *Digital Communications.* MacGraw-Hill, 1995.

[45] John G. Proakis, Dimitris G. Manolakis. *Digital Signal Processing; Principles, Algorithms, and Applications*, Fourth edition, 2007, Pearson Prentice-Hall, ISBN 0-13-187374-1.

[46] T.S.Rappaport. *Wireless Communications: Principles and Practise.* Prentice-Hall, 1996.

[47] Rysavy Research Web Site. Available at: `http://www.rysavy.com`. Accessed October 23, 2008.

[48] Tim C.W. Schenk, Peter F.M. Smulders and Erik R. Fledderus, ”Multiple Carriers in Wireless Communications – Curse or Blessing?,” Tijdschrift Ned. Elektron.- & Radiogenoot. (NERG), Vol. 70, No. 4, pp. 112-123, Dec. 2005.

[49] A. Sendonaris, E. Erkip, and B. Aazhang, ”User cooperation diversity–Part I: System description,” IEEE Trans. Commun., vol. 51, pp.1927-1938, Nov. 2003.

[50] R. Steele. *Mobile Radio Communication.* Pentech Press, 1994.

[51] G. Susinder Rajan and B. Sundar Rajan, ”Distributed Space-Time Codes for Cooperative Networks with Partial CSI,” Proceedings of WCNC 2007.

[52] Synopsys. *Getting Started With CoCentric System Studio*, v2002.05-SP1.

[53] Web Site for Synopsys. Available at: `http://www.synopsys.com/products/sls/system_studio/system_studio.html`. Accessed September 15, 2008.

[54] E. Telatar, ”Capacity of multi-antenna Gaussian channel,” European Trans. on Telecommunications, vol.lO, 1999.

[55] Web Site for the Telecommunications Institute. Available at: `http://www.it.pt/`. Accessed September 15, 2008.

[56] S. B. Weinstein, P. M. Ebert, ”Data Transmission by Frequency Division Multiplexing Using the Discrete Fourier Transform,” IEEE Trans. Commun. Tech., vol. COM-19, pp. 628-634, Oct. 1971.

[57] WiMAX.com Website . Available at: `http://www.wimax.com/`. Accessed December 11, 2007.

[58] WiMAX Fórum. Available at: `http://www.wimaxforum.com`. Accessed December 11, 2007.