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DEPARTAMENTO DE COMUNICACIONES

3GPP Long Term Evolution: Performance Analysis and Evolution towards 4G with Coordinated Multi-Point Transmission

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To my wife Ana and to my sons,
to my family and friends,
and to those who believe
that simulation is not playing dice.

Abstract

In today's information society, there is a growing need to access data communication services ubiquitously, with mobility and increasingly higher data rates. This society's demand together with the interests of operators, manufacturers and standardization bodies has motivated the development of the fourth generation of mobile communications (4G) and its evolution towards the fifth generation (5G). This development has required a revolution on the radio interface of the mobile communications systems, and, consequently, has significantly modified their capabilities and their radio resource management. This is the case of the technology known as Long Term Evolution (LTE) and its 4G version called LTE-Advanced.

This Doctoral Thesis addresses the modelling, the radio resource management analysis, and the performance evaluation of the downlink of LTE and LTE-Advanced where, among the different features of LTE-Advanced, the focus is on the Coordinated Multi-Point (CoMP) transmission.

The Thesis provides a detailed description of the main characteristics of LTE and LTE-Advanced. The high complexity of these systems, with multiple interrelated elements, has prompted the use of computer simulations as the primary research methodology. The Thesis makes a detailed description of the simulation methodology and the system modelling required by this methodology, including some contributions of the author in this field. Among them, it is of significant relevance the link-level simulation results used in the European project WINNER + for the LTE evaluation.

With regard to the analysis of the radio resource management in LTE, the fundamentals of link adaptation and scheduling are explained in the first place, describing the most common implemented algorithms. In relation to the scheduling, the Thesis includes a thorough study of the proportional fairness concept and the suboptimal implementation typically used in LTE to maximize this metric. This study has resulted in a series of ideas embodied in a modification of the typical implementation, which has proved to be capable of increasing the proportional fairness of the resource allocations. Moreover, the link adaptation analysis has revealed the "flash-light" effect problem, which is characterized by a high interference variability due rapid changes in the scheduling decisions. The Thesis demonstrates that a particular implementation that stabilizes the scheduling decisions can improve the system performance.

The radio resource management analysis of this Thesis is completed with the study of the LTE-Advanced capability to coordinate the transmissions of

multiple base stations, that is to say, CoMP. The current path towards the 5G has paid attention to techniques studied in the 3GPP in the first phase of the CoMP evolution, in which coordination is very fast. The CoMP proposal studied in this Thesis is a solution of this kind, with coordinated scheduling and beamforming (CS/CB), that takes into account realistic and robust assumptions concerning the knowledge that the coordinated points have about the channel state of the users they serve. The Thesis proposes this solution for its simplicity and its ability to improve high data rates coverage and capacity even with incomplete channel knowledge, in opposition to other joint transmission systems whose complexity is much greater and can be hardly damaged when they do not obtain a complete channel knowledge.

Concerning LTE and LTE-Advanced evaluation, it is performed in two different types of scenarios. On the one hand, the scenarios defined by the ITU-R in the process of evaluation of the IMT-Advanced candidate technologies. In this framework, it is evaluated the importance of different multi-antenna techniques, including CoMP, considering full-buffer traffic models. The most important conclusions in these scenarios are the significant performance improvement achieved with spatial multiplexing of users and the fact that CoMP mechanisms provide a reduced benefit. The second group of scenarios are those defined by the European project METIS for the evaluation of 5G technologies. Specifically, an indoor office scenario and an outdoor sports stadium have been selected. In these scenarios, a realistic traffic model is used, and it has been demonstrated the utility of CoMP to satisfy the first 5G requirement definitions with feasible frequency bandwidths. In these scenarios with less homogeneous deployments, or with a limited number of transmitters originating the major part of interference, is where this Thesis has found CoMP to be more useful and where the Thesis promotes its use.

Resumen

En la actual sociedad de la información, hay una creciente necesidad de acceso a servicios de comunicación de datos de forma ubicua, móvil y a velocidades cada vez más altas. Esta demanda de la sociedad, junto con los intereses de operadores, fabricantes y organismos de estandarización, ha motivado el desarrollo de la cuarta generación de comunicaciones móviles (4G) y su evolución hacia la quinta generación (5G). Este desarrollo ha requerido una revolución en la interfaz radio de los sistemas de comunicaciones móviles y, consecuentemente, ha modificado en gran medida sus capacidades y la forma en la que se gestionan sus recursos. Este es el caso de la tecnología conocida como Long Term Evolution (LTE) y su versión 4G llamada LTE-Advanced.

En concreto, esta Tesis Doctoral aborda el modelado, análisis de la gestión de recursos radio y evaluación de prestaciones del enlace descendente de LTE y LTE-Advanced, donde, de entre las características de LTE-Advanced, se ha puesto el foco de atención en la transmisión multipunto coordinada (CoMP).

La Tesis proporciona, en primer lugar, una descripción detallada de las principales características de LTE y LTE-Advanced. La gran complejidad del sistema descrito, con múltiples elementos interrelacionados, ha motivado que la metodología de estudio se haya basado en simulación mediante ordenador. La Tesis realiza una descripción detallada de la metodología de simulación y del modelado del sistema que esta metodología requiere, incluyendo algunas aportaciones del autor en este campo. De entre éstas, destaca la provisión de resultados de simulación de nivel de enlace que se usaron en el proyecto europeo WINNER+ para la evaluación de LTE.

En lo concerniente al análisis de la gestión de recursos radio en LTE, en primer lugar, se explican los fundamentos de la adaptación al enlace y el *scheduling*, describiendo los algoritmos más utilizados. En relación con el *scheduling*, se realiza un estudio del concepto de *proportional fairness* y de la implementación subóptima típicamente usada en LTE para maximizar esta métrica. Este estudio ha dado como resultado una serie de ideas que se han plasmado en una modificación de la implementación típica y que han demostrado ser capaces de aumentar la *proportional fairness* en la asignación de recursos con un bajo incremento de complejidad. Además, el análisis de la adaptación al enlace ha revelado el problema del efecto de "luz de flash" consistente en la alta variabilidad de la interferencia debida a rápidos cambios en las decisiones del scheduler. La Tesis demuestra que se pueden mejorar las prestaciones del sistema estabilizando dichas decisiones mediante una implementación concreta.

El bloque de análisis de la gestión de recursos, se completa con el estudio de la capacidad de LTE-Advanced para coordinar las transmisiones entre estaciones base, es decir, CoMP. El desarrollo hacia la 5G ha puesto de actualidad técnicas estudiadas en el 3GPP en la primera fase de evolución de CoMP, en las que la coordinación es muy rápida. La propuesta de CoMP estudiada en esta Tesis es una solución de este tipo, con coordinación de *scheduling* y conformación de haz (CS/CB), que tiene en cuenta suposiciones reales y robustas en cuanto al conocimiento que los puntos coordinados tienen de los canales radio de los usuarios servidos. La Tesis propone esta solución por su sencillez y capacidad de mejorar la eficiencia de los sistemas de comunicaciones móviles, tanto en cobertura de velocidades altas de transmisión como en capacidad, aun teniendo un conocimiento incompleto del canal, en contraposición a otros sistemas de transmisión conjunta cuya complejidad es mucho mayor y pueden verse muy perjudicados cuando no tienen un conocimiento muy completo del canal radio.

En cuanto a la evaluación de LTE y LTE-Advanced, ésta se realiza en dos tipos de escenarios diferentes. Por un lado, los escenarios definidos por la Unión Internacional de Telecomunicaciones, por medio de la ITU-R, dentro del proceso de evaluación de tecnologías IMT-Advanced. Dentro de este marco, se evalúa la importancia de diferentes técnicas de transmisión multiantena, incluyendo CoMP, y considerando tráfico de tipo *full-buffer*. Las conclusiones más importantes son la gran mejora de prestaciones que se puede conseguir con la multiplexación espacial de usuarios y que el esquema CoMP proporciona una discreta mejora. El segundo grupo de escenarios son los definidos por el proyecto europeo METIS para evaluación de tecnologías 5G, concretamente se han elegido un escenario de interiores con una oficina, y uno de exteriores con un estadio deportivo. En estos escenarios se utiliza un tráfico realista y se ha demostrado la utilidad de CoMP para satisfacer los requisitos de la 5G empleando anchos de banda razonables. En estos escenarios con despliegues menos uniformes, o con un número limitado de transmisores provocando la mayor parte de la interferencia, es donde esta Tesis ha encontrado la mayor utilidad de CoMP y donde promueve su uso.

Resum

En l'actual societat de la informació, hi ha una creixent necessitat d'accés a serveis de comunicació de dades de forma ubiqua, mòbil i a velocitats cada vegada més altes. Aquesta demanda de la societat, junt amb els interessos d'operadors, fabricants i organismes d'estandardització, ha motivat el desenrotllament de la quarta generació de comunicacions mòbils (4G) i la seua evolució cap a la quinta generació (5G). Aquest desenrotllament ha requerit una revolució en la interfície ràdio dels sistemes de comunicacions mòbils i, conseqüentment, ha modificat en gran manera les seues capacitats i la forma en què es gestionen els seus recursos. Aquest és el cas de la tecnologia coneguda com a Long Term Evolution (LTE) i la seua versió 4G anomenada LTE-Advanced.

En concret, aquesta Tesi Doctoral aborda el modelatge, anàlisi de la gestió de recursos ràdio i avaluació de prestacions de l'enllaç descendent de LTE i LTE-Advanced, on, d'entre les característiques de LTE-Advanced, s'ha posat el centre d'atenció en la transmissió multipunt coordinada (CoMP).

La Tesi proporciona, en primer lloc, una descripció detallada de les principals característiques de LTE i LTE-Advanced. La gran complexitat del sistema descrit, amb múltiples elements interrelacionats, ha motivat que la metodologia d'estudi s'haja basat en simulació per mitjà d'ordinador. La Tesi realitza una descripció detallada de la metodologia de simulació i del modelatge del sistema que aquesta metodologia requereix, incloent-hi algunes aportacions de l'autor en aquest camp. D'entre aquestes, destaca la provisió de resultats de simulació de nivell d'enllaç que es van usar en el projecte europeu WINNER+ per a l'avaluació de LTE.

Pel que fa a l'anàlisi de la gestió de recursos ràdio en LTE, en primer lloc, s'expliquen els fonaments de l'adaptació a l'enllaç i el scheduling, descrivint els algoritmes més utilitzats. En relació amb el scheduling, es realitza un estudi del concepte de proportional fairness i de la implementació subòptima típicament usada en LTE per a maximitzar aquesta mètrica. L'estudi ha donat com a resultat una sèrie d'idees que s'han plasmat en una modificació de la implementació típica i que han demostrat ser capaços d'augmentar la proportional fairness en l'assignació de recursos amb un baix increment de complexitat. A més, l'anàlisi de l'adaptació a l'enllaç ha desvetllat el problema de l'efecte de "llum de flaix" consistent en la alta variabilitat de la interferència deguda a una ràpida variació de les decisions del scheduler. La Tesi demostra que es poden millorar les prestacions del sistema estabilitzant les decisions del scheduler mitjançant una implementació concreta.

El bloc d'anàlisi de la gestió de recursos d'aquesta Tesi, es completa amb l'estudi de la capacitat de LTE-Advanced per a coordinar les transmissions entre estacions base, és a dir, CoMP. El desenrotllament cap a la 5G ha posat d'actualitat tècniques estudiades en el 3GPP en la primera fase d'evolució de CoMP, en les que la coordinació és molt ràpida. La proposta de CoMP estudiada en aquesta Tesi és una solució d'aquest tipus, amb coordinació de scheduling i conformació de feix (CS/CB), que té en compte suposicions reals i robustes quant al coneixement que els punts coordinats tenen dels canals ràdio dels usuaris servits. La Tesi proposa aquesta solució per la seua senzillesa i capacitat de millorar l'eficiència dels sistemes de comunicacions mòbils, tant en cobertura de velocitats altes de transmissió com en capacitat, tot i tindre un coneixement incomplet del canal, en contraposició a altres sistemes de transmissió conjunta la complexitat del qual és molt major i poden veure's molt perjudicats quan no tenen un coneixement molt complet del canal ràdio.

Quant a l'avaluació de LTE i LTE-Advanced, aquesta es realitza en dos tipus d'escenaris diferents. D'una banda, els escenaris definits per la Unió Internacional de Telecomunicacions, per mitjà de la ITU-R, dins del procés d'avaluació de tecnologies IMT-Advanced. Dins d'aquest marc, s'avalua la importància de diferents tècniques de transmissió multi-antena, incloent-hi CoMP, i considerant tràfic de tipus full-buffer. Les conclusions més importants són la gran millora de prestacions que es pot aconseguir amb la multiplexació espacial d'usuaris i que l'esquema CoMP proporciona una discreta millora. El segon grup d'escenaris són els definits pel projecte europeu METIS per a l'avaluació de tecnologies 5G, concretament s'han triat un escenari d'interiors amb una oficina, i un d'exteriors amb un estadi esportiu. En aquests escenaris s'utilitza un tràfic realista i s'ha demostrat la utilitat de CoMP per a satisfer els requisits de la 5G emprant amplades de banda raonables. En aquests escenaris amb desplegaments menys uniformes, o amb un nombre limitat de transmissors provocant la major part de la interferència, és on aquesta Tesi ha trobat la utilitat més gran de CoMP i on promou el seu ús.

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David Martín-Sacristán

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Acronyms

3D Three Dimensions

3G Third Generation

3GPP Third Generation Partnership Project

4G Fourth Generation

5G Fifth Generation

ABS Almost Blank Subframe

ACK Positive Acknowledgement

AM Acknowledged Mode

AoA Angle of Arrival

AoD Angle of Departure

AP Access Point

ARQ Automatic Repeat-reQuest

AWGN Additive White Gaussian Noise

BD Block Diagonalization

BER Bit Error Rate

BLER Block Error Rate

BS Base Station

CA-BF Collision Avoidance BeamForming

CB	Codebook Based
CBR	Constant Bit Rate
CC	Correlated Co-polarized
CDF	Cumulative Distribution Function
CEUSE	Cell-Edge User Spectral Efficiency
CIPE	Central Image Processing Entity
CMA-PFS	Coupled and MCS constraint Aware PFS
CoMP	Coordinated Multi-Point
CP	Cyclic Prefix
CQI	Channel Quality Indicator
CRC	Cyclic Redundancy Check
CS	Coordinated Scheduling
CS/CB	Coordinated Scheduling and Beamforming
CSE	Cell Spectral Efficiency
CSI	Channel State Information
CSI-IM	Channel State Information Interference Measurement
CSI-RS	Channel State Information Reference Signals
CSU	Cell Scheduling Utility
D2D	Device-to-Device
DFT	Discrete Fourier Transform
DL	Downlink
DL-SCH	Downlink Shared Channel
DMA-PFS	Decoupled and MCS constraint Aware PFS
DRA	Dynamic Resource Allocation
DS	Delay Spread

D-PFS	Decoupled PFS
E2E	End-to-End
EESM	Exponential Effective SINR Mapping (ESM)
eICIC	enhanced Inter-Cell Interference Coordination (ICIC)
eNodeB	evolved Node B
EPC	Evolved Packet Core
EPS	Evolved Packet System
ESM	Effective SINR Mapping
E-UTRA	Evolved UMTS Terrestrial Radio Access
E-UTRAN	Evolved UMTS Terrestrial Radio Access Network
FBMC	Filtered Bank Multi-Carrier
FDD	Frequency Division Duplexing
FDMA	Frequency Division Multiple Access
FDS	Frequency Domain Scheduling
FEC	Forward Error Correction
feICIC	further enhanced ICIC
FFT	Fast Fourier Transform
FTP	File Transfer Protocol
HARQ	Hybrid ARQ
HDR	High Data Rate
HSPA	High Speed Packet Access
ICIC	Inter-Cell Interference Coordination
IEEE	Institute of Electrical and Electronic Engineers
ILLA	Inner Loop Link Adaptation
IMT	International Mobile Telecommunication

IMT-2000 International Mobile Telecommunication (IMT) 2000

IMT-Advanced International Mobile Telecommunications Advanced

InH Indoor hotspot

IP Internet Protocol

IRC Interference Rejection Combining

IS-PFS Interference Stabilization Proportional Fair Scheduler

ISI Inter Symbol Interference

ITU International Telecommunication Union

ITU-R International Telecommunication Union (ITU) Radiocommunication Sector

JP Joint Processing

KPI Key Performance Indicator

LA Link Adaptation

LoS Line of Sight

LTE Long Term Evolution

LTE-Advanced Long Term Evolution (LTE) Advanced

LUT Look-Up Table

L2S Link to System

MAC Medium Access Control

MBMS Multimedia Broadcast and Multicast Service

MBSFN MBMS over Single Frequency Networks

MCS Modulation and Coding Scheme

METIS Mobile and wireless communications Enablers for Twenty-twenty Information Society

MIESM Mutual Information Effective Signal to Interference plus Noise Ratio (SINR) Mapping

MIMO	Multiple-Input Multiple-Output
ML	Maximum Likelihood
MME	Mobility Management Entity
MMSE	Minimum Mean Square Error
MRC	Maximum Ratio Combining
MS	Mobile Station
MU-MIMO	Multi-User MIMO
NACK	Negative Acknowledgement
NGMN	Next Generation Mobile Networks
NLoS	Non Line of Sight
OFDM	Orthogonal Frequency Division Multiplexing
OFDMA	Orthogonal Frequency Division Multiple Access
OLLA	Outer-Loop Link Adaptation
OtoI	Outdoor to Indoor
PDCCH	Physical Downlink Control Channel
PDCP	Packet Data Convergence Protocol
PDN	Packet Data Network
PDN-GW	Packet Data Network Gateway
PDSCH	Physical Downlink Shared Channel
PDU	Protocol Data Unit
PFS	Proportional Fair Scheduling
PHY	Physical
PMI	Precoding Matrix Indicator
PUCCH	Physical Uplink Control Channel
PUSCH	Physical Uplink Shared Channel

QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
RAN	Radio Access Network
RAT	Radio Access Technology
RB	Resource Block
RE	Resource Element
RF	Radio Frequency
RI	Rank Indicator
RIT	Radio Interface Technology
RLC	Radio Link Control
RMa	Rural Macro-cell
RRH	Remote Radio Head
RRM	Radio Resource Management
RRC	Radio Resource Control
RTT	Round Trip Time
SAE	System Architecture Evolution
SC-FDMA	Single Carrier-Frequency Division Multiple Access
SDU	Service Data Unit
SF	Shadow Fading
SFN	System Frame Number
S-GW	Serving Gateway
SIMO	Single Input Multiple Output
SINR	Signal to Interference plus Noise Ratio
SISO	Single Input Single Output

SLNR Signal to Leakage plus Noise Ratio

SMa Suburban Macro-cell

SNR Signal to Noise Ratio

SON Self-Optimized Network

SRIT Set of Radio Interface Technologies

SU-MIMO Single User MIMO

TB Transport Block

TC1 Test Case 1

TC4 Test Case 4

TCP Transmission Control Protocol

TDD Time Division Duplexing

TDS Time Domain Scheduling

TM Transparent Mode

TTI Transmission Time Interval

UC Uncorrelated Co-polarized

UE User Equipment

UF-OFDM Universal Filtered OFDM

UL Uplink

UM Un-acknowledged Mode

UMa Urban Macro-cell

UMi Urban Micro-cell

UMTS Universal Mobile Telecommunications System

USE User Spectral Efficiency

VoIP Voice over IP

WCDMA Wideband Code Division Multiple Access

WINNER+ Wireless World Initiative New Radio +

WiMAX Worldwide Interoperability for Microwave Access

ZF Zero Forcing

Notation

This section describes the notation used in this Thesis.

$\mathbf{A}$ (boldface capital letter) and $\mathbf{A}_k$ represent matrices.
$[\mathbf{A}]_{ij}$ is the element of matrix $\mathbf{A}$ in row i and column j.
$[\mathbf{A}]_{i:}$ represents the i-th row of matrix $\mathbf{A}$.
$[\mathbf{A}]_{:j}$ represents the j-th column of matrix $\mathbf{A}$.
$\mathbf{A}^H$ denotes the conjugate transpose or Hermitian transpose of $\mathbf{A}$.
$\mathbf{a}$ (boldface lower-case letter) and $\mathbf{a}_k$ represent vectors.
$[\mathbf{a}]_i$ represents the i-th element of vector $\mathbf{a}$.
a and A (not boldface italic letters) represent scalars.
$\mathcal{A}$ (calligraphic letter) represents a set of numbers.
$|\mathcal{A}|$ denotes the number of elements in the set $\mathcal{A}$.

Chapter 1

Introduction

This Thesis presents the results of a research activity focused on the field of the radio resource management of a specific cellular technology known as Long Term Evolution (LTE) and its evolution, LTE Advanced (LTE-Advanced). The present chapter contains an introduction to the Thesis that explains the motivation of the work done and provides a historical and technological background. This background is given by means of the explanation of some concepts or by the presentation of significant references.

With this aim, this chapter has been divided into the next sections:

- Section 1.1 presents the historical and technological context in which this Thesis has been developed. The main features of LTE and LTE-Advanced standards are briefly presented.

- Section 1.2 analyses the state of the research conducted in the field covered by this Thesis up to the start of the Thesis work. Concurrent research activities are also commented.

- Section 1.3 identifies the main problems treated in the research community concerning the field covered in this Thesis, the research hypothesis and research objectives of the Thesis.

- Section 1.4 outlines the structure of the Thesis summarizing the main contents of each chapter.

- Section 1.5 lists the publications related with this Thesis including journal and conference papers, reports and book chapters.

1.1 Background

This section presents the context in which the Thesis has been developed. This context includes the evolution process of the mobile and wireless communication sector and the main technical features of the developed standards.

1.1.1 Historical background: from 3G to 5G

The International Telecommunication Union (ITU) is the international entity recognized to define global mobile and wireless communication standards. This definition includes the specification of set of requirements in terms of transmission capacity and quality of service, in such a way that if a certain technology fulfils all these requirements it is included by the ITU in the global set of standards. This inclusion firstly endorses technologies and motivates operators to invest in them, but furthermore, it allows these technologies to make use of the frequency bands specially designated for the global standard, what entails a great motivation for mobile operators to increase their offered services and transmission capacity.

At the end of 1999, the Third Generation (3G) of mobile and wireless communication systems (International Mobile Telecommunication (IMT) 2000 (IMT-2000) in the ITU terminology) was completely defined by the ITU. At that moment, forecasts expected an increasing demand of wireless communications and higher data rate requirements. This motivated that a work to enhance IMT-2000 and to develop systems beyond IMT-2000 was immediately started. Initial activity was conducted by the ITU Radiocommunication Sector (ITU-R) Working Party 8F under a work item known as Question ITU-R 229-1/8 "Future development of IMT-2000 and systems beyond IMT-2000". This work item produced a number of recommendations and reports to guide the standardization process of a new global standard, referred to as"beyond IMT-2000" temporarily and was the seminal work that paved the way towards the Fourth Generation (4G) of mobile and wireless communication systems. At the end of 2002, the vision on beyond IMT-2000 was completed [1]. Afterwards, in 2005, the name International Mobile Telecommunications Advanced (IMT-Advanced) was made official [2] substituting the temporary term. Future services and market analysis, future technology assumptions, and spectrum requirements were also covered in Working Party 8F recommendations and reports, most of them finalized in 2006. All those documents were important inputs for the Radiocommunication Assembly (RA-07) and the World Radiocommunication Conference (WRC-07) conducted by the ITU in 2007. Since RA-07 celebration, work on IMT was addressed to the ITU-R Working Party 5D. Additionally, the original question was revised and called Question

ITU-R 229-2/5 “Future development of the terrestrial component of IMT”. In WRC-07, the increasing market demand and its enormous economic benefits, together with the new challenges that come with the requirements in higher spectral efficiency and services aggregation, raised the need to allocate new frequency channels to mobile communications systems.

After this previous work, the pace towards IMT-Advanced had a milestone in March 2008, when a Circular Letter was sent out by the ITU-R to invite submissions of IMT-Advanced technology proposals [3]. This action was the start of an ITU-R schedule spanned over the 2008-2011 time frame as shown in the lower part of Figure 1.1 (based on [4]). The radio interface development process was covered in several steps, the first one represented by the issuance of the Circular Letter (step 1), after which step 2 coped with the development of candidate Radio Interface Technologies (RITs) and Set of Radio Interface Technologies (SRIT). In December 2008, a set of reports with overall requirements, evaluation criteria and submission templates [5], requirements related to technical system performance [6] and guidelines for evaluation [7] were released. Step 3 represented the submission/reception of the RIT and SRIT proposals (and acknowledgement of receipt) to Working Party 5D. Step 3 was completed in October 2009.

In the next phase, step 4, the evaluation of candidate RITs or SRITs by evaluation groups was carried out, ending in June 2010. Steps 5, 6, and 7 referred to the review and coordination of outside evaluation activities, the review to assess compliance with minimum requirements and, finally, the consideration of evaluation results, consensus building and decision. These steps were concluded in October 2010. The outcome of steps 4–7 is recorded in [8]. Step 8 referred to the development of radio interface recommendation(s). The outcome of this phase was included in [9], first drafted in October 2011. Final approval came in January 2012, during a Radiocommunication Assembly (RA-12).

Several external evaluation groups were constituted to perform a parallel and independent evaluation of proposed candidates to IMT-Advanced. One of these groups was the European project Wireless World Initiative New Radio + (WINNER+). This project was a consortium whose aim was to develop, optimize and evaluate technologies compliant with IMT-Advanced. It was founded on the successful experience of previous projects WINNER and WINNER II, which had a great impact on the definition and evaluation of IMT-Advanced. WINNER+ started in April 2008 and finalized its work in June 2010.

In parallel to the ITU process to define 4G, the organisms in charge of the most prominent mobile and wireless standards have been evolving their technologies. In 2004, Third Generation Partnership Project (3GPP) partners decided that they should develop a 3G long-term evolution in order to be competitive for more than a decade from that date. This evolution should provide reduced latency, higher user data rate, improved system capacity and coverage, and reduced cost for the operator. Besides, it should support a flexible bandwidth operation and should be packet-optimized. These requirements could be partially met if Universal Mobile Telecommunications System (UMTS) was upgraded including additional features. In fact, UMTS Release 99 was evolved to High Speed Packet Access (HSPA) and HSPA+, which are technologies based on Wideband Code Division Multiple Access (WCDMA), following this approach. But, as far back as in 2004, the 3GPP envisaged that a more disruptive approach could be necessary. LTE is the name given to the new standard developed by the 3GPP to cope with the increasing throughput requirements of the market. The 3GPP Radio Access Network (RAN) working groups started LTE standardization in December 2004 with a feasibility study for an evolved Evolved UMTS Terrestrial Radio Access (E-UTRA) and Evolved UMTS Terrestrial Radio Access Network (E-UTRAN) [10]. This is known as the study item phase. In December 2007, all LTE functional specifications were finished. In 2008, the 3GPP working groups were running to complete all protocol and performance specifications, being these tasks done in December 2008 hence ending Release 8. This process is illustrated in the upper part of Figure 1.1.

The IMT-Advanced definition by the ITU-R produced an effect on the 3GPP ongoing activities. In 2008, the 3GPP held two 3GPP IMT-Advanced workshops. The goal of these workshops was to investigate which were the main changes that could be brought forward to enhance the E-UTRAN interface as well as the E-UTRA in the context of IMT-Advanced. This process led to a new standard called LTE-Advanced. The standardization and evaluation process of LTE-Advanced was very similar to the one of LTE. First, a study item description on Further Advancements for E-UTRA was approved [11]. The study item comprised technology components to be considered for the evolution of E-UTRA, e.g. to fulfil the requirements of IMT-Advanced. The study item was initialized in March 2008 and was almost finalized in August 2009. Outcomes of the study item supported the 3GPP submission of "LTE Release 10 & beyond (LTE-Advanced)" to the ITU-R as a candidate technology for the IMT-Advanced family of standards in September 2009. LTE Release 10 and beyond (LTE-Advanced) was accepted as a 4G technology by the ITU-R Working Party 5D in October 2010. The first version of the Release 10 specifications was frozen in March 2010 while stable protocols were available in June 2011.

Work on LTE-Advanced did not stop in the 3GPP with Release 10. Afterwards, Releases 11 and 12 have been completed, while Releases 13 and 14 are ongoing. Concerning these latter versions of the standard, the most relevant aspect for this Thesis is the first explicit inclusion of Coordinated Mul-

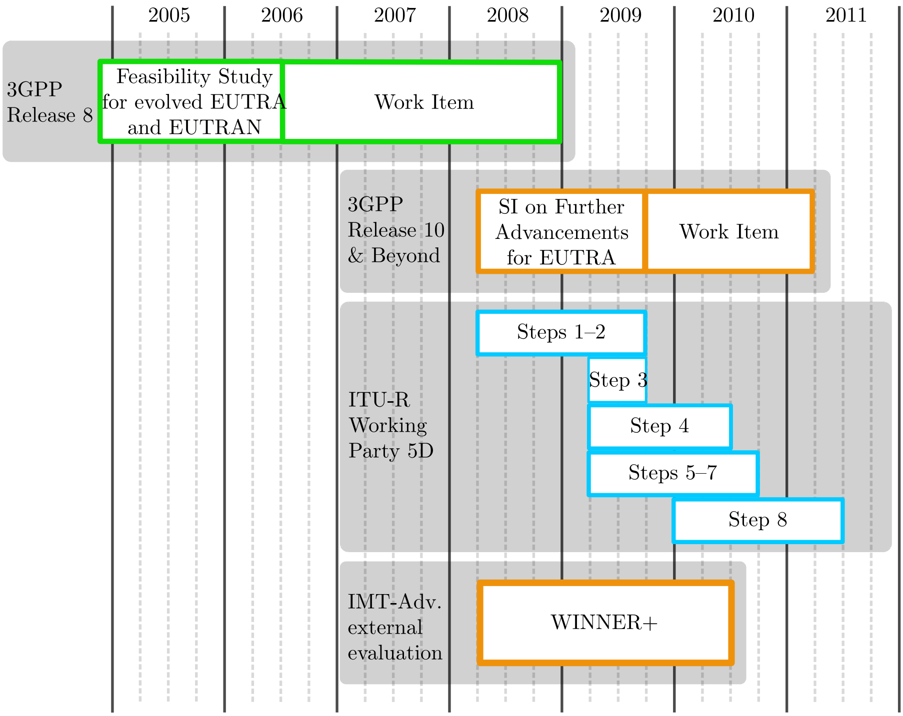


Figure 1.1: 4G time line

ti-Point (CoMP) transmission/reception in the specification and its following development. Next, the work developed in the 3GPP in relation to CoMP is detailed.

Although CoMP was a feature studied in LTE-Advanced study item, it was not finally included in the Release 10 specifications due to the huge specification work, and it was postponed to Release 11. In fact, in March 2010, a study item on CoMP Operation for LTE started. But again, it was considered not a priority and work started in fact in December 2010. In comparison with the previous studies, the main change was the consideration of deployment types different to the homogeneous macro networks in which the performance evaluation of LTE-Advanced especially focused. Deployment types with distributed Remote Radio Heads (RRHs) and heterogeneous networks were considered in the study item. A technical report [12] presents the outcomes of the study item. This milestone is represented in Figure 1.2 as M1. The work mainly focused on CoMP with high capacity and low latency communication between transmission points. However, a preliminary evaluation of constraints from lower capacity/higher latency communication between transmission points on performance gain of CoMP was conducted. As a conclusion, the study item recommended a way forward on actual design principle with its related scenarios for the high capacity/low latency inter-point communication. The study item finalized in September 2011 and, immediately, a work item started to modify the specifications according to the study item findings. The work focused on providing the means to implement CoMP techniques among transmission points of the same evolved Node B (eNodeB) or among points of different eNodeB but assuming an ideal backhaul (high capacity, low latency). With this aim, means to estimate appropriately the channel quality between a user and multiple transmission points were included in the standard. The core part of the downlink specifications (not accounting for performance requirements nor conformance tests) finalized in December 2012, while the rest of the work ended in December 2013.

In parallel to the Release 11 specification, a Release 12 study item on “CoMP for LTE with Non-Ideal Backhaul” started in June 2013. The justification of this study item was that CoMP in Rel-11 did not address the support of CoMP with multiple eNodeBs with non-ideal backhaul. Due to this limitation, the operators having non-ideal backhaul may not be able to take performance benefit from CoMP operation. The study considered realistic backhaul delays. Moreover, estimation errors, downlink overhead, complexity, feedback overhead, backward compatibility and practical User Equipment (UE) implementations were considered. CoMP between macro eNodeBs, between macro eNodeBs and small cells, and between small cells was also studied. In December 2013, the work was completed and gathered in a technical report [13] shown as milestone M2 in Figure 1.2. Following the study item, a work item on enhanced inter-eNodeB signalling support for Multi-Cell Coordination in LTE started. Based on the identification of the cases in which CoMP can provide performance gains done by the study item, enhancement on the network interface and signalling messages was carried out by the work item. The specification should allow implementing both centralized and distributed coordination, focusing primarily on macro-pico heterogeneous networks but also considering macro-macro homogeneous networks. As a result, existing messages in an interface between eNodeBs were extended to allow the exchange of information useful for CoMP such as hypotheses on activity or muting of specific transmission points, and benefit of such transmission/muting. Besides, the measurement of wideband power received from multiple transmission points was allowed. Release 12 version of technical specification [14] includes explicit comments about inter-eNodeB CoMP and the information that can be exchanged between nodes. The Release 12 CoMP-related work item finalized in December 2014.

In parallel to the Release 12 work item, in June 2014 another work item started in the Release 13, called Enhanced Signalling for Inter-eNodeB CoMP. The purpose of this work item is quite similar to the Release 12 one: extend the interface between nodes to allow the exchange of useful information. This work item is being finalized at the moment of writing this Thesis.

In order to complete the historical framework since the beginning of the 4G designing until today, it is necessary to present the first activities headed to the definition of a new generation of mobile and wireless communication systems: the Fifth Generation (5G) mobile. One of the most prominent international consortia dealing with the definition and design of 5G was the Mobile and wireless communications Enablers for Twenty-twenty Information Society (METIS) consortium. METIS was co-funded by the European Commission as an Integrated Project under the Seventh Framework Programme for research and development. The project objective was to lay the foundation for a future mobile and wireless communications system for 2020 and beyond. The consortium has had two phases. A first one started in November 2012 and finalized in April 2015. The second one, METIS-II, started in July 2015 and will finalize in June 2017. The METIS-II activity builds on the successful METIS project and will develop the overall 5G radio access network design and provide the technical enablers needed for an efficient integration and use of the various 5G technologies and components currently developed. The most relevant aspect of METIS for this Thesis is that CoMP has been considered as an integral part of the new 5G system, and its impact on system performance has been considered in multiple scenarios. New deployment and evaluation scenarios defined by METIS are being a de-facto standard for the scientific community with enormous relevance, and consequently, it is very interesting to elucidate the importance of CoMP in these scenarios.

1.1.2 LTE standard

The most important technologies included in the LTE Radio Access Technology (RAT) are Orthogonal Frequency Division Multiplexing (OFDM), multi-dimensional (time, frequency) Dynamic Resource Allocation (DRA) and Link Adaptation (LA), Multiple-Input Multiple-Output (MIMO) transmission, turbo coding and Hybrid ARQ (HARQ) with soft combining. These technologies are shortly explained in the following paragraphs.

- OFDM is a multi-carrier transmission technique with a relatively large number of subcarriers that offers a lot of advantages. First of all, by

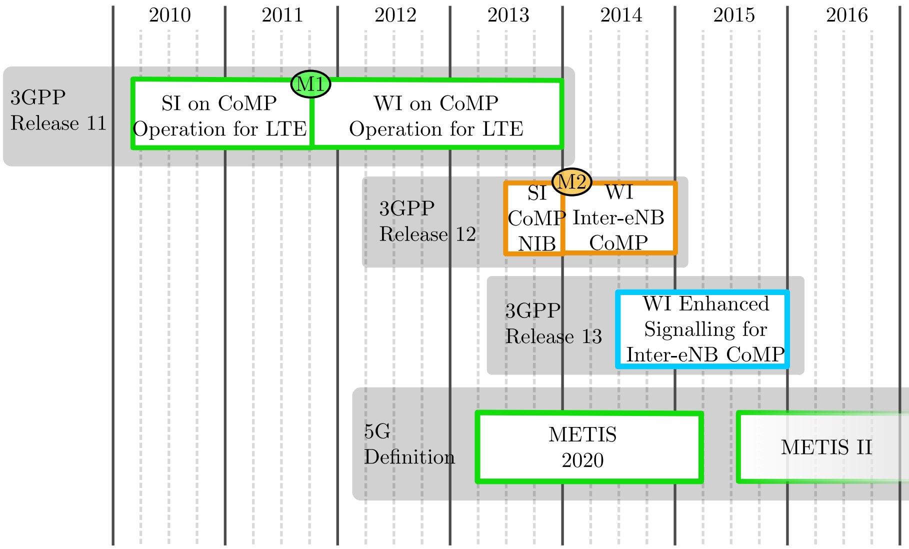


Figure 1.2: Time line with CoMP evolution in 3GPP and first 5G activities

using a multiple carrier transmission technique, the symbol time can be made substantially longer than the channel delay spread, which reduces significantly or even removes the Inter Symbol Interference (ISI). In other words, OFDM provides a high robustness against frequency selective fading. Secondly, due to its particular structure, OFDM allows for low-complexity implementation by means of Fast Fourier Transform (FFT) processing. Thirdly, the access to the frequency domain in Orthogonal Frequency Division Multiple Access (OFDMA) implies a high degree of freedom to the scheduler. Finally, it offers spectrum flexibility that facilitates a smooth evolution from already existing RATs to LTE. In the uplink, Single Carrier-Frequency Division Multiple Access (SC-FDMA) is used rather than OFDM. SC-FDMA is also known as Discrete Fourier Transform (DFT)-spread OFDM modulation. Basically, SC-FDMA is identical to OFDM unless an initial FFT is applied before the OFDM modulation. The objective of such modification is to reduce the peak to average power ratio, thus decreasing the power consumption in the user terminals.

- Multidimensional DRA and LA. In LTE, both uplink and downlink transmission schemes can assign smaller, non-overlapping frequency bands to the different users, offering Frequency Division Multiple Access (FDMA). This assignment can be dynamically adjusted in time and is referred to as scheduling. Accordingly, the LTE resources can be represented as a time-frequency grid. The minor element of this grid is called resource element and consists of one subcarrier during an OFDM symbol. However, the minor LTE resource allocation unit is the resource block that consists of 12 subcarriers during one slot. LA is closely related to scheduling and deals with how to set the transmission parameters of a radio link to handle variations of the radio-link quality. In LTE, LA is achieved through adaptive channel coding and adaptive modulation. Specifically, in LTE available modulations are Quadrature Phase Shift Keying (QPSK), 16-Quadrature Amplitude Modulation (QAM) and 64-QAM.

- MIMO transmission is one of the most important means to achieve the high data rate objectives for LTE. In LTE downlink, it is supported one, two or four transmit antennas in the eNodeB and one, two or four receive antennas in the UE. Multiple antennas can be used in different ways: to obtain additional transmit/receive diversity or to get spatial multiplexing (increasing the data rate by creating several parallel channels if conditions allow to). Nevertheless, in LTE uplink although one, two or four receive antennas are allowed in the eNodeB, only one transmitting antenna is allowed in the UE. Therefore, multiple antennas can be only used to obtain receive diversity.

- Turbo Coding is the channel coding scheme of user data channels. In case of the downlink user data channel, a turbo encoder with rate 1/3 is used followed by a rate matching to adapt the coding rate to the desired level. In each subframe of 1 ms, one or two (with multi-codeword MIMO) codewords can be coded and transmitted.

- HARQ is a technique that deals with the retransmission of data in case of errors through the combination of Forward Error Correction (FEC) and Automatic Repeat-reQuest (ARQ). In an ARQ scheme, the receiver uses an error-detecting code to check if the received packet contains errors or not. The transmitter is informed by a Negative Acknowledgement (NACK) or Positive Acknowledgement (ACK), respectively. In case of a NACK, the packet is retransmitted. In LTE, HARQ scheme is built around a Cyclic Redundancy Check (CRC) code for error detection and a turbo code for error correction. Furthermore, HARQ comes with soft combining, where the erroneously received packet is stored in a buffer and later combined with the retransmission(s) to obtain a single packet that is more reliable than its constituents. Besides, full incremental redundancy is applied, which means that the retransmitted packets are typically not identical with the first transmission but carry complementary information.

1.1.3 LTE-Advanced standard

The main technical features of LTE-Advanced are the following:

- Support of Wider Bandwidth. Carrier aggregation, where two or more component carriers, each with a bandwidth of up to 20 MHz, are aggregated, is considered for LTE-Advanced in order to support downlink transmission bandwidths larger than 20 MHz and up to 100 MHz.

- Extended Multi-Antenna Configurations. Extension of LTE downlink spatial multiplexing is considered. LTE-Advanced supports spatial multiplexing of up to eight layers for the downlink direction and up to four layers for the uplink direction. Enhanced Multi-User MIMO (MU-MIMO) transmission is supported in LTE-Advanced.

- CoMP. This technique is considered for LTE-Advanced as a tool to improve the coverage of high data rates, the cell-edge throughput and/or to increase system throughput. Downlink CoMP transmission implies dynamic coordination among multiple geographically-separated transmission points. This coordination may involve coordination of scheduling, of beamforming, or even transmission of the same information from the coordinated points. Initially, the focus of CoMP was on intra-site schemes but in last Releases inter-site schemes support has been developed. Uplink CoMP reception can involve joint reception of the transmitted signal at multiple reception points and/or coordinated scheduling decisions among cells to control interference. Uplink CoMP reception has very limited impact on the specifications.

- Relaying Functionality. Relaying is considered for LTE-Advanced as a tool to improve the coverage of high data rates, group mobility, temporary network deployment, the cell-edge throughput, and/or to extend coverage to new areas.

1.2 State of the art analysis

This section presents an analysis of the state of the art in the field of study of this Thesis based on a deep literature review. This analysis is divided into

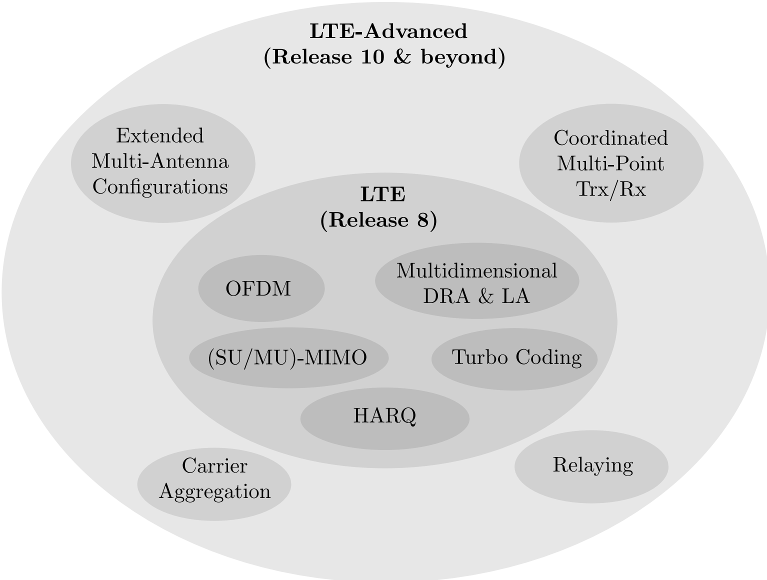


Figure 1.3: LTE & LTE-Advanced main features

three parts. The first one concerns the topic of performance evaluation in LTE and LTE-Advanced. The second part deals with the most important features of Radio Resource Management (RRM) in beyond 3G systems. The third part presents the topic of cooperation/coordination in wireless networks.

1.2.1 Performance evaluation and system modelling

Different methodologies can be followed to evaluate the performance of a wireless system or any of its parts. When the system is developed and deployed, performance can be evaluated through real measurements conducted over the operating system. Nevertheless, if the system is not fully deployed, or in order to reduce the high cost of real campaigns, other methods may be used. Analytical methods can only be used to perform simple assessments. Due to the high complexity of wireless systems, the preferred method is simulation. In order to simulate a system it is necessary to model it properly. A model implies a certain degree of abstraction and simplification. The tolerable level of simplification depends on the tolerable level of discrepancy between the performance of a simulated system and the performance of a real one. Besides a system model, it is necessary to define the scenario to test the system, since its performance depends on the scenario of deployment. Additionally, it is necessary to define the tests to be carried out and the performance indicators to derive.

Next, references from the main forums containing evaluation methodologies and performance results concerning LTE are presented. Afterwards, open questions related to system modelling are presented.

Conducted evaluations

Performance evaluation of LTE has been conducted by many different actors. The first group of actors is that composed by the members of the 3GPP, that is, the developers of LTE. The second group of actors is that composed by evaluation groups helping ITU-R in making a decision about the LTE fulfilment of IMT-Advanced requirements. Additionally, independent research groups have studied LTE performance.

A. Performance evaluation of LTE in the 3GPP

In December 2004 the study item description on E-UTRA and E-UTRAN was approved. The first step of this process was to develop a document that provided guidance and collected requirements that an E-UTRA and E-UTRAN system should meet [15]. It is important to take into account that some requirement definitions involve explicitly a specific evaluation methodology including the reference scenario. In the third quarter of 2006 it was concluded the report describing the physical layer basis of LTE [16]. The purpose of this report was to help the 3GPP define and describe the potential physical layer evolution under consideration and compare the benefits of each proposed technique, along with the complexity evaluation of each technique. This document was intended to gather all information in order to compare the solutions and gains versus complexity, and draw a conclusion on the way forward. In this document, the common framework for LTE performance evaluation was established. The framework is complemented with [17], the physical layer framework for performance evaluation.

In June 2007, the first performance evaluation results were included in the technical report for the study item about E-UTRA and E-UTRAN [18]. In this document, performance results of the 3GPP partners were averaged and compared with the requirements of [15]. Detailed results of each 3GPP partner can be found in [19] and [20]. These documents are quite good starting points for the performance evaluation point of view since they give valid results for calibration. In parallel to the standardization process, the 3GPP partners developed a hard task of evaluation.

The assessments focused on specific parts of the system as well as on the overall performance. Besides, in the final phase of LTE standardization process, once the technical specifications were determined, a deepest performance evaluation was required to determine a set of realistic minimum performance requirements that any LTE compliant equipment should met. These requirements are collected in [21] and [22].

B. Performance evaluation of LTE-Advanced in the 3GPP

The standardization process and evaluation of LTE-Advanced has been very similar to the one of LTE. First, a study item description on Further Advancements for E-UTRA was approved [11]. Requirements for further advancements for E-UTRA are gathered in [23], whereas physical layers aspects of the technology components studied during the study item are included in [24]. The 3GPP Technical Report [25] gathers all technical outcomes of the study item, and draws a conclusion on the way forward. In addition, this document includes the results of the work supporting the 3GPP submission of "LTE Release 10 & beyond (LTE-Advanced)" to the ITU-R as a candidate technology for the IMT-Advanced family of standards.

C. Performance evaluation of IMT-Advanced

After the successful definition of a global standard for the 3G of mobile and wireless networks, IMT-2000, the ITU decided the creation of a global standard for the 4G, IMT-Advanced. In this process, the ITU-R has defined the IMT-Advanced requirements [6] and the evaluation methodology that should be followed to check that candidate technologies meet these requirements [7]. Several organisms have proposed candidate technologies to IMT-Advanced. For example, the 3GPP is the proponent of LTE-Advanced while the Institute of Electrical and Electronic Engineers (IEEE) proposed an evolution of Worldwide Interoperability for Microwave Access (WiMAX) called 802.16m. Proponents have presented reports to demonstrate that their technologies fulfil IMT-Advanced requirements. The 3GPP results can be found in [25]. This report includes LTE-Advanced results and also LTE results. Moreover, IEEE has made publicly available its results and, what is more interesting for this Thesis, a document with its evaluation methodology [26]. Furthermore, independent evaluation groups were constituted to perform a parallel and independent evaluation of proposed candidates. One of these groups is the European project WINNER+, whose reports are available in [27].

The LTE (and LTE-Advanced) standard is opened at some points. Some examples of this lack of concreteness are the RRM algorithms such as admission control, LA or scheduling algorithms. Practical implementation of the receiver is also opened, including the MIMO processing, channel estimation or decoder implementation. Therefore, many of the building blocks of the system can be implemented in different ways and hence the performance of a specific system implementation can differ a lot from a system to another.

Previously cited references provide system level results of specific implementations of the system (or average performances of specific implementations, in case of several partners reporting mean results). However, the effect of each building block implementation on the system performance is not usually separated. Even in many cases the specific implementation of key building blocks is not completely defined. Thus, with these references it is possible to get an idea of what can be expected from LTE, but it is not possible to deduce to which extent it is important each part of the standard and how performance can be improved if each building block is enhanced. In order to get such knowledge, the main information sources are the particular technical reports of the 3GPP partners and the publications of experts in the field.

Concerning link level evaluation, that is, the performance of a single link between an eNodeB and a UE, performance of several specific system configurations are provided for example in [28–32]. MIMO is especially well covered in [33] and [34].

System modelling

Guidelines provided by the 3GPP and by the ITU-R are the basis to model an LTE system. Additionally, some indications were provided by the Next Generation Mobile Networks (NGMN) Alliance [35]. All those guidelines give certain freedom to the evaluators in some issues. For example, the practical implementation of the channel models (not the channel models itself), interference modelling, the channel estimation error modelling and the link to system interfaces are not specified. Thus, the evaluator can find efficient implementations that maintain an acceptable precision. Next, these opened issues are commented.

A. Channel model implementation

Several papers have dealt with the topic of channel model implementation for efficient simulation of OFDMA, such as the paper of Monghal *et al.* [36]. In this paper, the accuracy of simplified models is studied comparing the effect of these simplifications on the channel correlation properties and Signal to Interference plus Noise Ratio (SINR) error in reception. However, the effect on a specific system is not studied.

B. Interference modelling

Explicit modelling of interferences is required in the performance evaluation according to IMT-Advanced guidelines. However, in order to perform an efficient evaluation and to reduce the computational burden simplifications are needed. In [37], Skillermark *et al.* studied whether it is necessary to simulate in detail (that is, with small scale variation) every link among user equipments and base stations or not. It is demonstrated that it is enough to model in detail only the strongest interferers. Evaluation accuracy is measured comparing SINR and throughput statistics obtained for different numbers of strongest interferers modelled in detail.

C. Channel estimation modelling

In many LTE assessments it is not considered the effect of channel estimation error on the performance, see e.g. the works of Spiegel *et al.* [33], Wei *et al.* [28] and Mogensen *et al.* [29]. Others perform simulations without modelling channel estimation error and then correct the performance values using a correction factor, as proposed by NGMN in [35]. Those values are different for uplink and downlink and for different technologies. Finally, other studies use more accurate models. The most common procedure is to include a penalty in the SINR experienced in the receiver before using the link to system level mapping. In some studies the SINR penalty is fixed and in others it depends on the SINR itself, so that the higher the SINR, the lower the penalty (as proposed by NGMN in [35]). Despite this research effort, the validity of these simplified models has not been fully demonstrated not even contrasted with different estimators.

D. Link to system interfaces

The development of link to system interfaces valid for systems beyond 3G has received a lot of attention. Evolution from actual value interface used in previous systems to Effective SINR Mapping (ESM) models has resulted from such effort. In the 3GPP, it has been widely extended the use of Exponential ESM (EESM). Recently, the Mutual Information Effective SINR Mapping (MIESM) has gain importance. A good reference about this topic was published by Brueninghaus *et al.* [38]. Models based on effective SINR are determined by a characteristic function, a set of look up tables that translate any channel quality parameter to a link performance measure obtained in Additive White Gaussian Noise (AWGN), and a set of parameters ensuring the optimum fit of the model to reality. Some authors have made publicly available their set of obtained parameters. Nevertheless, these parameters correspond to specific system implementations. Therefore, these parameters are not generally applicable. If the decoder or the MIMO processing in the receiver changes, then these parameters change as well. Finally, it is worth highlighting that an important topic that has received less attention is the joint effect of time/frequency granularity of the simulation on the accuracy of the link to system interfaces predictions.

E. Dynamic simulation

Many LTE performance assessments are based on static or quasi-static simulations. With this kind of simulations it is not possible to properly perceive the effect that some dynamic processes have on system performance. Consider a scenario with high speed mobiles and rapidly changing interferences. In this scenario, the channel quality changes fast, link adaptation cannot track channel quality properly and performance degrades. In a static simulation this effect is neglected. Therefore, the system level simulator developed in this Thesis must be dynamic to allow a full and proper characterization of the system.

1.2.2 Radio Resource Management

RRM is a key element of LTE. In this Thesis the focus is on the RRM techniques related to resource allocation. Next, the main characteristics of LTE affecting RRM are highlighted.

Scheduling

The term scheduling is a synonym of multidimensional DRA. Frequency domain packet scheduling has received a lot of attention especially by Aalborg and Nokia researchers. In [39], Pokhariyal *et al.* presented a basic study. In [40] the same authors assumed a fractional load in the network under study. The work follows with [41], where Pedersen *et al.* considered limited and noisy channel feedback. Then, Wei *et al.* [42] and Kovacs *et al.* [43] analysed the implication of MIMO and, finally, all these studies were summarized by Wei *et al.* [44].

Link Adaptation

The effect of realistic channel quality indicators on the system performance is also an important topic. Kolding *et al.* [45] studied low-bandwidth channel quality indication for OFDMA frequency domain packet scheduling with a Maximum Ratio Combining (MRC) receiver. In [46], Kolehmainen *et al.* assessed different channel quality indication reporting schemes for LTE downlink. Moreover, Wei *et al.* considered MIMO adaptation in [47]. The effect of non-ideal channel feedback on dual-stream MIMO-OFDMA system performance is assessed by Kovacs *et al.* in [48]. In these papers an Outer-Loop Link Adaptation (OLLA) is used to fix the Block Error Rate (BLER) of the first transmission of a radio block to a desired value. Zhu *et al* made an interesting contribution in [49], since it is considered the update of the Channel Quality Indicator (CQI) received assuming Single User MIMO (SU-MIMO) when the used mode is MU-MIMO. System level performance of downlink MU-MIMO transmission for 3GPP LTE-Advanced is covered by Kusume *et al.* in [50]. Besides, a realistic Channel State Information (CSI) overview and performance evaluation of LTE downlink with MIMO is performed by Simonsson *et al.* in [51]. Finally, concerning link adaptation, it is worth noting that link adaptation performance with realistic CQI in MIMO scenarios where interference changes in a fast pace is a topic not sufficiently covered.

Hybrid ARQ

The HARQ topic has been well covered at link level, see e.g. the work of Beh *et al.* [52]. Nevertheless, the implications of HARQ on other processes, like scheduling, have been treated only partially, while in many studies this effect is totally neglected. In [40] (by Pokhariyal *et al.*) and related work of the authors, it is assumed that in the scheduling process priority is given to retransmissions. Nevertheless, there is a need to evaluate implications of this prioritization.

1.2.3 Cooperation/coordination in wireless networks

One of the main goals of the 4G of mobile telecommunications is to increase the spectral efficiency. Joint application of MIMO and higher order modulations can achieve such an objective, but only in high SINR regime. Then, one possibility to get the most of 4G is to reduce noise and interference as far as possible. Noise power is not the main constraint to the 4G systems performance, instead, it is said that 4G systems are limited by interference. Specifically, inter-cell interference is the main problem in 4G systems, since intra-cell interference is practically cancelled thanks to the multiplexing schemes in use (OFDMA). As a conclusion, work to improve spectral efficiency can be focused in the reduction of inter-cell interference levels experienced in networks.

In the 3GPP, it has been used a concept called interference mitigation to refer to those techniques that try to reduce the inter-cell interference. A classification of interference mitigation schemes used in the 3GPP is found in [53] and [16] among others:

- Inter-cell interference randomization. It consist in randomizing interfering signal to allow interference suppression in the receiver. Cell specific scrambling using pseudo-random scrambling is one possible technique.

- Inter-cell interference cancellation. It can be achieved through spatial suppression using multiple antennas at the user equipment or by means of detection/subtraction of inter-cell interference.

- Inter-cell interference coordination/avoidance. It implies a restriction to the resource management in a coordinated way among cells. Restrictions in a cell allow the improvement of SINR in neighbour cells.

Besides reducing interference, there is a more aggressive approach to increase SINR based on base station coordination. This approach turns interference into useful signal through joint processing of transmitted data. That is, coordination of multi-cell multi-user MIMO systems is used. Thus, the problem of overcoming interference in MIMO cellular networks is not limited to interference mitigation. Instead, in this Thesis it is used the concept of interference management as a more general concept that can accommodate this more aggressive approach.

Based on the cooperative nature of interference management schemes one can distinguish:

- Non-cooperative techniques. Including randomization, cancellation, advanced transmission/reception and distribution of antennas.

- Cooperative techniques. This concept includes network coordination involving multiple cells (from the interference coordination schemes, like frequency reuse, to coordinated multi-cell multi-user MIMO schemes) and also relay-assisted cooperation.

The focus of this Thesis is on cooperative techniques. Nevertheless, this does not exclude the use of non-cooperative techniques in combination with cooperative techniques. For example, advanced receivers can be assumed to be used by coordinated cells for interference suppression. In fact, in this Thesis a Minimum Mean Square Error (MMSE) receiver with interference suppression is widely used for spatial multiplexing reception.

Specifically, the focus is on network coordination including only base stations, thus relay-assisted cooperation techniques are not considered. In the 3GPP, network coordination is known as CoMP, not including in this concept the use of relays. The main idea behind network coordination is to coordinate transmissions of different cells to reduce inter-cell interference and even turn interference into useful signal.

In order to perform the network coordination, the multiple coordinated points need to share some information, like data and channel state. It is possible to classify the different coordination approaches depending on the level of data and channel state sharing among the coordination points. Figure 1.4 presents several techniques considered by Samsung in [53] and the level of data and channel state sharing needed in each technique. Related to the data information sharing among coordination points, in [54], Gesbert *et al.* refer to the group of techniques that do not involve data sharing as "interference coordination" while the group of techniques that involve some kind of data sharing are referred to as "multi-cell processing". Moreover, in LTE-Advanced specifications [24], two classes of CoMP have been defined: joint processing and coordinated scheduling/beamforming. These classes correspond to multi-cell processing and interference coordination, respectively. Next, the state of the art in these topics is commented.

Interference coordination

Interference coordination is characterized by a modest amount of backhaul communication but can be useful with a high number of users as shown by Gesbert *et al.* [54]. Neither sharing of data information nor synchronization of base stations is needed. Transmission strategies, including scheduling, power control, beamforming and advanced coding methods, can be jointly considered in the coordination. Gesbert *et al.* [54] identify three types of interference coordination techniques:

- Coordinated power control. These techniques include joint scheduling and power control. Thus, classic Inter-Cell Interference Coordination (ICIC) methods, such as frequency reuse, fractional reuse or soft frequency reuse, fall in this group of techniques. Coordinated power control is a synonym of coordinated scheduling in the 3GPP terminology.

- Coordinated beamforming. These techniques can be used when the cells present multiple antennas. Then, coordination of the beamforming vectors of different cells can provide a performance improvement. Two kind of coordinated beamforming techniques are included in Figure 1.4 as an example: Collision Avoidance BeamForming (CA-BF) and Precoding

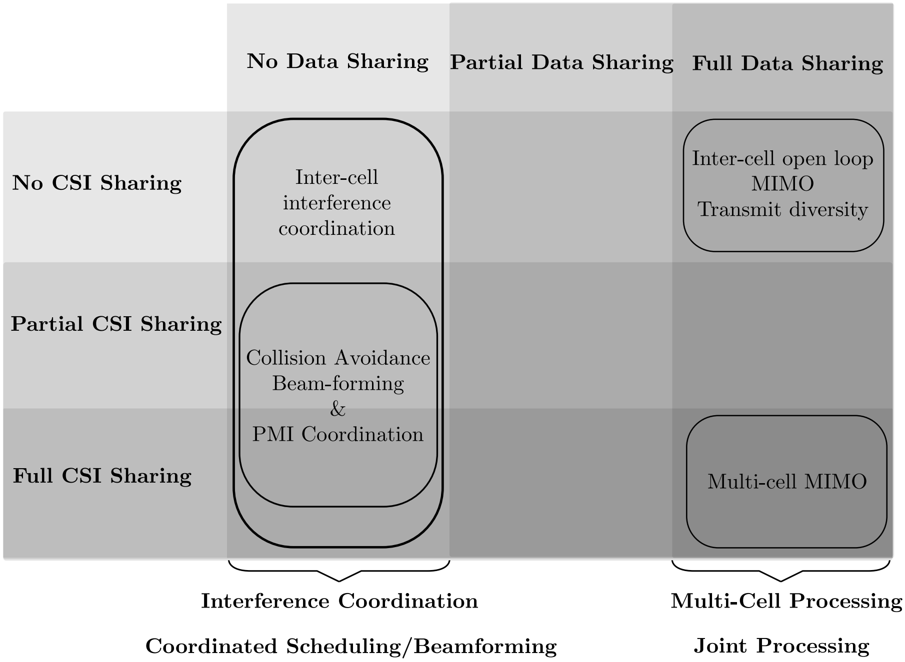


Figure 1.4: Grade of CSI/data sharing in cooperative interference management techniques

Matrix Indicator (PMI) coordination. In CA-BF, the UEs send feedback information about least interfering precoder of the interfering eNodeB and suggest the interfering eNodeB to use it. In the PMI coordination technique, a set of PMIs is recommended to be used in the interfering eNodeBs. PMI coordination improves the scheduling flexibility and thus brings higher overall network throughput at the cost of increased overhead. Coordinated beamforming is the logical step forward from coordinated scheduling with MIMO capable cells.

- Coding for interference mitigation. It includes special coding that permits the receiver to detect interference and subtract it from the received signal.

Multi-cell processing

Multi-cell processing is also known as coordinated multi-cell multi-user MIMO, networked MIMO or Joint Processing (JP) among other names. In these schemes cells are synchronized and share data streams through high-capacity backhaul links. In multi-cell processing the antennas of multiple cells are viewed as a single antenna pool. Therefore, concepts developed in the field of MU-MIMO can be reused in this field. The uplink channel in this new scenario is a multiple access channel like those considered in MU-MIMO with a single virtual receiver. The downlink channel is similar to a broadcast channel with a single virtual transmitter.

The main ideas of multi-cell processing can be found in, for example, the works of Zhang *et al.* [55], Karakayali *et al.* [56], Foschini *et al.* [57], Jing *et al.* [58] and Simeone *et al.* [59].

The ideal scenario where the main benefit from multi-cell processing could be obtained is a coordinated network with perfectly synchronized points, with complete and perfect knowledge of the channel state and complete sharing of data information. Nevertheless, in general, this ideal scenario is neither realistic nor practical. It is not realistic to assume perfect channel state knowledge due to channel estimation errors and changing channel and interference conditions. Besides, it is not realistic to assume perfect synchronization. In addition, it is not practical to assume the coordination of an entire network with complete channel state and data sharing due to the high back-haul requirements implied. This fact is especially clear in large networks. Furthermore, the common assumption of centralized implementation is not practical in large networks due to the high complexity that this requires, that is to say, there are scalability problems. The next scheme summarizes the ideal assumptions that limit the practical application of multi-cell processing:

- Non-realistic assumptions
  - Perfect channel state knowledge
  - Perfect synchronization
- Non-practical assumptions
  - Complete channel state sharing
  - Complete data information sharing
  - Centralized implementation in large networks

Recently, the gap from theory to practice is being filled. In this process, there is a need to relax the ideal assumptions and study how performance degrades. For example, it is necessary to study how the performance decreases when channel estimation errors are considered. The next step forward is to develop algorithms robust to non-ideal behaviours. Next, some topics related to this non-ideal framework are commented.

A. Impact of channel state accuracy

Accuracy of channel state knowledge in coordinated points is affected by several issues like channel estimation errors, channel state information quantization and variation in channel state conditions (channel and interference). Similar problems are found in the single-cell MU-MIMO downlink transmission and even in the single-link MIMO transmission. The more important difference here is the latency existing in the distribution of estimates through the coordinated network. In multi-cell processing it is important to consider that enough resources (e.g. pilot signals) should be devoted to channel estimation. The amount of needed resources increases with the size of the coordinated network, then there is a problem with scalability. Moreover, it is important to consider that channel estimation of distant sources is more difficult and thus large coordinated networks will be more affected by channel state inaccuracy.

B. Efficient representations of channel state information (use of limited channel state information)

This topic tries to reduce the signaling overhead needed in the coordination. Techniques employed in multi-cell processing are equal to those used in multi-user MIMO, for example in [60] and [61]. This topic is highly related with the previous one. The idea is to obtain representations of the channel state with high accuracy but low overhead. Another option is to obtain representations whose accuracy is at least known (in terms of statistics) in order to develop robust algorithms that take into account this accuracy.

C. Distributed precoding and decoding algorithms

In the first studies about multi-cell processing it is assumed a centralized architecture in which all the coordinated points transmit channel state and data information to a central controller. This central controller makes coordination decisions (precoding and decoding algorithms) and sends them back to the coordination points. Nevertheless, this scheme presents scalability problems. In large networks high backhaul requirements (in terms of capacity and delay) and high complexity in the central controller algorithms are expected. If the centralized option presents problems, then the logical option is to distribute the work. The simplest situation is that in which every coordinated point has full coordination information and performs the same algorithms that the central controller.

The advantage of this situation is that communication of decisions from the central controller is avoided and hence a reduction in the delay is obtained. However, this advantage is not so important. Where distribution presents an important advance is when considering partial sharing of channel state and/or information. Then, the multiple coordinated points are agents that cooperate although they have not the same view of the system. This problem has been dealt with by Zakhour and Gesbert in [62] using team decision theory.

D. Clustering

The coordination of an entire network presents scalability problems. Instead of coordinating the whole network, it is possible to create clusters of coordinated points, that is, divide a network into a set of coordinated subnetworks. This technique reduces the computational complexity and the back-haul demands. Nevertheless, since some cells in the network are not coordinated, some inter-cell interference appears, especially in the cells found in the cluster borders. Impact of clustering on the information-theoretic capacity of multi-cell processing has been considered by Bacha *et al.* [63], Somekh *et al.* [64] and Huh *et al.*[65], among other works. The combination of clustering with frequency planning has been proposed by Caire *et al.* [66] and Katranaras *et al.* [67] to reduce inter-cluster interference. Some papers have proposed to use cooperation only with users in the cell edge. These ideas are referred to as dynamic clustering. The problem is to group users appropriately [68][69]. Since cells in the edge of clusters suffer higher levels of inter-cluster interference, unfairness for mobiles in these cells appears. A solution to enhance fairness among cells is to define different clustered configurations of the network in such a way that the clustering configuration dynamically changes and cells become border or center cells [63][66][67].

Concerning the state of CoMP in the 3GPP specifications, Release 10 of LTE (LTE-Advanced) did not include explicitly CoMP. It was not until Release 11 when means to implement CoMP in intra-eNodeB schemes with ideal backhaul (low latency, high capacity) were provided. In subsequent Releases, signalling between eNodeBs has been provided to implement inter-eNodeB schemes, i.e., to consider non-ideal backhauls. However, it is important to highlight that the support of inter-eNodeB schemes especially focuses on interference coordination more than in multi-cell processing.

1.3 Problem and Thesis scope

1.3.1 Problem definition and hypothesis

This thesis focuses on the downlink of an LTE system, dealing with three main problems. First, there is a need to evaluate the LTE standard considering a realistic network deployment. The second problem is the development of advanced MAC-related techniques. The last problem is the need to develop realistic and practical coordination techniques among base stations in LTE-Advanced, constrained to the framework of CoMP, and to evaluate the performance of such kind of coordinated schemes in a realistic scenario.

Concerning the first problem, the evaluation of LTE, many assessments have been made in this field. Nevertheless, this work is not concluded. First, the precision of evaluation models have an impact on the obtained performance, but this effect has not received enough attention. The impacts of channel implementation, channel estimation, interference model and link abstraction granularity (in time and frequency) need to be further explored. Secondly, not many assessments have paid attention to the particular relevance of each LTE building block on the overall system performance. Moreover, the interactions between some building blocks of LTE have been neglected so far. For example, performance of scheduling is affected by HARQ constraints and this fact has not been sufficiently studied. Furthermore, realistic conditions, including actual channel quality indicator reports, need to be taken into account.

Concerning the second problem, since LTE is a standard opened in some aspects, different implementations are allowed and encouraged. This fact implies that the assessment of a particular implementation is only valid for the conditions of this assessment, including the specific implementation and deployment scenario. Although reference implementations can be used to perform LTE assessments providing reference performance figures, results obtained with these implementations can be improved if more developed algorithms or techniques are used. Therefore, although some work has been done in this field, it does not preclude the finding of implementations better-suited to specific scenarios than those already studied. For example, improvements of link adaptation algorithms in scenarios with MIMO and with fast changing interference conditions are needed according to the literature review. Besides, envisioned interactions among LTE building-blocks suggest that an improved performance could be obtained taking into account these interactions.

Concerning the third problem, many proposals about CoMP assume non-realistic and/or non-practical assumptions. Moreover, the first 3GPP proposals are restricted to intra-site coordination. As a conclusion, there is a need to develop realistic, practical and low complexity inter-site coordination techniques constrained to the 3GPP CoMP framework. Additionally, it is required to evaluate the effect of realistic deployments (including realistic channel quality indicators and realistic back-haul capability assumptions) on previously proposed CoMP methods and the effect on the new methods proposed in this Thesis.

The problematic presented has motivated the definition of several hypotheses:

- Hypothesis 1 : system performance is affected by system modelling precision.
- Hypothesis 2 : key building blocks of the system interact and the joint effect is not just the addition of particular effects.
- Hypothesis 3 : link adaptation algorithms for MIMO can be improved in LTE.
- Hypothesis 4 : scheduling can be improved in LTE.
- Hypothesis 5 : low complexity inter-site CoMP algorithms can be defined in LTE-Advanced providing a considerable performance improvement on the system performance compared to non-coordination and intra-site coordination under realistic conditions.
- Hypothesis 6 : ideal CoMP algorithms are heavily affected by realistic conditions, while sub-optimal solutions could be more robust to them and hence be more suited to actual scenarios.

1.3.2 Thesis objectives

This Thesis aims at making the most of LTE-Advanced through the improvement of some radio resource management techniques in the downlink of the system including scheduling, LA and through cooperation among base stations within the framework of CoMP as defined in LTE-Advanced, considering realistic and practical assumptions, and defining low complexity algorithms.

In order to achieve the objective of this Thesis it is required, firstly, to study and assess the performance of LTE and, then, to develop an adequate coordination mechanism to enhance user experience, mainly in the cell edge where performance indicators are highly degraded, while keeping a low and realistic signalling load. Then, the main research objective can be further divided into several partial objectives listed below:

- To develop a complete simulation platform for LTE, including link-level simulator, system-level simulator and link-to-system level abstraction model.

- To assess the performance of the LTE system and its evolution, LTE-Advanced, including:
  - Evaluation of the impact of the system modelling accuracy on the system performance.
  - Evaluation of the impact of each key building block separately and the interaction among key building blocks including the impact of HARQ constraints on scheduling.
- To optimize the performance of the LTE system and its evolution, LTE-Advanced, including:
  - Enhancement of link adaptation algorithms suited for MIMO operation.
  - Enhancement of scheduling algorithms operation.
- To design low complexity and inter-site capable cellular coordination mechanisms for LTE-Advanced, considering the standardized signalling and based on the distribution of the decision making and clustering.
- To analyse the performance improvement provided by CoMP and its behaviour in realistic scenarios, both for already proposed methods and new methods.

1.4 Thesis outline

The Thesis is divided into six chapters and three appendices. The first chapter is the introduction and the last chapter concludes the work and presents the future work. A brief description of the rest of chapters is provided below:

Chapter 2 describes the main features of the LTE system as they are presented in the standard. The RRM framework of LTE is presented and the state-of-the-art implementations of the functions most relevant for this Thesis are presented. This general description is important to clarify the framework in which the rest of the Thesis is developed.

Chapter 3 describes the methodology followed in this Thesis to evaluate the performance of the LTE system. Evaluation objectives as well as evaluation scenarios and methods are well defined. Concerning the evaluation methods, special focus is on simulation that is the primary methodology of this Thesis. An important issue is the description of the system modelling used in this Thesis and the contributions of this Thesis to the field of system modelling and evaluation of LTE.

Chapter 4 studies the support that the LTE specifications provide to perform the LA, with special focus on the opened issues. Concerning these issues it is worth noting that some typical LA processes are explained, such as the OLLA process. Moreover, some mechanisms to calculate precoders from CSI are explained.

Chapter 5 starts with the definition of the LTE scheduling problem and the proportional fairness scheduling algorithms. Afterwards, a set of modifications to the typical implementations are presented and their performance assessed. It is demonstrated that the modifications proposed are useful to optimize the operation of the common implementation.

Chapter 6 explains the CoMP feature and its effect on system performance. The focus is on the study of a low complexity algorithm with realistic assumptions, which falls in a special family of CoMP algorithms with coordinated scheduling and beamforming.

Chapter 7 presents a reference performance evaluation of LTE and LTE-Advanced using typical implementations of the RRM algorithms, or those suggested in previous chapters, in the scenarios of the IMT-Advanced evaluation process.

Chapter 8 evaluates the performance of a modified LTE-Advanced variant in some of the target scenarios of the future 5G. In this kind of evaluation, the focus is on realistic scenarios with realistic traffic models and with usually harsh environments or very demanding service requirements. In these scenarios is where CoMP has provided the best results in this Thesis.

Chapter 9 summarises the main conclusions of the Thesis, comments a set of new research trends in 5G, and presents future research lines.

In order to support the work, two appendix chapters are provided:

Appendix A describes the system level simulation platform used in this Thesis and the calibration process performed to ensure the validity of its outputs.

Appendix B explains the assessment methodology used to ensure the statistical significance of results obtained through simulation along the Thesis.

1.5 Publications

The work developed during this Thesis made possible the publication of the following journal and conference papers, reports with special relevance and book chapters. Also, a set of publications resulting from work closely related with the Thesis (preliminary work or work based on the outcome of the Thesis) is also provided.

Journals

[J1] K. Safjan, V. D'Amico, D. Bultmann, **D. Martin-Sacristan**, A. Saadani, and H. Schoneich, "Assessing 3GPP LTE-advanced as IMT-advanced technology: the WINNER+ evaluation group approach," *IEEE Communications Magazine*, vol. 49, no. 2, pp. 92–100, Feb. 2011.

[J2] **D. Martin-Sacristan**, J. F. Monserrat, J. Cabrejas-Peñuelas, D. Calabuig, S. Garrigas, and N. Cardona, "On the way towards fourth-generation mobile: 3GPP LTE and LTE-advanced," *EURASIP Journal on Wireless Communications and Networking*, vol. 2009, pp. 1–10, Mar. 2009.

[J3] **D. Martin-Sacristan**, J. F. Monserrat, J. Cabrejas-Peñuelas, D. Calabuig, S. Garrigas, and N. Cardona, "3GPP Long Term Evolution: Paving the Way towards Next 4G," *WAVES*, vol. 2009, pp. 96–105, 2009.

[J4] **D. Martin-Sacristan**, J. F. Monserrat, V. Osa, and J. Cabrejas, "LTE-Advanced System Level Simulation Platform for IMT-Advanced Evaluation," *WAVES*, vol. 2011, pp. 15–24, 2011.

[J5] J. F. Monserrat, S. Inca, J. Calabuig, and **D. Martin-Sacristan**, "Map-Based Channel Model for Urban Macrocell Propagation Scenarios," *International Journal of Antennas and Propagation*, vol. 2015, pp. 1–5, Apr. 2015.

Conferences

[C1] **D. Martin-Sacristan**, J. Cabrejas, D. Calabuig, and J. F. Monserrat, "MAC Layer Performance of Different Channel Estimation Techniques in UTRAN LTE Downlink," in *IEEE 69th Vehicular Technology Conference Spring (VTC Spring 2009)*, Apr. 2009, pp. 1–5.

[C2] **D. Martin-Sacristan**, J. F. Monserrat, D. Calabuig, and N. Cardona, "Time-Frequency Coupled Proportional Fair Scheduler with Multicarrier Awareness for LTE Downlink," in *IEEE 81st Vehicular Technology Conference Spring (VTC Spring 2015)*, May 2015, pp. 1–5.

Reports with special relevance

- [R1] WINNER+, "Evaluation of IMT-Advanced candidate technology submissions in Documents IMT-ADV/6, IMT-ADV/8 and IMT-ADV/9 by WINNER+ Evaluation Group," Doc. IMTADV/ 22, Jul. 2010.
- [R2] J. Olmos, S. Ruiz, M. García-Lozano, and **D. Martin-Sacristan**, "Link Abstraction Models Based on Mutual Information for LTE Downlink," COST 2100 TD(10)11052, Jun. 2010.
- [R3] METIS, "Simulation Guidelines," Deliverable D6.1, Sep. 2013.
- [R4] METIS, "Intermediate system evaluation," Deliverable D6.3, Aug. 2014.
- [R5] METIS, "Report on simulation results and evaluations," Deliverable D6.5, Mar. 2015.

Book chapters

- [B1] **D. Martin-Sacristan**, "Chapter 8: Análisis de prestaciones de LTE", in *3GPP LTE: Hacia la 4G móvil*, N. Cardona, J. J. Olmos, M. García and J. F. Monserrat, Marcombo S.A., 2011.
- [B2] J. F. Monserrat, G. Auer, **D. Martin-Sacristan** and P. Sroka, "Chapter 2: Radio Resource Management", in *Mobile and Wireless Communications for IMT-Advanced and Beyond*, A. Osseiran, J. F. Monserrat and W. Mohr, Wiley, 2011.
- [B3] M. Werner, V. D'Amico, **D. Martin-Sacristan**, J. F. Monserrat, P. Skillermark, A. Saadani, K. Safjan and H. Schöneich, "Chapter 10: The End-to-end Performance of LTE-Advanced", in *Mobile and Wireless Communications for IMT-Advanced and Beyond*, A. Osseiran, J. F. Monserrat and W. Mohr, Wiley, 2011.
- [B4] J. F. Monserrat, M. Fallgren, **D. Martin-Sacristan** and J. Lianghai, "Chapter 14: Simulation Methodology", in *5G Mobile and Wireless Communications Technology*, A. Osseiran, J. F. Monserrat and P. Marsch, Cambridge University Press, 2016.

Related work

- [W1] J. Calabuig, J. F. Monserrat, **D. Martin-Sacristan** and J. Olmos, "Comparison of multicast/broadcast services in Long Term Evolution Advanced and IEEE 802.16m networks," in *Wireless Communications and Mobile Computing*, vol. 14, issue 7, pp. 717–728, May 2014.

[W2] J. Cabrejas, P. Gualda, J. F. Monserrat and **D. Martin-Sacristan**, “Application of MIH for the lightweight deployment of LTE-advanced systems through mobile relaying,” in *EURASIP Journal on Wireless Communications and Networking*, vol. 2012, Mar. 2012.

[W3] D. Calabuig, J. F. Monserrat, **D. Martin-Sacristan**, and N. Cardona, “Joint Dynamic Resource Allocation for QoS Provisioning in Multi-Access and Multi-Service Wireless Systems,” in *Mobile Networks and Applications*, vol. 15, issue 5, pp. 627–638, Oct. 2010.

[W4] **D. Martin-Sacristan**, J. F. Monserrat, D. Calabuig, and J. Díaz, “Estudio de la Adaptación al Enlace en el Sistema de Comunicaciones Móviles 3.5G HSDPA,” *Gerencia Tecnológica Informática*, vol. 7, pp. 37–47, 2008.

[W5] **D. Martin-Sacristan**, J. F. Monserrat, D. Calabuig, and N. Cardona, “HSDPA Link Adaptation Improvement Based on Node-B CQI Processing,” in *International Symposium on Wireless Communication Systems (ISWCS)*, Oct. 2007, pp. 597–601.

[W6] J. F. Monserrat, A. Saul, G. Auer, T. Clessienne, A. Otyakmaz, S. Redana, R. Schoenen, P. Sroka, **D. Martin-Sacristan**, and N. Papaoulakis, “Advanced Radio Resource Management for IMT-Advanced in the Framework of WINNER+ Project,” in *ICT Mobile Summit Conference*, 2009.

[W7] J. F. Monserrat, P. Sroka, G. Auer, J. Cabrejas, **D. Martin-Sacristan**, A. Mihovska, R. Rossi, A. Saul, and R. Schoenen, “Advanced Radio Resource Management for IMT-Advanced in WINNER+ (II),” in *Future Network and Mobile Summit*, 2010.

Chapter 2

LTE Description

This Thesis focuses on the study of the Long Term Evolution (LTE) radio interface in Downlink (DL) and its evolution. In order to perform any assessment, it is necessary to have a deep knowledge of the system under study. This chapter provides this required information. The most relevant LTE standard features, from the point of view of this Thesis, are presented. However, it is remarkable that the LTE standard is opened to different implementations of some functions and algorithms. In this chapter, state-of-the-art implementations of these opened features are also presented.

The chapter has been divided into the next sections:

- Section 2.1 describes the network architecture, including both the radio access network and the core network.

- Section 2.2 presents the LTE radio protocol architecture for both control and user planes, describing the main functions of each protocol.

- Section 2.3 introduces the main features of the LTE physical layer and the configuration of this layer assumed in this Thesis. Moreover, the processing for the downlink shared data channel both in the evolved Node B (eNodeB) and the User Equipment (UE) is detailed. Finally, Multiple-Input Multiple-Output (MIMO) processing done in the physical layer is explained in-depth.

- Section 2.4 explains the Medium Access Control (MAC) layer and its main functions. Therefore, particular attention is paid to link adaptation, Hybrid ARQ (HARQ), and scheduling.

- Section 2.5 outlines additional LTE features not especially relevant for this Thesis, such as relaying, carrier aggregation, and self-organizing networks, providing references for the interested readers.
- Section 2.6 summarises the main conclusions of this chapter.

2.1 Network architecture

LTE is one of the two parts of a more global concept known as Evolved Packet System (EPS). This technology provides the users with Internet Protocol (IP) connectivity to a Packet Data Network (PDN) in order to access the Internet, or any other service [70]. With this aim, the EPS has a radio access part and a core part. The term LTE refers to the evolution of the technology used in the radio access of the EPS. While the technology itself is termed Evolved UMTS Terrestrial Radio Access (E-UTRA), the network providing connectivity to the users is called Evolved UMTS Terrestrial Radio Access Network (E-UTRAN). The term System Architecture Evolution (SAE) refers to the evolution of the core part of the system, being Evolved Packet Core (EPC) the name given to the core network of the EPS. It is worth mentioning that it is quite extended the use of the term LTE to refer to the whole EPS system.

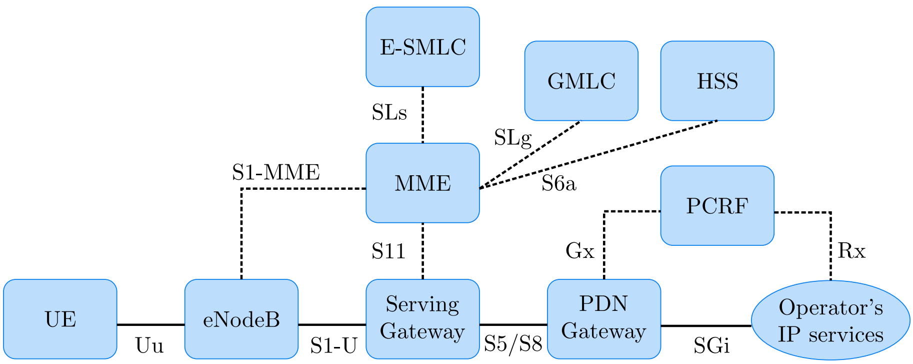


Figure 2.1: The EPS network elements [70].

In an EPS network architecture we find a plurality of nodes, see Figure 2.1 for an overall description of the network elements and their interfaces. Firstly, we find the mobile terminals, known as UEs in the specifications. Secondly, base stations, known as eNodeBs. Finally, multiple nodes that form the core network. From the latter group, it is interesting to highlight the Serving Gateway (S-GW), the Packet Data Network Gateway (PDN-GW), and the Mobility

Management Entity (MME). The S-GW routes data coming from the eNodeBs to the PDN-GW that connects the core to a PDN. Meanwhile, the MME has functions related to the mobility and session control of the users.

In contrast to previous standards, the E-UTRAN is very simple and consists exclusively of one type of node: the eNodeBs. An exemplary illustration of the E-UTRAN architecture is shown in Figure 2.2. The eNodeBs have an interface to the users, the radio interface, known as Uu interface in the specifications. Another relevant interface for this Thesis is the X2 interface that interconnects the eNodeBs, as shown in Figure 2.2.

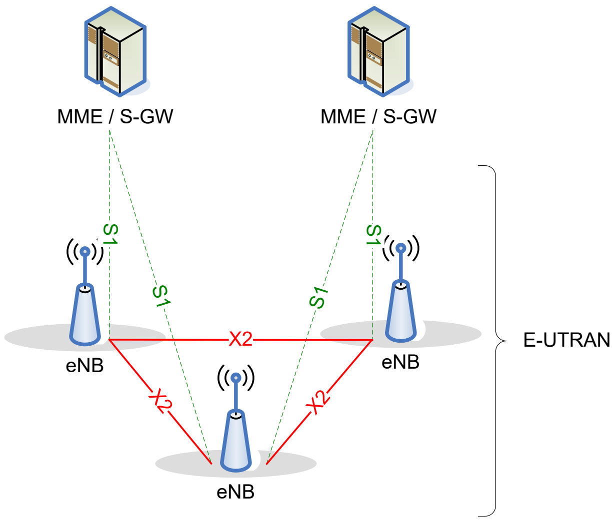


Figure 2.2: E-UTRAN architecture [14].

Concerning the provision of Quality of Service (QoS) to the users, the EPS system provides EPS bearers to the flows between UEs and PDN-GWs. Those bearers are characterized by a specific set of QoS parameters. To guarantee this QoS, other bearers are also defined in each network interface located between EPS bearer ends. Specifically, in the radio interface this bearer is called radio bearer.

2.2 Radio protocol architecture

The E-UTRA considers two types of entities: the eNodeB and the UE. Both entities present a layered protocol architecture that can be divided into two planes: a control plane and a user plane. Let us focus on the radio interface, the Uu interface, and the protocols that are terminated in this interface. Figure 2.3 shows the user plane protocol stack in the left side, and the the control plane protocol stack in the right side. The layers with stripped pattern are not terminated in the radio interface but are used between UE and core network elements instead.

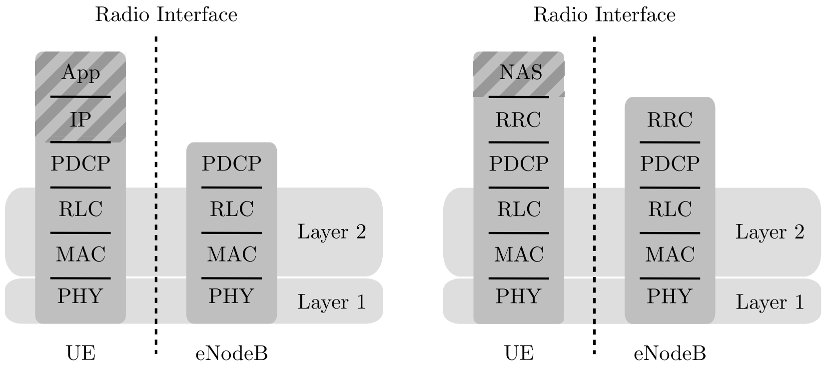


Figure 2.3: Radio protocols. User plane protocols in left side and control plane protocols in right side.

Four layers are common to both planes: the Packet Data Convergence Protocol (PDCP) layer, the Radio Link Control (RLC) layer, the MAC layer, and the Physical (PHY) layer. However, the Radio Resource Control (RRC) layer is only found in the control plane. The remainder of this section introduces all these protocols.

The RRC layer is in charge of controlling the broadcasting of system information, the paging, the establishment/modification/ending of radio bearers, the configuration and reporting of measurements, the configuration of the radio interface (the lower protocol layers) and the fulfilment of QoS among other functions. The latter two functions are heavily related since the specific configuration of the lower protocol layers depends on the expected QoS.

The PDCP layer receives packets from higher layers through the radio bearers. In the user plane, these bearers are known as data radio bearers and are used to transmit IP packets. In the control plane they are called signalling radio bearers and are used to convey RRC signalling. PDCP functions include IP header compression, ciphering, and control data integrity protection. From this set of functions, the only relevant one in this Thesis is IP header compression. When IP packets are used, this function has an effect the amount of information that must be managed by lower layers. There is one PDCP entity in the eNodeB per radio bearer.

The RLC layer is in fact a layer 2 sub-layer. RLC functions are performed by RLC entities. For each RLC entity configured at the eNodeB, there is a peer RLC entity at the UE and vice versa. A transmitting RLC entity receives RLC Service Data Units (SDUs) from higher layers, and sends RLC Protocol Data Unit (PDU) to its counterpart that makes the contrary operation. The RLC entities can be configured to perform data transfer in one of three modes: Transparent Mode (TM), Un-acknowledged Mode (UM), and Acknowledged Mode (AM). In TM, RLC entity performs no actions over transferred data. In UM, PDUs can be ordered in reception. Additionally, in AM, erroneous PDUs can be retransmitted. The latter mode requires the use of special entities able to make both transmissions and receptions (each entity is able to transmit and receive SDUs and acknowledgements). We can see the RLC in the transmitter side as a big buffer where packets are stored in different queues, depending on the radio bearer they come from, waiting for transmission opportunities granted by lower layers. In the receiver, the RLC layer passes the packets to higher layers ordering them, and even requesting retransmissions if losses are detected. It is interesting to note that both PDCP and RLC entities are basically equal in the eNodeB and in the UE, the main difference is between transmitting and receiving entities.

The MAC layer is also a layer 2 sub-layer. MAC services provided to upper layers are data transfer and radio resource allocation. The access to these services is through logical channels via the RLC layer. There is a one-to-one mapping of radio bearers to logical channels, i.e., there is no multiplexing in RLC layer. The interface between the MAC and its lower layer is through transport channels. The information is sent through transport channels in blocks known as Transport Blocks (TBs). Each transport channel has a specific Transmission Time Interval (TTI) defined as the time between transmissions of TBs through the transport channel.

Some functions conducted by MAC are common to eNodeB and UE: mapping between logical channels and transport channels, multiplexing of MAC SDUs from one or different logical channels onto TBs to be delivered to the physical layer on transport channels in the transmitter, demultiplexing of MAC SDUs from one or different logical channels from TBs delivered from the physical layer on transport channels in the receiver, and error correction through HARQ. However, in contrast to the previous layers, there are important differences between the MAC layer in the eNodeB and in the UE. Note that the common part is related to transmission of data once the TB is formed. But the difference is on the process previous to the TB building. Basically, the MAC at the eNodeB holds more intelligence and is in charge of priority handling between UEs by means of dynamic scheduling (both in downlink and uplink), decision of the transport format (both in downlink and uplink), and priority handling between logical channels of one UE in downlink. The MAC at the UE is in charge of priority handling between logical channels in uplink. In order to accomplish this set of functions, three features are of paramount importance in MAC: scheduling, HARQ management, and Link Adaptation (LA). These features are explained in subsequent sections.

PHY layer, or equivalently layer 1, is the lowest layer of the protocol architecture. The set of services provided to higher layers by the physical layer is: error detection on the transport channel and indication to higher layers, Forward Error Correction (FEC) encoding/decoding of the transport channel, HARQ soft-combining, rate matching of the coded transport channel to physical channels, mapping of the coded transport channel onto physical channels, modulation and demodulation of physical channels, radio characteristics measurements and indication to higher layers, MIMO antenna processing, transmit diversity and beamforming. All these functions have been considered in the present Thesis. Additionally, power weighting of physical channels, frequency and time synchronisation and Radio Frequency (RF) processing are not used. Instead, synchronization is assumed to be ideal. The access to PHY services is through the use of a transport channel via the MAC layer. The PHY configuration is explained in the next section.

Figure 2.4 presents the main functions of each protocol layer and the relations among layers. One of the most relevant features of the LTE protocol architecture is that some cross-layering exists between layer 1, layer 2 and even layer 3 as shown in Figure 2.4.

The following sections explain, in detail, the key PHY and MAC features and interactions among them.

2.3 PHY layer

2.3.1 Fundamentals and selected configuration

LTE physical layer is based on Orthogonal Frequency Division Multiplexing (OFDM) with Cyclic Prefix (CP) in DL, and on Single Carrier-Frequency Division Multiple Access (SC-FDMA) in Uplink (UL). Two duplex modes are supported: Frequency Division Duplexing (FDD), supporting full duplex and half duplex operation, and Time Division Duplexing (TDD). The focus in this Thesis is on the FDD mode of LTE DL.

Layer 1 is defined in a bandwidth agnostic way based on Resource Blocks (RBs), allowing the LTE Layer 1 to adapt to various spectrum allocations. A RB spans 180 kHz, which corresponds to either 12 sub-carriers with a subcarrier bandwidth of 15 kHz or 24 subcarriers with a subcarrier bandwidth of 7.5 kHz each, over a slot duration of 0.5 ms. It is possible to define transmission

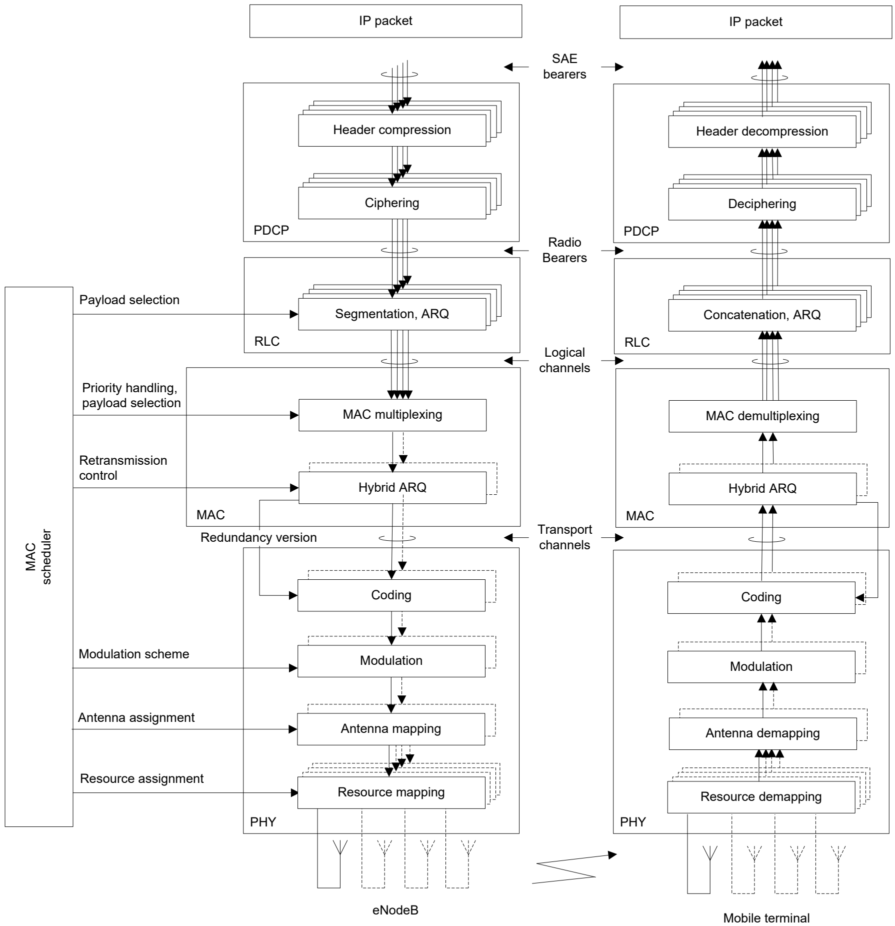


Figure 2.4: Protocol architecture [71].

bandwidths of 6, 12, 25, 50, 75 and 100 RBs. They correspond to 1.4, 5, 10, 15 and 20 MHz channel bandwidths, respectively (where guard bands are considered to determine the channel bandwidth).

A normal CP and an extended CP have been configured that differ in the CP length, being the second one only allowed with the 7.5 kHz sub-carrier spacing.

Two radio frame structures are defined. The radio frame structure type 1 is used for FDD (for both full duplex and half duplex operation) and has a duration of 10 ms. The frame is divided into 20 slots of 0.5 ms. Two adjacent slots form one subframe of length 1 ms. The radio frame structure type 2 is used for TDD and consists of two half-frames with a duration of 5 ms each and containing each one 8 slots of length 0.5 ms and three special fields (DwPTS, GP and UpPTS) which have configurable individual lengths and a total length of 1 ms. A subframe consists of two adjacent slots, except for subframes 1 and 6, which consist of DwPTS, GP and UpPTS. Both 5 ms and 10 ms switch-point periodicity are supported. Further details on the LTE frame structure can be found in [72].

The number of OFDM symbols per subframe depends on the CP configuration as well as on the sub-carrier bandwidth: 7 symbols for normal CP and 15 kHz subcarrier bandwidth, 6 symbols for extended CP and 7.5 kHz sub-carrier bandwidth, and 3 symbols for extended CP and 7.5 kHz sub-carrier bandwidth. It is worth noting that in this Thesis the configuration with normal CP and 15 kHz is used.

Figure 2.5 summarises the above mentioned characteristics of the physical layer. In the higher part of the figure, a frame of a LTE system spanning 6 RBs in frequency is shown. In the lower part, there is a zoom of a specific pair of RBs. Moreover, a new concept shown in this figure is the Resource Element (RE) that is the smallest resource unit of the system, spanning one subcarrier in frequency and one OFDM symbol in time. Each RE is able to transmit a modulated symbol in case of data transmission. Three modulation schemes are supported in LTE: Quadrature Phase Shift Keying (QPSK), 16-Quadrature Amplitude Modulation (QAM) and 64-QAM. It is also important to note that a matrix of resources, or resource grid, exists in each antenna of the transmitter. This way, each antenna may transmit a different signal. This is the reason to show two different grids for the pair of resource blocks shown in detail. In fact, we have coloured specific REs in a different way in each grid. This colouring denotes the mapping of a special type of signals known as reference signals, which is different for each antenna.

Different physical channels conveying coded information (from higher layers or the layer 1 itself) are defined for DL and UL. The most important channels in this Thesis are:

- the Physical Downlink Shared Channel (PDSCH), which transmits data from the eNodeB to the UE,

- the Physical Downlink Control Channel (PDCCH), which sends resource allocation and HARQ information related to the PDSCH from the eNodeB to the UE, and

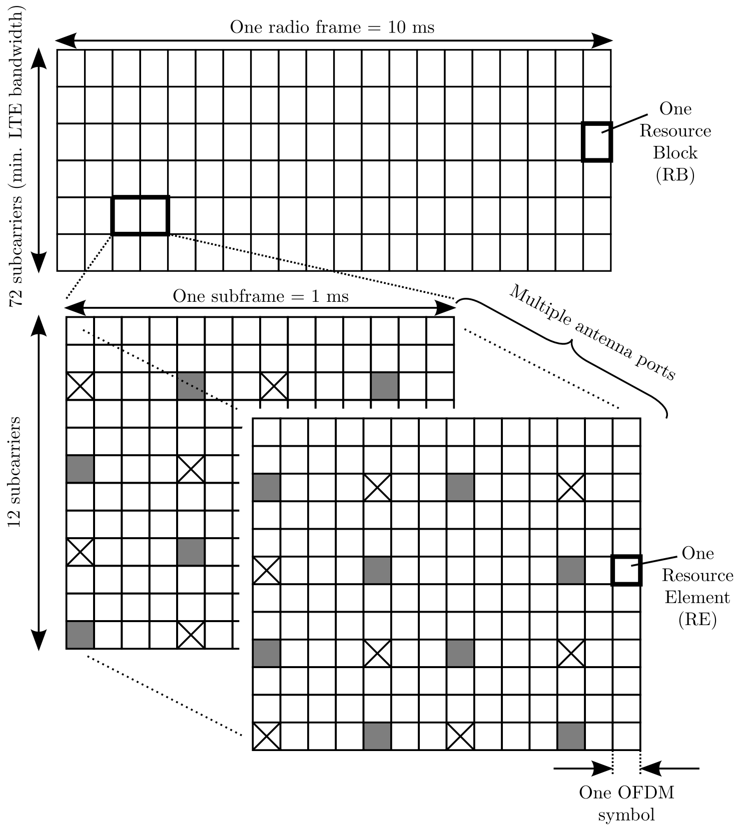


Figure 2.5: PHY resources. Normal cyclic prefix length and 15 kHz subcarrier bandwidth are assumed [70].

- the Physical Uplink Shared Channel (PUSCH) and Physical Uplink Control Channel (PUCCH), which can transmit HARQ acknowledgements corresponding to the DL data flow, channel quality and scheduling requests from the UE to the eNodeB.

In addition, physical signals, without higher layers information but used to perform physical layer related functions are also defined. These signals are the reference signals (used, e.g., to estimate the channel) and the synchronization signals.

Table 2.1 summarizes the physical layer configuration assumed in this Thesis.

Table 2.1: Physical configuration in the Thesis

Parameter	**Value**
Duplex mode	FDD
Sub-carrier bandwidth	15 kHz
CP length	normal

2.3.2 Reference signals

In LTE Release 8, three types of reference signals were included:

- **Cell-specific reference signals** are transmitted in all downlink subframes over the whole bandwidth on one or several of antenna ports 0 to 3. Antenna port is the specific term used in the standard to refer to an antenna or group of antennas. They are cell-specific in the sense that each cell transmits a different signal sequence. They are used for Channel State Information (CSI) calculation and data demodulation. Their exact position depends on the slot number within a radio frame, the OFDM symbol number within the slot, the type of cyclic prefix, and the cell identifier. A pseudo-random sequence is used for the reference signal generation. The resource elements used for the reference signal transmission in any of the antennas will not be used for transmission on any other antenna in the same slot. Figure 2.5 illustrates the resource elements selected for the transmission of pilot symbols within a pair of RBs considering two antennas and normal cyclic prefix. Resources in grey denote the position of pilot symbols, while resources with a cross denote no transmission. Note that in the RE where one antenna transmits a pilot, the other one does not transmit anything. This configuration allows the receiver to clearly differentiate the pilots coming from each antenna.

- **MBMS over Single Frequency Networks (MBSFN) reference signals** are used in Multimedia Broadcast and Multicast Service (MBMS) mode (see 2.5) to facilitate reception in System Frame Number (SFN) areas where multiple transmitters broadcast the same information. Their exact position in the resource grid depends on the cell identifier of the transmitting cell. The spacing of the resources devoted to these signals is larger in time than in frequency domain. The reason is that MBMS is designed for reception by non-high-speed terminals, but must support a highly selective channel in frequency.

- **UE-specific reference signals** are transmitted in downlink subframes and are UE-specific in the sense that signal sequence for a UE is transmitted only over the set of resources devoted to that UE. These signals are transmitted on antenna port 5 and are used with demodulation purposes, in such a way that UE-specific signals are precoded in the same way that data signals.

In LTE Release 9, reference signals were enhanced with the following actions:

- **Positioning reference signals** are included in specifications, transmitted on antenna port 6. They are used for the purpose of helping the terminals determine its position.

- **UE-specific reference signals** are allowed to be used on two antenna ports (7 and 8). Code domain orthogonalization of signals allows both antennas to use the same resources to transmit the reference signals. This way, overhead is minimized.

In LTE Release 10, reference signals are enhanced with the following actions:

- **UE-specific reference signals** are allowed to be used on 8 ports (7 to 14). In this case, ports 7, 8, 11 and 12, use the same resources to transmit the reference signals, although they are orthogonalized in code domain. The rest of antenna ports use a different set of resources, and, again, orthogonalize the signals between them in code domain.

- **Channel State Information Reference Signals (CSI-RS)** are included in specifications, transmitted on one, two, three or four antenna ports (using ports 15, 15 and 16, 15 to 18 and 15 to 22, respectively). Four different sets of CSI-RS resources are defined, each one used by a couple of antenna ports. The same set is used by ports 15 and 16, another by 17 and 18, and so on. Signals in different sets are orthogonal in frequency and time domains, while signals in the same set are orthogonal in code domain. CSI-RS resources can be configured for a UE as non-zero power or as zero-power, where zero-power means that the resources are left unused.

2.3.3 PDSCH processing at the eNodeB PHY

The transport channel providing access to PDSCH is the Downlink Shared Channel (DL-SCH). This transport channel can send one or two TBs per subframe from MAC to PHY, being its TTI length of 1 ms. Figure 2.6 represents the processing chain applied in the physical layer to the data blocks coming from the DL-SCH.

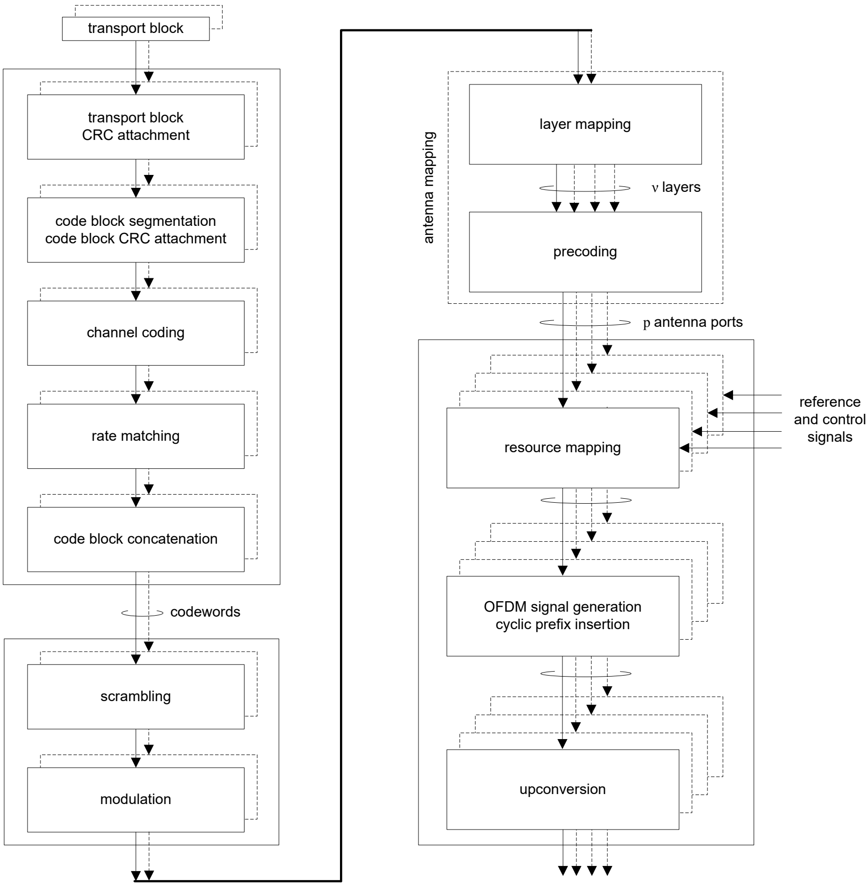


Figure 2.6: PHY layer processing chain for PDSCH transmission at the eNodeB

The first group of blocks deals with coding. First a Cyclic Redundancy Check (CRC) is added to the bits of the TB. This field is used in reception to check correctness of reception. Next, the transport block may be segmented and, in this case, a CRC is added to each segment. Segmentation occurs if the size of the input block is larger than a specific value, and will ensure that each segment size plus the CRC length is less than the largest permitted value. The reason for segmentation is to keep complexity of the next phase reduced. The addition of CRCs to each segment may be used in reception to stop the processing if any received segment presents errors. The third block is in charge of channel coding, made through a turbocoder with 1/3 rate. The coder is almost equal to the Universal Mobile Telecommunications System (UMTS) turbocoder, but presents changes in the input interleaver. In the next step, bits can be repeated or punctured to adjust the effective coding rate, and ensure that the number of output bits fits the resources allocated to the transmission. After coding and rate matching are applied over each segment, the resulting code blocks are concatenated to obtain a codeword, or two codewords in case of having two input TBs.

The following group of blocks deals with scrambling of bits, and modulation. The output is one or two blocks of modulated symbols.

These blocks enter the antenna mapping that will make some changes over them and will obtain as output as many streams of symbols as transmitting antenna ports. In a subsequent section this processing will be detailed.

The next step is the resource mapping that allocates a resource element to each symbol coming from the antenna mapping. This mapping takes into account that some resource elements are used to transmit control information or physical signals.

The last steps are the generation of the OFDM signal (Fast Fourier Transform (FFT) and cyclic prefix insertion) and the upconversion of the complex signal.

It is worth noting that several processing blocks are controlled by MAC functions. That is to say, there is a cross layering between the physical layer and the MAC layer. This cross layering is clearly depicted in Figure 2.4.

2.3.4 PDSCH processing at the UE PHY

Specifications do not state how the UE should do the physical layer processing in order to receive the data transmitted in downlink. Figure 2.7 presents an exemplary functional block diagram of the processing at the UE. Basically, the idea is to undo the processing done in the transmitter.

With this aim, the first important block is the channel estimation block. In LTE, a pilot-assisted channel estimation is done. Reference signals are used by the receiver to estimate attenuation and phase change suffered by the signal in the frequency domain. The usual method is to obtain a channel estimate on the pilot positions, i.e. the reference signals, and refine this estimation to obtain the channel in the rest of positions by interpolation. This interpolation may be a polynomial interpolation or a Minimum Mean Square Error (MMSE) filtering. The prior knowledge of correlations in time, frequency or spatial domains may be used to obtain an optimum interpolation filter. On the contrary, robust filters may be designed to cope with the most difficult scenarios. Relying on channel estimates, frequency domain equalization can be performed, which is a simple method to correct changes introduced by the channel in the transmitted data.

The equalization and demodulation block uses the channel estimates and the received signal to obtain the inputs to the decoder. These inputs can be hard or soft estimates of the bits that went out of the decoder in the transmitter. Implementations depend on the exact MIMO processing applied at the transmitter and can differ significantly depending on the level of complexity allowed: from least squares estimates to maximum likelihood demodulators.

Before the decoder a soft-buffer is found (HARQ soft combining in Figure 2.7). This buffer is where combination of multiple transmissions can be done. Through the application of the inverse of the rate matching operation done at the transmitter, each soft-output o the MIMO processing corresponds with one position of the soft-buffer. The idea is to obtain a block of information the more similar as possible to the block found after the coder in the transmitter, to reverse coding. With coding rates higher than 1/3, not all buffer positions will be filled. However, the main point is that in each transmission of the same block, the soft values will be mapped to different positions, therefore there will be more information in the buffer. Some values will be mapped to the same position. This fact will occur even for the first transmission if the code rate is lower than 1/3. But this is not a problem, since the new value can be added to the oldest one to obtain a more reliable soft-value.

The specific implementation of the decoder can be also very diverse and dependent on the computational complexity allowed. An interesting point is the parallelisation allowed by the fact of having a channel coding function applied over different segments. Moreover, the fact of having each segment a different CRC saves calculations through stopping other segments decoding in case of detection of an erroneous block.

2.3.5 MIMO processing

Once the main characteristics of the PHY layer have been explained, it is possible to go in depth into the specific issue of MIMO processing.

MIMO has been supported from the first LTE release as a means to achieve high throughput. Besides, MIMO capabilities have been improved along LTE releases. Next, this evolution is explained.

Transmission with MIMO was supported in LTE Release 8 with configurations in DL with two or four transmit antennas and two or four receive antennas, which allows for multi-layer transmissions with up to four streams. Multi-User MIMO (MU-MIMO), i.e. allocation of different streams to different

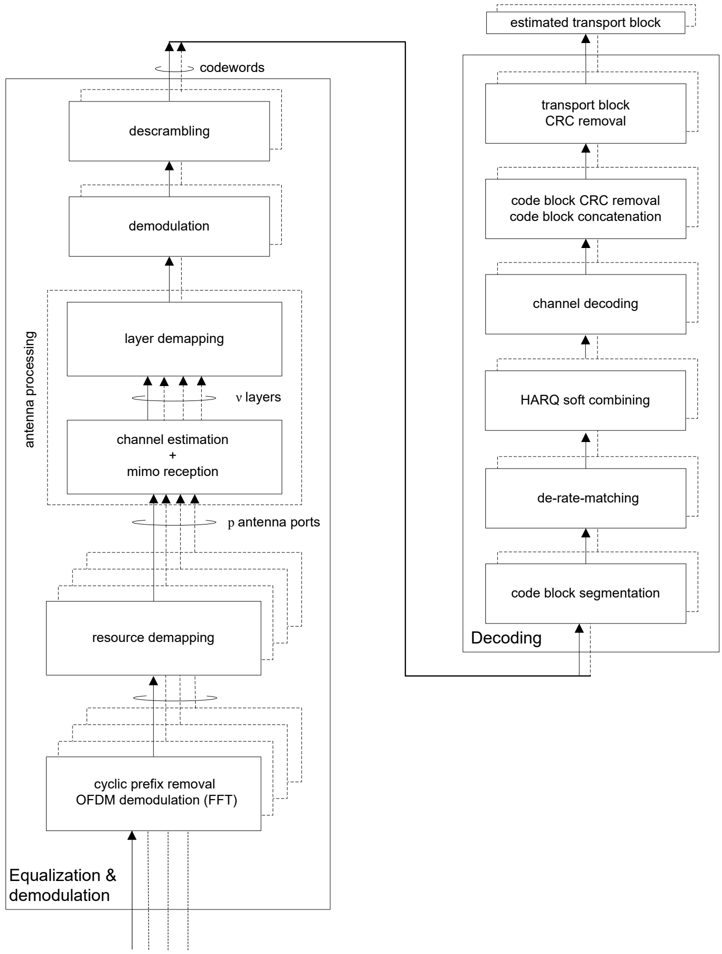


Figure 2.7: PHY layer processing chain for PDSCH reception at the UE

users, is supported in both UL and DL. In order to understand MIMO capabilities, reference signals are of paramount importance. Remember that in LTE Release 8 three types of reference signals were included: cell-specific reference signals, MBSFN reference signals and UE-specific reference signals.

The following transmission schemes were accordingly included:

- **Transmission mode 1 - Single-antenna port** in this mode only one antenna port is used, so Single Input Single Output (SISO) and Single Input Multiple Output (SIMO) schemes can be achieved through this mode.

- **Transmission mode 2 - Transmit diversity** sends replicas of the same information over multiple antennas with different coding and over different frequency resources to make transmission more robust. Specifically, this scheme implements a Space Frequency Block Coding with 2 antenna ports similar to Alamouti scheme. Besides, an extension of this scheme to 4 antenna ports in which only 2 antennas work at the same time (Frequency Switched Transmit Diversity) is also provided.

- **Transmission mode 3 - Large delay cyclic delay diversity** implements an open-loop spatial multiplexing mode specially suited to scenarios where channel information is missing or where it is useless due to the rapid change of channel quality. Precoding is codebook-based, that is, precoders are get from a predefined codebook. In addition to precoding, a delay is supplied to every antenna in order to obtain artificial frequency diversity.

- **Transmission mode 4 - Closed-loop spatial multiplexing** is a codebook-based spatial multiplexing scheme in which the UE sends CSI reports to the eNodeB so as to perform link adaptation. CSI reports are calculated using the cell-specific reference signals and consist of indications about the preferred number of layers (rank) and precoder.

- **Transmission mode 5 - Multi-user MIMO** is a scheme that supports MU-MIMO transmission. It is similar to the transmission mode 4 with the difference that each layer must be devoted to a unique user and each user can use only one layer. Concerning the CSI feedback support, it is limited: wideband Channel Quality Indicator (CQI) and subband CQI are allowed but only wideband Precoding Matrix Indicator (PMI). Feedback calculation assumes a Single User MIMO (SU-MIMO) transmission. Concerning associated signalling, a power offset of 3 dB between reference signals and data of multiplexed users can be signalled to indicate multiplexing of users. In practice, this implies that the maximum number of multiplexed users is two. Also, signalling constraints the precoding to be wideband.

- **Transmission mode 6 - Closed-loop spatial multiplexing using a single transmission layer** is equal to mode 4 but only one layer is allowed. Using this mode a special kind of codebook-based beamforming can be implemented: 4 and 16 beampatterns can be obtained with 2 and 4 antennas, respectively.

- **Transmission mode 7 - Single-antenna port (port 5)** uses the UE-specific reference signals transmitted over the antenna port 5. This mode allows the implementation of single-user single-layer beamforming, where data and reference signals are precoded in the same way. Port 5 is said to be a virtual antenna port, since actual configuration may include several antennas but appears to the user as a single antenna. UEs do not send feedback information to the eNodeB to help in the beamforming process. Instead, eNodeB obtains information from uplink signals.

All the previous transmission modes are semi-statically configured. A user configured in a transmission mode can dynamically use its mode and the fall-back transmit diversity mode. Nevertheless, in order to use a different transmission mode it needs to perform a mode reconfiguration.

In LTE Release 9 reference signals were enhanced and UE-specific reference signals were allowed to be used on ports 7 and 8. This was done to enable a new transmission mode:

- **Transmission mode 8 - Dual layer transmission (port 7 and 8)** is an extension of mode 7 that allows dual layer beamforming. UE-specific reference signals from different ports are code division multiplexed. Besides, two different non-orthogonal sequences can be used per antenna port. Therefore, MU-MIMO with up to 4 rank 1 UEs or 2 rank 2 UEs is supported. This mode supports dynamic switching between SU-MIMO and MU-MIMO. In fact, MU-MIMO is transparent to the UE. There is no need to signal power offset of the reference signals, for example. UEs can send PMI feedback (including frequency-selective feedback) to help in the beamforming process.

With the new reference signals in Release 10 (UE-specific reference signals on ports 7 to 14, and CSI reference signals) a new transmission mode was defined:

- **Transmission mode 9 - Up to 8 layer transmission (ports 7 to 14)** is the extension of the mode 8 to up to 8 layers. It implies a SU-MIMO extension to support 8 layers and improved MU-MIMO capabilities. As previous modes, it uses a maximum of two codewords per user. Codeword-to-layer mapping for 5 to 8 layers is new. This mode can be used to transmit data in MBSFN subframes not actually used for MBMS transmissions. This implies a reduction of the relative control channel overhead for data transmission in these subframes. It supports a maximum of 8 layers, but in practice a maximum of 4 multiplexed users is considered, since a higher value would not generally provide better results. MU-MIMO is transparent to the UE as in transmission mode 8. In case of 8 CSI-RS ports, an enhanced spatial feedback is available. As in transmission mode 8, UEs can send PMI feedback (even frequency-selective feedback) to help in the beamforming process.

In Release 11, a new transmission mode related to Coordinated Multi--Point (CoMP) feature was defined:

- **Transmission mode 10 - Up to 8 layer transmission (ports 7 to 14)** is the extension of the mode 9 with CoMP enabling functionalities. In this mode, it is possible to configure multiple non-zero-power CSI-RS resources, zero power CSI-RS resources, and interference measurement Channel State Information Interference Measurement (CSI-IM) resources for a UE. CSI-IM are similar to zero power CSI-RS resources, because the serving cell does not use them to transmit reference signals, but are used by the UE to estimate the interference coming from other cells. Using combinations of these resources the UE can hypothesise about the quality of its channel depending on the activity of its interferers. The UE is able to feedback multiple CSIs, each one corresponding to an interfering hypothesis. Moreover, this knowledge can be shared among multiple transmission points.

2.4 MAC and RRM

Concerning the MAC layer, the description of its characteristics focuses on three specific characteristics: management of HARQ, LA and scheduling. These are the features with more impact on this Thesis.

2.4.1 HARQ

HARQ with soft combining is a technique that deals with retransmission of data in case of errors. In an Automatic Repeat-reQuest (ARQ) scheme, the receiver uses an error-detecting code to check if the decoded packet contains errors or not. The transmitter is informed by a Negative Acknowledgement (NACK) or Positive Acknowledgement (ACK), respectively. In case of a NACK, the packet is retransmitted. Since ARQ is a stop-and-wait protocol, it can be more efficient to have multiple ARQ processes in parallel. HARQ consists of a combination of ARQ and a FEC. The concept of soft combining means that when a packet is erroneously received it is stored in a buffer and later combined with the retransmission(s) to obtain a single packet that is more reliable than its constituents.

In LTE Advanced (LTE-Advanced), full incremental redundancy is applied, which means that the retransmitted packets are typically not identical with the first transmission but carry complementary information. Besides, in LTE-Advanced, there is a HARQ entity in the UE's MAC that manages multiple parallel HARQ processes to implement a multi-channel stop-and-wait HARQ protocol. The eNodeB has as many HARQ entities in its MAC as users being served by this eNodeB. Then, each HARQ entity in the eNodeB has a paired entity in the UE.

HARQ soft-combining is performed at PHY, nevertheless HARQ management is performed in MAC. That is, at MAC level, the eNodeB knows the state of each HARQ process and decides on new transmissions or retransmissions.

2.4.2 Link Adaptation

One of the key techniques of LTE is the LA that allows the transmitter to adapt the transmission format to the channel quality variations. LA algorithms are not included in the LTE-Advanced specifications. However, specifications indicate which transmission parameters can be adapted, how can they be adapted and what type of physical measurements can be performed to help LA decisions. Each developer can use the standard to develop its own LA algorithms. In this subsection the most important features of LTE related to LA are highlighted.

Concerning the transmission format, the most important characteristics that can be adapted in LTE are the multi-antenna scheme, Modulation and Coding Scheme (MCS), transmission rank and the precoder:

- The multi-antenna scheme can be dynamically changed to get the most of the channel.
- The transmission rank represents the number of independent streams or layers in LTE terminology transmitted in MIMO modes. It is worth noting that the maximum number of independent codewords that can be transmitted is two. That is to say, in case of transmitting more than two layers, then a codeword is transmitted over multiple layers.
- The set of MCS that can be used in a transmission depends on the resource block allocation. Supported modulations are QPSK, 16-QAM and 64-QAM. Given a resource block allocation, a limited set of combinations of modulation and transport block size is available. Each one of these combinations provides a different effective coding rate. In multi-codeword transmission, each codeword can use a different MCS. However, each codeword must use the same MCS in all its allocated resource blocks.

- The precoder in LTE can be a codeword from a predefined codebook or calculated according to proprietary algorithms. The precoder can be wideband or frequency-selective. That is to say, both the same or different precoders can be used in the multiple resources allocate to a user.

Concerning how the LA obtains the knowledge about the channel state, in LTE three mechanisms are used to obtain channel state information at the transmitter side:

- Implicit feedback of channel state is the most common approach in LTE. Following this approach, the UE performs Signal to Interference plus Noise Ratio (SINR) measurements and calculates the most suitable format for transmission. This information is conveyed in a set of reports known as CQI, PMI and Rank Indicator (RI). The RI is related to the number of allowed spatial layers. The PMI indicates the best precoder that can be used from a predefined codebook. Finally, the CQI is a subjective measure of the channel quality that indicates the most efficient MCS that can be used with a Block Error Rate (BLER) lower than 10%.

- Direct estimation through uplink sounding exploiting channel reciprocity is another option. DL channel state can be estimated directly by the eNodeB through measurements of the UL channel, instead of reporting the channel state from the UE to the eNodeB. For example, through a processing of UL measures based on channel reciprocity, the channel matrix or channel covariance matrix could be estimated. In TDD, accurate DL CSI is simple to obtain. However, this technique may provide poorer results in FDD.

- Explicit feedback of channel estimates measured at the UE has been considered in LTE as a possibility for some study items, but it has not been included in the specifications. Feedback may be representative of the whole system bandwidth (wideband feedback) or can be reported per frequency subband (subband feedback). Also, feedback can represent an instantaneous measure or the average over a time interval. Some channel characteristics that can be reported are the channel matrix or its main eigencomponents, and the transmit channel covariance or its main eigencomponents.

Based on the information about the channel gained through any of the above mentioned methods, the eNodeB is able to make a decision on the transmission parameters.

2.4.3 Scheduling

Scheduling is one of the main functions of the Radio Resource Management (RRM) and is carried at the eNodeB by an entity that is usually referred to as packet scheduler. Scheduling decides how to allocate resources to the users in each subframe, that is, at a fast pace. The resources to allocate are spread over frequency, time, space and power domains. In LTE this means resource blocks, layers and power.

The scheduling process does not only select the resources devoted to each user. It also decides if the user is going to perform a new transmission or a retransmission and even which HARQ process will carry the transmission. Furthermore, scheduling selects the transmission format of each user. In order to make optimal decisions, scheduling may consult the LA module. Thanks to this interaction the scheduler can know, for example, the data rate that a user can achieve for a resource allocation, and the corresponding transmission format. This format decision capability makes the packet scheduler to be a part of the LA process, let us say an executive part.

Scheduling algorithms are not included in the LTE specifications. Three are the most common schedulers. First, the Round Robin algorithm, which allocates resources equally among the users. Second, the MaxCQI (or maximum throughput) algorithm, which allocates each resource to the user with best quality. Third, the Proportional Fair algorithm, which allocates the resources to the users with a priority that is directly proportional to the channel quality of this user but inversely proportional to the throughput experienced in the past by this user. However, it is worth noting that the world of scheduling algorithms is much wider than this three typical algorithms.

The specific scheduling implementation in a real system must take into account the QoS required by each user, and by each one of its traffic flows. With this aim, the scheduler is up-to-date with the status of the RLC buffers for each user and traffic flow. The capability of deciding about retransmissions is also very important to satisfy QoS requirements. For example, if a user needs low delay in its packets it can be interesting to prioritize retransmissions over new transmissions. However, if the user prefers a high data rate the contrary policy could be desirable. Decision on transport format may also depend on QoS requirements. Hence, users with low packet error requirements may be assigned a robust MCS instead of a more efficient one.

A good survey about scheduling in LTE technology can be found in [44]. In chapter 5, typical implementations of the scheduler will be detailed.

2.4.4 General view

Figure 2.8 summarises the description of the MAC layer. It is clear that there is a close interaction among key processes of the MAC layer being the scheduler the centre of such interaction.

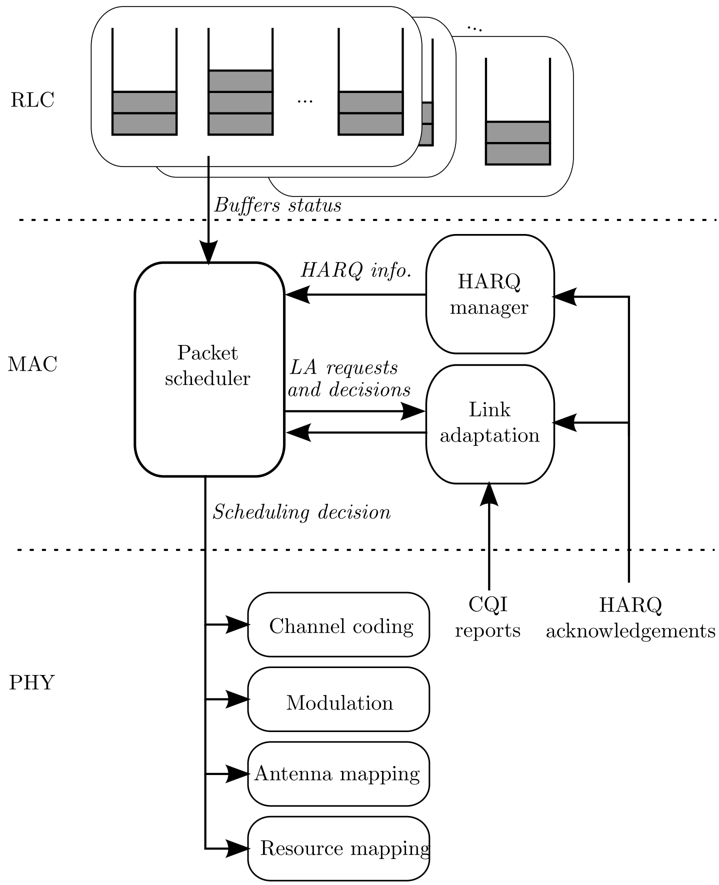


Figure 2.8: Main features of MAC and interaction with other layers

2.5 Additional features

In this section some additional features of LTE and LTE-Advanced are briefly presented, in order to complete the whole picture of the LTE system. Although they are not specifically treated in this Thesis, some comments about them will be find hereinafter. For more information on these topics, the reader is referred to general books such as those from Sesia [70] or Dahlman [71], and to the Third Generation Partnership Project (3GPP) reports and specifications that can be found in [73].

2.5.1 Additional LTE features

First, additional features of Releases 8 and 9 are presented:

- MBMS

  MBMS is a special mode of operation of the LTE system that supports multicast and broadcast services, i.e., the transmission of the same information from one or multiple cells to multiple users. Transmission from multiple cells is supported through the creation of SFN areas, known as MBSFN in LTE. Nevertheless, single cell operation is also allowed, meaning that each cell multicast or broadcast different services. In Release 9 lower layer aspects for MBMS are included in the specifications, while higher layer aspects were postponed to Release 10.

- Inter-Cell Interference Coordination (ICIC)

  ICIC was introduced in LTE Release 8. Specifically, the specifications include a message, called ”Load Information”, that can be sent between eNodeBs through X2 interface. This message informs about: (i) UL interference level per RB (known as overload indicator); UL RBs that are allocated to cell edge UEs, and hence are sensitive to UL interference (known as high interference indicator); and if DL transmission power is higher or lower than a set threshold value (known as relative narrowband transmit power indicator). It is not specified what the receiving eNodeBs should do with this information.

- Self-Optimized Network (SON)

  The term SON refers to three capabilities of LTE networks: self-configuration, self-optimisation and self-healing. The current standard only supports the first two capabilities. Self-configuration allows newly deployed nodes to be automatically configured for system operation. Some functions that fall into this group of capabilities are the configuration of IP address, downloading of eNodeB software, and neighbour list configuration. Self-optimization process allows the auto-tuning of the network based on measurements done by UEs and eNodeBs. Mobility load balancing and mobility robustness optimisation are some features included in self-optimisation capability. Note that SON is currently being extended in LTE-Advanced.

2.5.2 Additional LTE-Advanced features

In this section some additional features of LTE-Advanced, that is, Release 10 and beyond, are briefly presented. The description of CoMP has been intentionally postponed to chapter 6, where a detailed description of the status of CoMP specification in 3GPP is provided. Note also that MIMO enhancements of LTE-Advanced have been already presented in section 2.3.

- Carrier aggregation

  Carrier aggregation allows any capable eNodeB or UE to perform transmissions simultaneously over multiple LTE carriers. Each aggregated carrier is known as component carrier. In order to ensure compatibility with previous Releases, aggregation is based on Release 8/9 carriers. Component carriers can be contiguous or non-contiguous, and in this second case, the aggregated carriers can be in the same frequency band (intra-band) or in different bands (inter-band). It is usually stated that a maximum of five component carriers can be aggregated and, consequently, the maximum aggregated bandwidth is 100 MHz. However, this is not exactly true. In fact, in Release 10 it is possible to aggregate only two component carriers in DL and UL carrier aggregation is not possible. In Release 11, aggregation of two carriers is allowed in UL, while release 12 supports aggregation of up to 3 carriers in DL. Moreover, each release presents limitations to the type of aggregation allowed concerning the contiguity of carriers, and whether they are in the same band or not.

- Heterogeneous networks and Home eNodeB

  The term heterogeneous networks in LTE-Advanced refers to the coexistence of macro-, micro-, pico- and femto-cells in the same network, that is to say, cells with different transmission power and some peculiarities. Several features have been added to the LTE specifications to mitigate a number of interference problems originated from the heterogeneity of the networks. It is especially relevant the introduction of the Home eNodeB in the specifications. It was introduced in Release 9 as a low power eNodeB used in indoor scenarios to provide services to closed subscriber groups, such as the employees of an office. This kind of new nodes makes difficult to manage the interference in a system due to its private owning and usual lack of coordination with macro-cells. Moreover, some architectural changes have been made in the radio access network to facilitate its management.

- Evolution of ICIC

  LTE Release 10 included enhanced ICIC (eICIC) that defines a new kind of subframes called Almost Blank Subframes (ABSs). In ABSs only reduced-power control channels and cell-specific reference signals are transmitted. In heterogeneous networks, high-power nodes may use ABSs periodically to facilitate the communication between UEs and low-power nodes. LTE Release 11 includes and evolution of ICIC, termed further enhanced further enhanced ICIC (feICIC), that deals with interference cancellation for control signals in the UE.

- Relaying

  Relay nodes are a special type of low-power cells used to provide enhanced coverage and capacity at cell edges, hot-spots, and remote areas. The relay node is connected to a cell (known as Donor eNodeB) via the air interface. Type 1 relays use the same frequency in the relay-to-cell link (backhaul link) and in the link to the users (access link), while in Type 1a relaying different frequencies are used.

2.6 Conclusions

This chapter has performed a detailed description of the main characteristics of LTE and LTE-Advanced that presents a system with high complexity and with multiple interrelated elements.

The first consequence of this fact is that a holistic evaluation of the system is not feasible. Because of that, this Thesis is specifically focused on the downlink of the system and its FDD mode. And to be more precise, on a set of RRM functions of the MAC layer.

The second consequence is that the huge complexity of the system prompts the use of computer simulations as the primary research methodology as will be indicated in the next chapter. An analytical approach to the studies conducted in this Thesis would be very complex or impossible.

Finally, the high complexity of the system facilitates the adaptation to different scenarios. It is worth noting that the fact of narrowing the set of functions of the system to be studied, and the number of configurations used (for instance the use of only linear receivers) means that the evaluations conducted in this Thesis be valid only for the conditions of the study. This fact has been taken into account in the selection of the scenarios, algorithms, and configurations chosen that are the most commonly used in the literature.

Chapter 3

Methodology and System Modelling

This Thesis focuses on the study of the Long Term Evolution (LTE) radio interface and its evolution. In order to perform any assessment, it is necessary to have a deep knowledge of the system under study and to define an evaluation methodology. This chapter deals with the second issue.

The definition of the evaluation methodology involves the selection of evaluation objectives, conditions (scenario, system configuration, etc.), methods, procedures, metrics and also the definition of assumptions and system models.

Concerning the system modelling, the key point is to find a good trade off between minimization of modelling complexity and maximization of evaluation accuracy. This is usually accomplished by a top-bottom strategy in which the system is first modelled with high precision and simplified afterwards. Simplifications are allowed as long as they do not compromise accuracy. These considerations reinforce the requirement to have a deep knowledge of the system so as to model it properly.

The chapter has been divided into the next sections:

- Section 3.1 explains the fundamentals of the evaluation methodology followed in this Thesis. Evaluation objectives, conditions and methods are detailed. The explanation specially focuses on simulation and its performance based on the division of the problem into two levels of abstraction known as link and system levels.

- Section 3.2 outlines those system level models and assumptions more related to the scenario where the system is tested than to the system itself. This models could be used with different wireless systems on top.

- Section 3.3 presents system level models and assumptions more closely related to LTE-specific performance evaluation.
- Section 3.4 details a set of contributions of this Thesis to the field of LTE modelling.
- Section 3.5 outlines the main conclusions of this chapter.

3.1 Fundamentals of evaluation methodology

System performance can be evaluated from different points of view. Therefore, it is necessary to clarify which is the perspective of the evaluation conducted in this Thesis in order to understand the proposed methodology.

3.1.1 Evaluation objectives

In this Thesis, LTE air interface is evaluated under both optimal and normal operation conditions. Concerning optimal conditions, it is analysed the peak data rate, that is to say, the highest data rate that can be achieved by a user consuming all the system resources and experiencing perfect channel conditions. Although this is an interesting metric, the focus of this assessment is on the performance evaluation under normal operation conditions. One part of this evaluation concerns the performance analysis of a unique link between a User Equipment (UE) and an evolved Node B (eNodeB). However, special attention is paid to the performance analysis of a system with multiple UEs and eNodeBs. In this scenario it is important to obtain metrics related to the eNodeBs capacity but also to the UEs experienced Quality of Service (QoS). Two metrics that reflect both perspectives are the cell spectral efficiency and the cell edge user spectral efficiency. The first one refers to the mean spectral efficiency experienced by the eNodeBs, which is proportional to the mean spectral efficiency experienced by the UEs, while the second one refers to the lower spectral efficiency percentiles. Evaluation metrics are defined in detail in subsection 3.1.8.

3.1.2 Evaluation methods

In this Thesis two evaluation methods are used: analytical evaluation and simulation. Analytical evaluation consists on the exact calculation of a performance indicator value. This method can be used to evaluate the peak data rate.

Although mathematical models are used whenever possible, given the enormous complexity of current and future wireless systems, it is impossible to fully evaluate their performance using only analytical methods. For this reason, system modelling and computer simulations represent a good alternative for the assessment of these systems, achieving a good trade-off between complexity, cost, time of development and accuracy. Simulation-based evaluation is applied to evaluate numerical features too complicated or even impossible to be exactly calculated. In this Thesis, this method is widely used to evaluate the performance over normal operating conditions.

Table 3.1 presents the relation between the performance indicators and evaluation methods used in this Thesis.

Table 3.1: Performance indicators and evaluation methods.

Performance indicator	**Evaluation method**
Physical layer peak data rate	Analysis
Single link spectral efficiency	Simulation
Cell spectral efficiency	
Cell edge user spectral efficiency	

3.1.3 Link and system level simulation

The simulation of a small system can be performed with a holistic system simulator, that emulates features of all the system layers, at an affordable computational cost. Unfortunately, as the number of links grows such approach results in a prohibitive complexity. System simulation is usually divided into two stages or levels of abstraction known as link level and system level, so as to reduce simulation complexity. That is to say, instead of performing holistic simulations it is used a divide and conquer approach.

At the link level a unique radio link between a Base Station (BS) and a Mobile Station (MS) is modelled, including in the simulation specific features like synchronization, modulation, channel coding, channel fading, channel estimation, demodulation, Multiple-Input Multiple-Output (MIMO) processing, etc. Link level simulations are used to assess the performance of the physical layer and several Medium Access Control (MAC) aspects directly related to the radio interface, such as the Hybrid ARQ (HARQ) mechanisms.

At system level, system modelling encompasses a set of base stations and all their associated user terminals. At this level the focus is on scheduling, link adaptation and other MAC layer processes. The peculiarity of the system level simulator compared to a holistic system simulator is that many physical layer and several MAC layer features are not emulated but an abstraction is used to predict their effect on link performance. Link abstraction models are configured based on link level simulations. These models predict the performance of a transmission over a unique radio link provided a small set of inputs that characterize the format of such transmission and the state of the link. The transmission format parameters could be the modulation and coding rate, for example. On the other hand, the link state measure could be the experienced Signal to Interference plus Noise Ratio (SINR) and the performance measure could be the experienced data rate.

It is worth noting that link level simulations are not only used to define link abstraction models, but also to assess the performance of several techniques emulated at such level and to optimize them. Concerning optimization, channel estimation and channel coding performance can be evaluated and improved. Furthermore, link level assessments provide interesting conclusions that can be used to define Radio Resource Management (RRM) algorithms at system level.

3.1.4 Evaluation scenarios

In real networks, there are as many operation conditions as deployed networks. Nevertheless, due to obvious reasons, a performance evaluation based on simulation should be restricted to a limited set of scenarios. Concerning the modelling and definition of these scenarios, it is worth noting that the more realistic they are the more significant the performance results will be.

Four test scenarios have been considered in the system level simulations of this Thesis:

- Indoor: an indoor environment targeting isolated cells at offices and/or in hotspots based on stationary and pedestrian users. This test environment is characterized by high user throughput or user density in indoor coverage.
- Micro-cellular: an urban micro-cellular environment with higher user density focusing on pedestrian and slow vehicular users.
- Base coverage urban: an urban macro-cellular environment targeting continuous coverage for pedestrian up to fast vehicular users.
- High-speed: macro-cellular environment with high-speed vehicles and trains.

One deployment scenario has been considered for each test scenario: indoor hotspot scenario for indoor test scenario, urban micro-cell scenario for micro-cellular test scenario, urban macro-cell scenario for base coverage urban and rural macro-cell scenario for high speed. Each deployment scenario

Table 3.2: Overview of deployment scenarios

Parameter	**Deployment scenario**			
	Indoor hotspot	Urban micro-cell	Urban macro-cell	Rural macro-cell
Center Frequency (GHz)	3.4	2.5	2.0	0.8
Inter-site distance (m)	60	200	500	1732
UE speed (km/h)	3	3	30	120
UE distribution	100% indoor	50% indoor	100% outdoor	100% outdoor
eNodeB antenna height (m)	6	10	25	35

is characterized by a deployment configuration. Table 3.2 summarises these configurations. The rest of parameters can be found in [7].

At link level, both scenarios similar to those of system level and a set of customized scenarios have been used. In fact, a simple scenario without fast fading modelling, and only Additive White Gaussian Noise (AWGN) noise has been quite relevant.

3.1.5 Link level simulation procedure

A set of simulation runs is conducted to evaluate the link level performance through simulation. In each simulation run the next steps and assumptions are considered:

1. Simulation is configured according to the parameters of a specific test scenario.

2. A mean Signal to Noise Ratio (SNR) is fixed for the duration of the run, understood as the ratio between the received signal power and the noise power per receiver antenna for a transparent channel that does not change transmitted signal. Given the mean SNR, the noise power (density) per antenna at the receiver side is calculated assuming that signal power is equal to the transmitted signal power.

3. Small scale channel parameters are configured according to any specific scenario and remain fixed during the whole simulation. Large-scale channel variations are not modelled. Small-scale variation must ensure the mean received power to be equal to the mean transmitted power.

4. An active phase is conducted in which transport blocks are generated each subframe with a configured size. Then, these blocks are coded, modu-

lated, and mapped on antennas and on resources. All the parameters involved in this process are configured at the beginning of the simulation. Besides, control information is also sent among simulation entities (one eNodeB and one UE). Multiple statistics are stored during the active phase. In simulations with HARQ, blocks can be generated in each subframe or taken from a retransmission buffer.

5. After a predefined number of transport blocks are transmitted, simulation is ended and performance statistics are collected.

3.1.6 System level simulation procedure

A set of simulation runs or simulation drops is conducted to evaluate the system performance through simulation. In each simulation run the next steps and assumptions are considered:

1. Simulation is configured according to the parameters of a specific deployment scenario.

2. Users are dropped over a predefined area of the network layout. Users position does not change during the simulation although spatial pointing, speed and movement direction are assigned to each mobile. Each mobile corresponds to an active user session that runs for the duration of the drop.

3. Mobiles are randomly assigned Line of Sight (LoS), Non Line of Sight (NLoS) and Outdoor to Indoor (OtoI) channel conditions. The rest of large-scale channel parameters are also randomly assigned to the mobiles. These parameters remain unchanged along the simulation run. Nevertheless, channels experience small-scale variations according to speed and movement direction.

4. Serving cell is assigned to a user based on a cell selection scheme.

5. An active phase is conducted in which packets arrive to the eNodeBs according to any traffic model and are scheduled to the users with an appropriate packet scheduler. Besides, control information is also sent among simulation entities (eNodeBs and UEs). Multiple statistics are stored during the active phase.

6. After a predefined simulation time, simulation is ended and performance statistics are collected. A transient time is defined and statistics coming from this period are neglected.

The process is repeated to perform multiple simulation runs with different values of the random variables involved in the simulation. Therefore, in each simulation run different user locations and channel properties are simulated.

It is worth noting that this kind of simulation is known as semi-static (or semi-dynamic) simulation since large-scale channel parameters are static, but small-scale parameters vary along the simulation run. In opposition, in static simulations these parameters do not change at all, and in fully dynamic simulations both kind of parameters change along the simulation.

In order to summarise the simulation process, a number of simulations, N_{sim}, must be conducted with enough simulation run-times, T_{sim}, collecting statistics once a transient time, ΔT_{sim}, has passed.

3.1.7 Simulation validity

In order to ensure the validity of the simulation results two actions have been taken:

Statistical significance analysis Simulation presents a stochastic behaviour. Therefore, it is necessary to analyse if the results obtained through a simulation campaign are representative enough so as to trust them. That is to say, it is required to be confident that results are not by random chance. Through statistical significance analysis the main variables of the simulation procedure are determined: the number of simulations, the simulation time, etc. Appendix B elaborates on this specific subject.

Calibration In order to validate the results, it is performed a cross-checking of the results obtained in a set of calibration simulations with those obtained by other researchers using the same simulation configurations. This cross-checking ensures the validity of the simulation results for the calibration configurations. The key factor to ensure a more general validity is to define a representative set of calibration configurations. Results concerning the calibration of the system level simulations of this Thesis are shown in Appendix A.

3.1.8 Evaluation criteria

This section describes the most important indicators used for the evaluation of a certain technology.

Link level criteria

At link level, Block Error Rate (BLER) is the most common evaluation metric. BLER is defined as the number of erroneously decoded transport blocks at the receiver divided by the number of received transport blocks. Throughput indicator is also used, defined as the number of correctly received bits per time unit. If the Modulation and Coding Scheme (MCS) is kept constant during the whole simulation, as normal, throughput and BLER are related through the nominal rate given by the MCS, $Rate_{MCS}$, by a simple equation: $Throughput = (1 - BLER) \times Rate_{MCS}$.

These link performance measures are usually represented as a function of mean SNR values.

Peak spectral efficiency

The peak spectral efficiency is the highest theoretical data rate (normalized by bandwidth), which is the received data bits assuming error-free conditions assignable to a single UE, when all available radio resources for the corresponding link direction are utilized (i.e., excluding radio resources that are used for physical layer synchronization, reference signals or pilots, guard bands and guard times).

Cell spectral efficiency

Let us define the layer 2 user throughput, as the number of correctly received bits, i.e. the number of bits contained in the Service Data Unit (SDU) delivered to layer 3, over a certain period of time.

Cell spectral efficiency is defined as the aggregate layer 2 throughput of all users normalized by the channel bandwidth and the number of cells. Cell spectral efficiency has units of b/s/Hz/cell. In a system with N users and M cells, the overall efficiency of the cell is expressed as:

$$\eta = \frac{\sum_{i=1}^{N} \chi_i}{T \cdot \omega \cdot M}, \tag{3.1}$$

where χ_i is the number of bits correctly received by the i-th user, T is the considered time interval and ω the channel bandwidth.

In each simulation drop an independent cell spectral efficiency value is obtained, then these values are averaged to obtain a single cell spectral efficiency value.

Cell edge user spectral efficiency

Let us define the user spectral efficiency as the user throughput normalized by the channel bandwidth. So defined, the spectral efficiency of user i-th (γ_i) is calculated as:

$$\gamma_i = \frac{\chi_i}{T \cdot \omega} \tag{3.2}$$

Cell edge user spectral efficiency is defined as the 5th percentile of user spectral efficiency, that is, the value over which 95% of user spectral efficiencies fall.

In order to obtain the cell edge user spectral efficiency after multiple simulations, the whole set of simulated users is considered to obtain the 5th percentile of user spectral efficiency instead of averaging the cell edge user spectral efficiencies of each simulation. That is, after N_{sim} simulations runs with K users in each simulation, a set of user spectral efficiencies γ_i with $i \in [1, N]$ and where $N = N_{sim} \cdot K$ is obtained. Being F defined as de Cumulative Distribution Function (CDF) of γ, the cell edge user spectral efficiency, β, is defined as:

$$\beta = F^{-1}(0.05). \tag{3.3}$$

In the Thesis, the user spectral efficiency CDF is estimated by an empirical CDF, $\hat{F}$, defined as:

$$\hat{F}(t) = \frac{1}{N} \sum_{i=1}^{N} \mathbf{1}_{\gamma_i \leq t}, \tag{3.4}$$

where $\mathbf{1}_{\gamma_i \leq t}$, the indicator of event $\gamma_i \leq t$, is 1 when $\gamma_i \leq t$ and 0 otherwise.

Using the empirical CDF, the estimated cell edge user spectral efficiency is calculated as the 5-th percentile of $\hat{F}$:

$$\hat{\beta} = \hat{F}^{-1}(0.05). \tag{3.5}$$

3.2 Technology independent system level models and assumptions

This section presents a set of models and assumptions that are not specific of LTE, such as positioning of base stations and mobile stations, channel modelling, interference modelling, wrap around as a means to avoid border effects, and traffic modelling.

3.2.1 Network layout

One of the key aspects of a system level model is how the multiple eNodeBs are distributed. Two eNodeB layouts are considered in the simulations of this Thesis. The first one is an hexagonal layout with eNodeBs placed in a regular grid as that shown in Figure 3.1. The network consists of 19 sites, each one with 3 cells. Wrap around technique is used to avoid border effects (see Section 3.2.5). The second one is the layout used for indoor hotspot scenario that consists of one floor of a building. The floor contains 16 rooms of 15 m × 15 m and a long hall of 120 m × 20 m. Two sites are placed in the middle of the hall at 30 m and 90 m with respect to the left side of the building. This layout is depicted in Figure 3.2

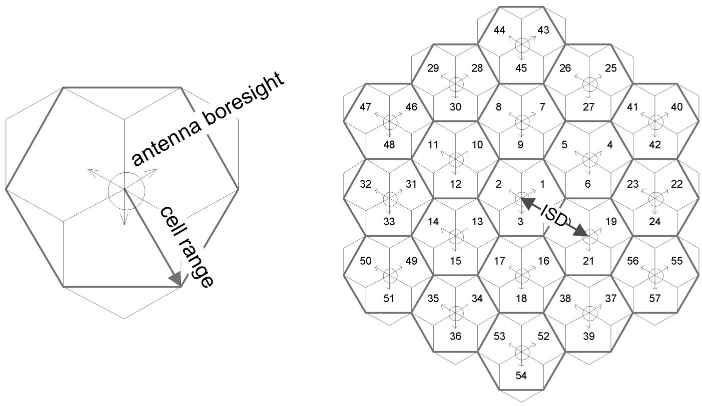


Figure 3.1: Layout for outdoor scenarios [7].

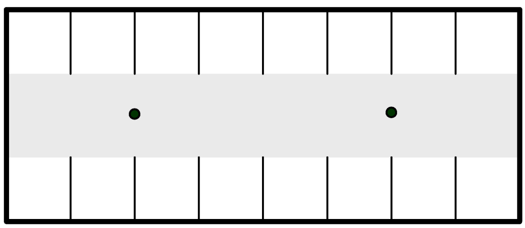


Figure 3.2: Layout for indoor scenario [7].

3.2.2 User distribution

Besides knowing eNodeBs location, it is also important how the UEs are distributed. In LTE assessments, a common approach to model user distribution is to fix a total number of users and drop them independently with uniform distribution over the network layout [7]. More precisely, in most cases what is fixed is the number of users per cell, U, then the total number of users is calculated from this value as the product $U \cdot C$, being C the number of cells. This approach has been followed in this Thesis unless otherwise stated.

It is worth noting that the fact of specifying the number of users per cell may lead to different interpretations. For example, users could be distributed guaranteeing that exactly U users are served by each cell. Also, in case of a hexagonal cell layout, users could be distributed guaranteeing that exactly U users are located in the hexagonal area of each cell. These different interpretations lead to different user distributions that affects system performance results. It is remarkable that these interpretations do not provide independent distributions of users since the position of user depends on the number of users located in/served by other cells.

3.2.3 Channel modelling

Channel modelling is done according to International Mobile Telecommunications Advanced (IMT-Advanced) guidelines in [7]. IMT-Advanced channel model defined by the International Telecommunication Union (ITU) Radiocommunication Sector (ITU-R) is a stochastic and geometric model that allows creating a bidirectional radio channel consisting of a plurality of rays. Although it is a geometric based model that knows the exact location of transmitting and receiving elements, it does not specify the position of the scatterers, rather only ray directions are known.

Figure 3.3 shows a transmitter and receiver and all existing rays between them. Moreover, this figure represents the concept of cluster, or propagation path – in space, time and angle – that consists of a set of rays affected by nearby scatterers. The figure also includes the concept of Angle of Departure (AoD) and Angle of Arrival (AoA), both at cluster and ray level.

There are two different sets of channel parameters in the IMT-Advanced channel model. The first one is related to the large-scale parameters, such as Shadow Fading (SF) and path loss. The second one concerns small-scale parameters, including AoA and AoD or delay of the rays.

In order to generate channel samples between one transmitter and one receiver, mobility and exact location of both ends must be known. Based on this information all large-scale parameters are generated, followed by the small-scale

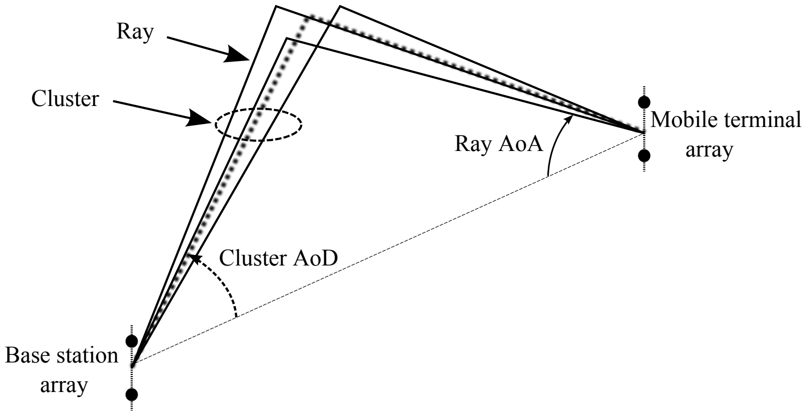


Figure 3.3: IMT-Advanced MIMO channel [74].

parameters. All these parameters are kept constant during the whole simulation. Channel samples are obtained by adding the contributions of all involved rays.

There are five different scenarios with their corresponding channels in [7]: Indoor hotspot (InH), Urban Micro-cell (UMi), Urban Macro-cell (UMa), Suburban Macro-cell (SMa) and Rural Macro-cell (RMa). Different models are characterized by different parameters of the statistical distributions used to generate the channel samples.

3.2.4 Interference modelling

According to [7], in an LTE assessment, fading signal and fading interference must be computed from each eNodeB into each UE. Nevertheless, a very detailed emulation of IMT-Advanced channel models in system level simulators demands high computational capabilities. In scenarios with many cells and users, computational and storing demands of a very precise emulation can be unaffordable.

In order to reduce computational cost, intermediate calculations can be stored. The problem is that storing demands can be also unaffordable. Therefore, for a fixed number of emulated interferers, there is usually a trade-off between computational and storing cost. That is to say, the more intermediate calculations are stored, the lower the computational cost of final calculations.

A complementary measure is to apply some simplification model, reducing computational and storing demands at the expense of reduced accuracy.

One simplification model is proposed in [37], where Skillermark *et al.* proposed to model in detail those channels that send the strongest signals to the user, while modelling of channels from the rest of cells considers only wideband and long-term variations. In [37] a fixed number of strongest interferers is considered and the effect of the simplification is studied. In this Thesis, this model is widely used.

3.2.5 Wrap around

Let us consider a simulation conducted with just the 57 cells of the 19 sites layout presented in subsection 3.2.1. In this simulation, outer cell statistics would differ greatly from the statistics of inner cells due to their different environment. Note that for example, interference experienced in outer cells would be generally lower than that experienced in inner cells.

In order to avoid those border effects, one option is to emulate a larger layout and get the statistics of only the 57 central cells. This option is not very efficient, since many cells are simulated but their statistics are not collected. In this Thesis, a more efficient option has been used in which a continuum of cells is emulated through a wrap around technique. In this technique, exact replicas of the original layout are located around it, and just the original cells are dynamically simulated. Users are dropped in the original layout. In the simulation, each user located in a cell of the original layout considers, and receives signals from, the set of 56 not repeated cells that surrounds this cell.

A complementary explanation of the wrap around technique is found in [26], Appendix G.

3.2.6 Traffic models

In this Thesis two traffic models have been used:

- A full-buffer traffic model (sometimes referred to as a full queue model), in which an infinite amount of data bits awaiting transmission in the transmit buffer associated with each data source is assumed.

- A Constant Bit Rate (CBR) traffic model characterized by a certain packet size and period of time between packet generations.

3.3 Technology dependent system level models and assumptions

This section presents a set of models and assumptions that are more LTE specific such as the models implied in the emulation of the Physical (PHY) and MAC layers. Concerning the PHY, a PHY layer abstraction is used at system level to avoid the complete emulation of all the radio links of a system. This abstraction has two parts. The first part allows the calculation of the SINR experienced by a user in a specific time and previously to the decoder. Therefore, this model takes into account radio propagation from serving and interfering cells and MIMO processing (subsection 3.3.4 is related with this part). The second part allows the translation of SINR values to transmission quality measures (subsection 3.3.2). Concerning the MAC layer, it is simulated in detail in system level simulation, therefore, key features such as Link Adaptation (LA) (subsection 3.3.5), scheduling or cell selection (subsection 3.3.6) are explicitly modelled.

3.3.1 Receiver modelling

In a real LTE system, the receiver takes signal samples from multiple antennas after Orthogonal Frequency Division Multiplexing (OFDM) demodulation and provides inputs to the decoder per each transmitted stream in the form of, for instance, soft inputs through a more or less complicated multi-antenna processing.

In a system level model, multi-antenna processing is simplified in such a way that, given a set of inputs, the receiver model provides just an estimate of the post-processing SINR values obtained at the output of a receiver (per each configured frequency chunk and per each transmitted codeword). The required inputs are the parameters that characterize the signal radiated by each eNodeB (power, number of layers, codewords, precoders, etc.) and the coefficients that represent the channels from each eNodeB transmitting antenna to each UE receiving antenna.

In the remainder of this subsection the mathematical models used for signal transmission and reception in this Thesis are explained. Moreover, receiver models used in this Thesis are presented with the mathematical expressions used to calculate the SINR. It is worth noting that the description focuses on linear receivers since they will be used along the Thesis because of its good trade-off between simplicity and good performance.

It is worth noting that it is assumed that the system is free of inter-symbol interference, that is to say, cyclic prefix is assumed to be longer than the maximum excess delay of the different channel paths. Furthermore, the system is assumed to be free of inter-carrier interference.

We consider $J+1$ eNodeB transmitters with P_j transmitting antennas in the j-th transmitter. And a number of receivers with Q receiving antennas, where $Q \leq P_J$.

Transmitted signal and linear precoding

In this Thesis we only consider transmit linear precoding, in which each eNodeB transmits L_j signal streams being $\mathbf{x}_j \in \mathbb{C}^{L_j \times 1}$ the signal before linear precoding, and $\mathbf{W}_j \in \mathbb{C}^{P_j \times L_j}$ the precoding matrix. The signal vector transmitted from the eNodeB in a specific resource element (identified by an OFDM symbol and a subcarrier in a subframe) is then:

$$\mathbf{s}_j = \mathbf{W}_j \mathbf{x}_j. \tag{3.6}$$

In case of single antenna transmission: $P_j = 1$, $L_j = 1$, $\mathbf{W}_j = 1$, and then:

$$\mathbf{s}_j = \mathbf{x}_j. \tag{3.7}$$

The expressions in the remainder of this section are valid for a specific resource element although, for the sake of clarity, explicit references to this resource are avoided without loss of generality.

Channel

Let us consider $\mathbf{H}_j \in \mathbb{C}^{Q \times P_j}$ as the complex channel matrix between the j-th eNodeB and a specific UE.

Signal received at UE

Let us define $\mathbf{s}_j \in \mathbb{C}^{P_j \times 1}$ as the signal transmitted by j-th eNodeB. The signal vector received at UE, $\mathbf{y} \in \mathbb{C}^{Q \times 1}$, is:

$$\mathbf{y} = \sum_{j=0}^{J} \mathbf{H}_j \mathbf{s}_j + \mathbf{n}, \tag{3.8}$$

where $\mathbf{n} \in \mathbb{C}^{Q \times 1}$ is the additive white complex Gaussian received noise, whose covariance matrix, $\mathbf{R_n}$, is:

$$\mathbf{R_n} = E\{\mathbf{n}\mathbf{n}^{\mathrm{H}}\} = \sigma_{\mathbf{n}}^2 \mathbf{I}_Q, \tag{3.9}$$

where $\mathbf{I}_Q \in \mathbb{C}^{Q \times Q}$ is the identity matrix and $\sigma_{\mathbf{n}}^2$ is the noise variance.

In case of a linear transmitter, the signal received at the UE is then:

$$\mathbf{y} = \sum_{j=0}^{J} \mathbf{H}_j \mathbf{W}_j \mathbf{x}_j + \mathbf{n}. \tag{3.10}$$

The received signal consist of a linear combination of useful signal, the signal coming from a specific cell and layer, interference, and noise.

$$\mathbf{y} = \mathbf{u} + \mathbf{i} + \mathbf{n}. \tag{3.11}$$

Let us consider that the useful signal, $\mathbf{u}$, is the signal that comes from layer ν and cell 0:

$$\mathbf{u} = \mathbf{H}_0 [\mathbf{W}_0]_{\nu:} [\mathbf{x}_0]_{\nu}, \tag{3.12}$$

Then, the interference signal, $\mathbf{i}$ is:

$$\mathbf{i} = \sum_{j=1}^{J} \mathbf{H}_j \mathbf{W}_j \mathbf{x}_j + \sum_{l=1, l\neq \nu}^{L_0} \mathbf{H}_0 [\mathbf{W}_0]_{\nu:} [\mathbf{x}_0]_{l}, \tag{3.13}$$

where the first term of the equality represents the interference coming from cells different to the serving one (cell 0), and the second term represents the interference from streams of the serving cell different to the desired stream.

These signal components can be characterized through a set of covariance matrices. The useful signal covariance matrix is:

$$\mathbf{R}_{\mathbf{u}}^{(\nu)} = \sigma_{\mathbf{x}}^2 \mathbf{H}_0 [\mathbf{W}_0]_{\nu:} [\mathbf{W}_0]_{\nu:}^H \mathbf{H}_0^H, \tag{3.14}$$

where $\sigma_{\mathbf{x}}^2 = E\left\{[\mathbf{x}_j]_{\nu} [\mathbf{x}_j]_{\nu}^H\right\}$ is assumed to be independent of j and ν.

The interference covariance matrix is:

$$\mathbf{R}_{\mathbf{i}}^{(\nu)} = \sigma_{\mathbf{x}}^2 \sum_{j=1}^{J} \mathbf{H}_j \mathbf{W}_j \mathbf{W}_j^H \mathbf{H} j^H + \sigma_{\mathbf{x}}^2 \sum_{l=1, l\neq \nu}^{L_0} \mathbf{H}_0 [\mathbf{W}_0]_{l:} [\mathbf{W}_0]_{l:}^H \mathbf{H}_0^H. \tag{3.15}$$

Additionally, a noise plus interference covariance matrix can be defined as:

$$\mathbf{R}_{\mathbf{in}}^{(\nu)} = \mathbf{R}_{\mathbf{i}}^{(\nu)} + \mathbf{R}_{\mathbf{n}}^{(\nu)}. \tag{3.16}$$

The superscript (ν) of the covariance matrices indicates that those matrices are calculated for a desired signal on stream ν.

Channel estimation

Let us define $\widehat{\mathbf{H}}_j \in \mathbb{C}^{Q \times P_j}$ as the channel estimate of $\mathbf{H}_j$:

$$\widehat{\mathbf{H}}_j = \mathbf{H}_j + \mathbf{E}, \tag{3.17}$$

where $\mathbf{E} \in \mathbb{C}^{Q \times P_j}$ is the estimation error.

Linear receiver

Let us assume that the receiver aims to obtain an estimation of $\mathbf{x}_0$, $\widehat{\mathbf{x}}_0$. Only linear receivers are used in this Thesis, in which the estimation is obtained through the multiplication of the received signal, $\mathbf{y}$, by a reception matrix, $\mathbf{Z} \in \mathbb{C}^{L_0 \times Q}$:

$$\widehat{\mathbf{x}}_0 = \mathbf{Z}\mathbf{y}. \tag{3.18}$$

The idea is that the receiver matrix produces a combination of input signal samples to obtain an estimate of the transmitted signal.

The estimated signal for the ν-th stream is:

$$[\widehat{\mathbf{x}}_0]_\nu = [\mathbf{Z}]_{\nu:}\mathbf{y}. \tag{3.19}$$

Using the covariance matrices previously defined, the post-receiver SINR for the stream ν, $\gamma^{(\nu)}$, can be obtained as:

$$\gamma^{(\nu)} = \frac{[\mathbf{Z}]_{\nu:}\mathbf{R}_{\mathbf{u}}^{(\nu)}[\mathbf{Z}]_{\nu:}{}^H}{[\mathbf{Z}]_{\nu:}\mathbf{R}_{\mathbf{in}}^{(\nu)}[\mathbf{Z}]_{\nu:}{}^H}. \tag{3.20}$$

The exact expression of $[\mathbf{Z}]_{\nu:}$ depends on the specific type of receiver being used. Next, several receivers are presented.

MRC receiver

In this Thesis, we use a Maximum Ratio Combining (MRC) receiver for single antenna transmission. In a MRC receiver, the receiver matrix is defined as:

$$\mathbf{Z} = \alpha\widehat{\mathbf{H}}_0^H \widehat{\mathbf{R}}_{\mathbf{n}}^{-1}, \tag{3.21}$$

where $\widehat{\mathbf{R}}_{\mathbf{n}}$ is an estimation of $\mathbf{R}_{\mathbf{n}}$, and α is a constant (and as such it does not affect the SINR estimation) that can be configured to obtain an unbiased estimation of the desired signal.

MMSE receiver

In this Thesis, the Minimum Mean Square Error (MMSE) receiver, also known as Interference Rejection Combining (IRC), is used with spatial multiplexing. In order to reject interference, it uses the whole interference covariance matrix. That is, it is assumed the knowledge of the spatial interference colouring. In the MMSE receiver, the ν row of $\mathbf{Z}$ is calculated as:

$$[\mathbf{Z}]_{\nu:} = \alpha(\widehat{\mathbf{H}}_0[\mathbf{W}_0]_{:\nu})^H \left(\widehat{\mathbf{R}}_{\mathbf{in}}^{(\nu)}\right)^{-1}, \tag{3.22}$$

where $\widehat{\mathbf{R}}_{\mathbf{in}}^{(\nu)}$ is an estimation of $\mathbf{R}_{\mathbf{in}}^{(\nu)}$, and α is a constant (and as such it does not affect the SINR estimation) that can be configured to obtain an unbiased estimation of the desired signal.

Using (3.22) in (3.20), and assuming $\widehat{\mathbf{R}}_{\mathbf{in}}^{(\nu)} = \mathbf{R}_{\mathbf{in}}^{(\nu)}$ (i.e. no error in interference plus noise covariance estimation), SINR calculation can be simplified to [75]:

$$\gamma^{(\nu)} = \sigma_{\mathbf{x}}^2 \left(\mathbf{H}_0[\mathbf{W}_0]_{:\nu}\right)^H \mathbf{R}_{\mathbf{in}}^{(\nu)^{-1}} \left(\mathbf{H}_0[\mathbf{W}_0]_{:\nu}\right). \tag{3.23}$$

3.3.2 Link abstraction models

Link abstraction models, also known as Link to System (L2S) mappings, are developed through link level simulations to provide to the system level an abstraction of the complex processes that occur at the air interface. In a system level simulator, link abstraction models are used to translate channel quality measures, such as SINR values, to transmission quality values, such as BLER or throughput.

In the typical approach to the system modelling of Universal Mobile Telecommunications System (UMTS), in system level simulations the wireless channel is modelled considering only the mean propagation loss and large-scale fading (also known as shadowing) while fast fading is accounted at the link level. This approach is valid in this scenario due to the way in which a UMTS transmission is spread over the whole bandwidth, which provokes a large correlation between the mean SINR of one radio block and its correct reception. Therefore, in this scenario a simple look up table for each modulation and coding scheme is needed to translate the instantaneous SINR to a Bit Error Rate (BER) or BLER value.

However, IMT-Advanced technologies are based on OFDM, which is a multi-carrier modulation. In such systems, it has been proved that the mean SINR experienced over the set of physical resources used by a transmission is not a good measure of the channel quality to be translated to transmission quality. Instead, it has been demonstrated that the behaviour of Forward Error Correction (FEC) techniques depends highly on how different instantaneous channel quality levels affect different coded bits. Therefore, it becomes crucial to model the behaviour of the channel in the frequency domain over both short and long time scales and new L2S models must be used in order to compress the information from a set of instantaneous channel states (that can be obtained at bit, symbol or Resource Block (RB) level) to one or two specific measures used for mapping link level data.

Although different proposals can be found in the literature (see, e.g., [38]), most common approaches are based on the Effective SINR Mapping (ESM) concept. ESM maps an instantaneous set of SINR samples, γ_n, into a scalar value. This scalar value is called effective SINR, γ_{eff}, and is calculated as:

$$\gamma_{eff} = \alpha_1 \Phi^{-1} \left(\frac{1}{N} \sum_{n=1}^{N} \Phi \left(\frac{\gamma_n}{\alpha_2} \right) \right) \tag{3.24}$$

being N the number of SINR samples, $\Phi(x)$ an information measure function, $\Phi^{-1}(x)$ its inverse, and α_1 and α_2 two configurable parameters. Once the effective SINR value, γ_{eff}, has been calculated, a Look-Up Table (LUT) obtained through simulations conducted in an AWGN scenario is used to translate the effective SINR value to, for instance, a BLER value. With this aim, an AWGN look up table must be obtained for each MCS.

For each MCS, α_1 and α_2 must be calibrated through link level simulations to minimize the error between real BLER (obtained through simulation) and predicted BLER (obtained through the ESM). If the model was perfect, the γ_{eff} would be the scalar SINR value that in an AWGN scenario would produce the same BLER obtained in the multi-carrier scenario with the measured SINR vector.

The most common ESM model in IMT-Advanced evaluation is the Mutual Information Effective SINR Mapping (MIESM). This model uses as its information measure the mutual information [76]. The mutual information corresponding to a SINR value is not obtained through a closed form expression, a look up table obtained via numerical methods is used instead. Additionally, it is worth noting that mutual information for a SINR value depends on the order of the modulation, the modulation constellation and the coding scheme. Then, in LTE three look up tables must be obtained, one for each allowed modulation (QPSK, 16QAM and 64QAM).

Brueninghaus [38] conducted a study using a system similar to LTE that demonstrated that MIESM method can provide in general better results than another widely used method known as Exponential ESM (EESM) method. Based on the results of this study, it can be obtained an important conclusion: EESM fitting factors are quite different for each transmission mode while MIESM fitting factors are close to 1. Therefore, EESM requires a calibration to operate while MIESM can be used even with non-calibrated fitting factors, providing successful results.

In this Thesis, MIESM is used as the link abstraction model. According to the findings in [38], configurable parameters α_1 and α_2 have been chosen to be equal to 1. It is worth noting that, in this Thesis, SINR samples are obtained with a granularity equal to one RB, that is to say, each time SINR is measured over a frequency bandwidth, as many SINR samples are obtained as RBs include this frequency bandwidth. On the other hand, the time domain granularity of SINR measurements is equal to one subframe, that is, 1 ms.

This granularity is a trade-off between accuracy and complexity. It is worth noting that SINR measures with a granularity of one subcarrier and one OFDM symbol could be also obtained, but it would result in a high computational cost when using link abstraction. These simplifications do not affect the validity of results as calibration data will show.

To conclude, it is worth noting that in LTE one or two different Transport Blocks (TBs), corresponding to the same number of codewords, can be simultaneously transmitted to a specific user. Besides, each TB can be transmitted over 1 or 2 physical streams. Therefore, a more LTE specific formula for the calculation of the effective SINR for a specific codeword, $\gamma_{eff}^{(c)}$, is used in this Thesis:

$$\gamma_{eff}^{(c)} = I^{-1}\left(\frac{1}{N_{RB} \times |V^{(c)}|} \sum_{\nu \in V^{(c)}} \sum_{n=1}^{N_{RB}} I\left(\gamma_n^{(\nu)}\right)\right), \tag{3.25}$$

where c is the index identifying a specific codeword, ν is the index of a specific layer, n is the index of a specific RB, $V^{(c)}$ is the set of layers over which codeword c is transmitted, and $|V^{(c)}|$ is the number of layers in this set. Additionally, I is the mutual information, I^{-1} is its inverse, N_{RB} is the span of the measurement in number of RBs, and $\gamma_n^{(\nu)}$ is the SINR value over layer ν in RB n.

3.3.3 Decoder model

In a system simulator, each time a TB is received in a UE, an SINR value is calculated per RB, as explained in Subsection 3.3.1. Then, these SINR values are translated to an effective SINR and, finally, to a BLER value, as explained in Subsection 3.3.2.

In order to emulate the effect of such BLER, a random value is drawn from a uniform distribution between 0 and 1, and if this value is lower than the BLER value it is assumed to have a decoding failure. Otherwise, a decoding success is produced. With this method, if this experiment was repeated a high number of times, the failure rate would approach the calculated BLER.

3.3.4 HARQ

In this Thesis, explicit scheduling of HARQ retransmissions is performed instead of using a simplistic model such that used in [44].

Only incremental redundancy is evaluated in this Thesis. Nevertheless, in order to facilitate the understanding of the incremental redundancy model let us start by explaining the modelling of Chase combining.

In Chase combining, the same set of bits is transmitted, therefore retransmissions should use the same transmission format as first transmission and even the same number of resource blocks. Nevertheless, the exact set of allocated resources can be different. In order to model chase combining, each time a TB is erroneously decoded in its first transmission, calculated SINR values per RB of this TB are stored in an array that models the soft information buffer. After each retransmission, the new calculated SINR values per RB are added to the corresponding values in the array. It is worth noting that addition is done in the linear scale. Let us denote γ_i as the SINR in the i-th attempt of transmission of a TB over a specific RB. Let us also denote $\widehat{\gamma}$ as the effective post-processing SINR after N TB transmission attempts over the same RB. This value is calculated in this Thesis as:

$$\widehat{\gamma} = \sum_{i=1}^{N} \gamma_i. \tag{3.26}$$

In incremental redundancy, a modified calculation is used:

$$\widehat{\gamma} = \sum_{i=1}^{N} \gamma_i + IR_{gain}(N, \mathrm{MCS}), \tag{3.27}$$

where $IR_{gain}(N, \mathrm{MCS})$ is a gain factor calculated based on the results of [77] that depends on the number of transmissions, N, and the modulation and coding scheme, MCS. Due to complexity reasons, also in the incremental redundancy implementation of this Thesis, retransmissions should use the same transmission format as first transmission and even the same number of resource blocks. Therefore, the model used is totally valid. Note that our assumption is widely accepted in scheduling-related research conducted in LTE as shown in subsequent chapters.

Uplink reports delays are realistically modelled in this Thesis. The standard specified delay of 4 Transmission Time Intervals (TTIs) has been considered from the reception of the TB until the transmission of the acknowledgement. Additionally, 2 more milliseconds are assumed until a retransmission of the negative acknowledged transmission can be conducted.

3.3.5 Link adaptation

Link adaptation is realistically modelled in this Thesis. This means that:

- Channel State Information (CSI) reports are realistically calculated at the UE, including the assumption of measurement errors,

- CSI reporting is realistically modelled, including reporting delays,
- CSI are used at the eNodeB to perform the link adaptation.

Concerning the calculation of CSI reports, it is assumed that the UE knows the channel and power received from each eNodeB. Therefore, it is possible to know the post-receiver SINR that can be achieved using any multi-antenna scheme (including any combination fo precoding matrix and number of spatial layers). Using the link abstraction model it is possible to translate the SINR values to BLER values given a set of MCS. With BLER values and the maximum throughput corresponding to each MCS, effective throughput values can be obtained for each MCS. With these values, the multi-antenna scheme configuration maximizing the effective throughput is selected and reported as the CSI. The set of resources over which a CSI expands is referred to as the frequency domain CSI reference resource. In order to emulate SINR measurement errors, it has been assumed that there is random per RB SINR error drawn from a normal distribution with standard deviation equal to 1 dB.

With regard to the reporting of CSI, realistic delays of 4 ms have been considered from the subframe when the measurements used to calculate the CSI are performed until the subframe when the CSI is reported. Moreover, 2 additional milliseconds are assumed to be needed until the CSI is used to adapt a transmission. This realistic delay modelling is very important to assess properly the effect of link adaptation.

3.3.6 Cell selection

Several cell selection algorithms are used in LTE assessments to assign users to serving cells. In this Thesis, the selection is based on received power considering a handover margin. The serving cell is randomly selected from a candidate set of cells composed by those cells whose received power is within a predefined margin from the highest power received from any cell. This power margin is known as the handover margin. This model reflects appropriately cell selection in real networks where handover margins are commonly used, and, as a consequence, a user may be served in a specific moment by a cell whose received power is not the highest one.

3.3.7 Transport block sizes modelling

In LTE specifications a limited set of TB sizes is allowed. In fact, for a given number of resources, it is specified the set of allowed TB sizes. In order to obtain performance analysis results as realistic as possible, this fact has been considered in this Thesis. Table 3.3 presents a subset of the TB sizes allowed in LTE, where we have selected just a few columns of the original table.

Table 3.3: TB size table [78]

Index	Number of physical RBs								
	1	2	5	6	10	15	25	50	100
0	16	32	120	152	256	392	680	1384	2792
1	24	56	176	208	344	520	904	1800	3624
2	32	72	208	256	424	648	1096	2216	4584
3	40	104	256	328	568	872	1416	2856	5736
4	56	120	328	408	696	1064	1800	3624	7224
5	72	144	424	504	872	1320	2216	4392	8760
6	328	176	504	600	1032	1544	2600	5160	10296
7	104	224	584	712	1224	1800	3112	6200	12216
8	120	256	680	808	1384	2088	3496	6968	14112
9	136	296	776	936	1544	2344	4008	7992	15840
10	144	328	872	1032	1736	2664	4392	8760	17568
11	176	376	1000	1192	2024	2984	4968	9912	19848
12	208	440	1128	1352	2280	3368	5736	11448	22920
13	224	488	1256	1544	2536	3880	6456	12960	25456
14	256	552	1416	1736	2856	4264	7224	14112	28336
15	280	600	1544	1800	3112	4584	7736	15264	30576
16	328	632	1608	1928	3240	4968	7992	16416	32856
17	336	696	1800	2152	3624	5352	9144	18336	36696
18	376	776	1992	2344	4008	5992	9912	19848	39232
19	408	840	2152	2600	4264	6456	10680	21384	43816
20	440	904	2344	2792	4584	6968	11448	22920	46888
21	488	1000	2472	2984	4968	7480	12576	25456	51024
22	520	1064	2664	3240	5352	7992	13536	27376	55056
23	552	1128	2856	3496	5736	8504	14112	28336	57336
24	584	1192	2984	3624	5992	9144	15264	30576	61664
25	616	1256	3112	3752	6200	9528	15840	31704	63776
26	712	1480	3752	4392	7480	11064	18336	36696	75376

3.3.8 Control channels overhead modelling

Performance analysis of mobile communication systems usually focuses on the performance of the data channels. However, in order to conduct this performance assessment, overhead due to physical signals and control channels cannot be neglected.

Although part of this overhead is fixed (synchronization signals, broadcast channels), the total amount of overhead depends on the actual system configuration. For example, configurations with many transmitting antennas require a high overhead in reference signals. This fact is explicitly considered in LTE assessments. Additionally, in a system configuration with many simultaneously scheduled users, a great amount of downlink control channels overhead is needed to signal transmission characteristics. However, this overhead is not necessarily constant in all subframes of a real system, but it may change over the time depending on, e.g., scheduling, traffic models, and channel conditions. As a consequence, overhead can be modelled in a static or dynamic way.

The approach used in IMT-Advanced evaluations, and the one used in this Thesis, is to consider a fix amount of OFDM symbols per subframe devoted to transmission of control channels. Specifically, an overhead related parameter known as L is usually considered in the simulations, being L the abovementioned number of symbols.

Some evaluators perform their assessments for just one of the L values, while for other values of L they scale performance figures according to the overhead relations. This strategy reduces simulation time at the cost of a lower accuracy.

With the aim of simplifying the scheduling algorithms, and having a fixed control overhead in all subframes, synchronization and broadcast overhead is neglected in this Thesis. Its consideration would imply a higher overhead in some specific subframes. However, the overhead increase would be very low for the LTE bandwidths of 10 or 20 MHz, which are widely used in this Thesis. As a consequence, it is expected that the conclusions of the present Thesis would remain invariable upon considering this additional overhead.

3.4 Contributions of the Thesis to the system modelling of LTE

This section presents a set of contributions of this Thesis to the modelling of LTE.

3.4.1 Link level abstraction

Link level simulations have been conducted in this Thesis in order to obtain the BLER vs SINR curves to characterize the physical layer operation of LTE. Simulations are performed in AWGN channels as required by the MIESM methodology.

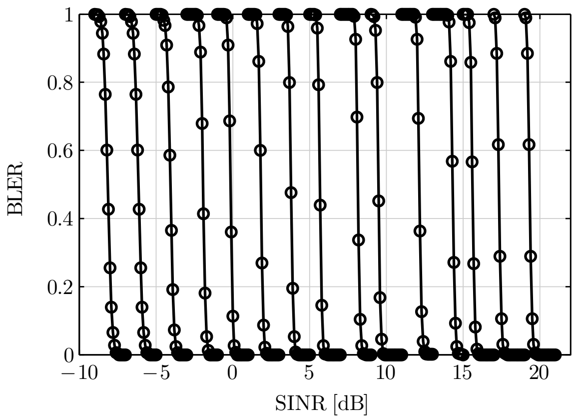


Figure 3.4: Example of BLER vs SINR curves.

It is worth noting that link level results obtained in this Thesis have been widely used in Wireless World Initiative New Radio + (WINNER+) by third parties in the evaluation of IMT-Advanced technologies.

As an example, Figure 3.4 shows the BLER vs SINR curves for the modulation and coding schemes corresponding to the 15 Channel Quality Indicator (CQI) levels included in the standard [78], assuming a bandwidth of 50 RBs and single antenna transmission. Moreover, note that these results have considered the same set of assumptions of [78] for the CSI reference resource, such as the use of 3 OFDM symbols for data transmission or the absence of synchronization and broadcast signals.

3.4.2 Channel estimation error modelling

In [79] a complete LTE Release 8 link level simulator was presented together with several performance results. Specifically, some channel estimators of different levels of complexity were compared, assessing their behaviour in realistic scenarios, where the receiver lacks a perfect knowledge of channel statistics. These channel estimators were mainly based on the use of Wiener filtering [80].

It was demonstrated that the use of robust estimators is able to keep the channel estimation error under control. This bounded error translates into a SINR demodulation penalty between 0.5 dB and 2 dB for the tested estimators. The use of Wiener filtering in frequency domain and linear averaging in time domain could be a reasonable implementation assumption with around 1 dB penalty. This penalty is used in this Thesis to model the effect of channel estimation errors in demodulation. Moreover, this mean degradation could be assumed to be equal to the variance of the error of the SINR measurements performed by the user equipments to report the channel quality to the serving cell. In fact, most researchers model this error as a lognormal distribution with zero mean and unit variance. Our results confirm this model, which is widely used along the Thesis.

3.4.3 Interference modelling

As commented in the introduction chapter, explicit modelling of interferences is required in the performance evaluation according to IMT-Advanced guidelines. However, in order to perform an efficient evaluation and to reduce the computational burden, simplifications are needed. In [37], Skillermark *et al.* studied whether it is necessary to simulate in detail (that is, with small-scale variation) every link among user equipments and base stations or not. It is demonstrated that it is enough to model in detail only the strongest interferers. Evaluation accuracy is measured comparing SINR and throughput statistics obtained for different numbers of strongest interferers modelled in detail.

The major concern raised in this Thesis with regards to the Skillermark evaluation is that in his work, a fixed number of accurately modelled interferers was determined for IMT-Advanced evaluations, but it was not provided any rule to determine the number of interferers to model in detail in a general scenario.

The proposal of this Thesis is to model in detail the strongest interferers, those received with the highest power by the receiver, and whose aggregate of powers plus noise power is, at least, the 95% of the total interference plus noise power.

Our rule selects a lower number of accurately modelled links than the rule implicitly promoted by Skillermark if the noise experienced by the user is comparable to the interference levels, or if there is a reduced set of cells strongly received. This lower number of links will not produce a higher error if the power threshold is well selected. On the other hand, the proposed criterion will model in detail more interferers than the Skillermark criterion if a user needs it, increasing the accuracy of the interference modelling of this user. Several power percentage thresholds have been tested in a scenario similar to that used by Skillermark and we found that the application of this rule reduces the estimation error for the same mean number of accurately modelled links per user. Figure 3.5 presents the CDF of the SINR error for several configurations. Two curves consider a fixed number of accurately modelled links, specifically 3 and 8. Three curves use the criterion proposed in this Thesis with three power

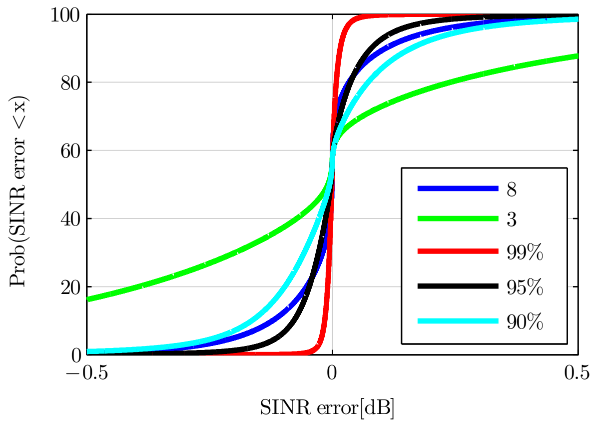


Figure 3.5: SINR error CDFs for different modelling strategies.

thresholds: 90%, 95% and 99%. The latter configurations correspond with a mean number of accurately modelled links approximately equal to 6, 8 and 16, respectively. It can be observed that the configuration with a fixed number of modelled interferers equal to 8 has more variance than the configuration with the power threshold set at 95%, while both configurations have the same mean number of accurately modelled links.

The general rule provided would keep the modelling error bounded in any scenario, no matter whether it is homogeneous (such as the IMT-Advanced scenarios) or heterogeneous. The use of such a general rule could be very interesting for, at least, the initial phase of evaluation in any heterogeneous scenario with many transmission points and many users. This kind of scenario is becoming the preferred one, since the next generation of mobile communications, the Fifth Generation (5G), aims at increasing the number of supported users by a factor between 10 and 100.

3.4.4 Shadowing correlation implementation

One important implementation aspect of a system level simulator is how to generate spatially correlated large-scale channel parameters, such as correlated shadowing.

The common approach is to sample the simulation scenario to represent it as a matrix or a discrete map. Then, filter this matrix in order to force a desired spatial correlation.

The conventional approach to performing the filtering is to define a correlation matrix, **R**, whose values indicate the correlation of the shadowing between the points in the matrix of sampled positions. Then, the Cholesky factor, **L**, of the correlation matrix is obtained, where $\mathbf{L}\mathbf{L}^T = \mathbf{R}$. Finally an uncorrelated Gaussian vector, **a** with as many elements as points in the map, is filtered with **L** to obtain a vector of correlated values **s**, in such a way that $\mathbf{s} = \mathbf{L}\mathbf{a}$. Finally, **s** would be shaped to fit the required matrix of correlated values [81].

The complexity of this approach is prohibitive to model correlation in wide areas with many sampled points. Therefore, simplifications have been proposed such as the Claussen model [82].

The approach considered in this Thesis is to reuse the conventional approach without generating maps. Instead, after dropping the users in the scenario their positions are collected and a correlation matrix is generated that considers only the correlations between the points were users have been placed. This is a quite simple consideration, but it is very useful in scenarios where the number of users simulated is much lower than the possible locations, to reduce calculations needed to correlate values or to reduce the memory required to store correlated matrices of large size.

3.5 Conclusions

This chapter has presented and described in depth the methodology used in this Thesis. The main Key Performance Indicators (KPIs) to be used in the following chapters have been listed, and the simulation procedure and its division into two levels, link-level and system-level, has been described.

The chapter has also detailed some important models, focusing on those in which the contribution of the Thesis is significant. Of particular relevance is the mathematical description of the receiver where the focus is on linear MRC and MMSE receivers. Those simple but effective receivers are the standard de-facto for LTE evaluations and will be used in the rest of the Thesis. For an accurate modelling of a real LTE-like system, other procedures must be taken into account, like LA, HARQ and the overheads from control channels. All these aspects have been analysed in detail.

The last section of the chapter has presented some additional contributions of the Thesis to the LTE modelling. In particular, link level abstraction represents one of the main contributions of this Thesis to the research community, since the link level results obtained in this Thesis are widely used in the performance evaluation of IMT-Advanced technologies.

Finally, it is worth recalling that this chapter complements with the description of the system simulator that can be found in appendix A.

Chapter 4

LTE Link Adaptation Analysis

This chapter presents a deep analysis of the link adaptation capability in the downlink of Long Term Evolution (LTE). With this aim, the chapter identifies the link adaptation aspects included in the the Third Generation Partnership Project (3GPP) specifications and those not covered by them. In the first case, a detailed exposition of the specifications is provided, with special focus on the aspects where different interpretations are possible and the implications of these different interpretations on the system performance and the operation of other algorithms. Concerning the aspects not covered by the specifications, the purpose of this chapter is to fill the gaps of the specifications with state-of-the-art solutions.

With these goals in mind, the chapter has been divided into the next sections:

- Section 4.1 introduces the Link Adaptation (LA) capabilities of the LTE system.
- Section 4.2 details the Channel State Information (CSI) definitions, reporting modes and reporting testing measurement as they are found in the 3GPP specifications.
- Section 4.3 presents the CSI calculation as an open issue with its different possibilities.
- Section 4.4 describes how base stations may use the CSI in order to adapt the transmissions.

- Section 4.5 outlines the main conclusions of this chapter.

4.1 Introduction

The LA topic was already introduced in section 2.4.2 as one of the key techniques of LTE. It was defined as the capability of the transmitters to adapt the transmission format to the channel quality variations. In this regard, the specifications indicate which transmission parameters can be adapted, how can they be adapted, and what type of physical measurements can be performed and sent to the base stations to assist LA decision making. However, LA algorithms are not included in the LTE Advanced (LTE-Advanced) specifications and each developer can use the standard to develop its own LA algorithms.

Concerning the transmission parameters, the most important characteristics that can be adapted in LTE were already presented in section 2.4.2:

- The multi-antenna scheme can be dynamically changed to get the most of the channel.
- The transmission rank represents the number of independent streams or layers in LTE terminology transmitted in Multiple-Input Multiple-Output (MIMO) modes. It is worth noting that the maximum number of independent codewords that can be transmitted is two. That is to say, in case of transmitting more than two layers, then a codeword is transmitted over multiple layers. Note also that the same number of layers is used in all the resources occupied by a transmission.
- The set of Modulation and Coding Scheme (MCS) that can be used in a transmission depends on the resource block allocation. Supported modulations are Quadrature Phase Shift Keying (QPSK), 16-Quadrature Amplitude Modulation (QAM) and 64-QAM. Given a resource block allocation, a limited set of combinations of modulation and transport block size is available. Each one of these combinations provides a different effective coding rate. In multi-codeword transmission, each codeword can use a different MCS. However, each codeword must use the same MCS in all its allocated resource blocks.
- The precoder in LTE can be a codeword from a predefined codebook or calculated according to proprietary algorithms. The precoder can be wideband or frequency-selective, that is to say, both the same or different precoders can be used in the multiple resources allocated to a user.

Concerning how the LA obtains the knowledge about the channel state, it is of special relevance the 3GPP document [24], which gathered the results of the

3GPP study item on LTE-Advanced physical layer aspects. In this report, possible feedback in support of Downlink (DL) Coordinated Multi-Point (CoMP) was identified, although it is also valid for non-coordinated transmissions. The ideas of this report are reproduced here with further explanations in order to understand the CSI feedback.

Two types of feedback mechanisms were identified:

- explicit CSI feedback: the channel state or its statistics are reported as observed by the receiver, without assuming any transmission or reception processing. Both a channel part and a noise part should be reported. Concerning the channel part, channel properties include channel matrix (full matrix or main eigencomponents), and transmit channel covariance (full matrix or main eigencomponents). Concerning the noise part, the feedback could consist of, e.g., total received power, or covariance matrix (full matrix or main eigencomponents) of the total received signal or the covariance matrix of the noise plus interferences.

- implicit CSI feedback: the channel state or its statistics are reported using hypotheses of different transmission and/or reception processing. For example, feedback could hypothesize about whether Single User MIMO (SU-MIMO) or Multi-User MIMO (MU-MIMO) will be used, or about the kind of transmit precoders used, or the type of receiver (Minimum Mean Square Error (MMSE), Maximum Ratio Combining (MRC), etc.), or about the interference situation.

Additionally, it was also considered a direct CSI estimation, where the channel is directly estimated in uplink using sounding reference signals sent by the User Equipment (UE) and exploiting channel reciprocity.

It is worth noting that it is possible to have hybrid approaches where the CSI knowledge is gained at the base station through multiple types of mechanisms.

Comparing explicit and implicit feedback, it is said that, in general, explicit feedback increases switching and scheduling ability due to its more complete information. However, explicit feedback provides less accurate link adaptation since the specific capabilities of each UE are not considered in the CSI reports. Moreover, explicit feedback requires higher uplink overhead. See [83] for a more detailed comparison of explicit and implicit feedbacks.

The selection of one type of feedback mechanism or another depends on multiple factors. For example, if scheduling flexibility is not a must because a small amount of users is simultaneously scheduled, implicit feedback could be of interest.

LTE current specifications consider the implicit CSI feedback as the only allowed feedback mechanism. Direct estimation through uplink sounding is not explicitly treated in the specifications but its support is just an implementation issue.

Concerning the CSI feedback periodicity and level of detail (frequency span, number of eigencomponents, etc.), it is important to consider the Uplink (UL) overhead (i.e. the number of bits associated with each specific feedback mechanism). DL performance could be enhanced up to some level increasing the UL overhead, but less resources could be used for data transmission in UL. Therefore, selection of feedback parameters should be the result of a trade-off between UL and DL performance. At least, if a given DL performance is required, minimum UL overhead should be used.

An aspect that has not received much attention is the time span of measurements used to estimate the CSI feedback. Feedback can represent an instantaneous measure or the average over a time interval. Let us consider the example of the interference covariance seen by a UE. It is desirable for the UE to report a value as close as possible to the value experienced in a following transmission. However, it is not clear whether an instantaneous measure is better than a long term average.

4.2 LA specification support

4.2.1 CSI definitions

Three CSI reports are defined in the specifications [78]:

- Rank Indicator (RI)

  The RI is the *"number of useful transmission layers"* estimated by the UE for a channel.

- Precoding Matrix Indicator (PMI)

  The PMI is an index that identifies a matrix from a codebook of precoding matrices that can be used as the precoding matrix for a downlink transmission. The specifications state that its calculation is conditioned on the last reported RI.

- Channel Quality Indicator (CQI)

  Finally, the CQI is a subjective measure of the channel quality that indicates the most efficient MCS, from a set of MCSs, that can be used with a Block Error Rate (BLER) lower than 0.1, conditioned on the reported RI and PMI. The exact definition from [84] (with just the modification of a reference to a table) is: "*Based on an unrestricted observation interval in time and frequency, the UE shall derive for each CQI value reported* *in uplink subframe n the highest CQI index between 1 and 15 in Table 4.1 which satisfies the following condition, or CQI index 0 if CQI index 1 does not satisfy the condition: A single PDSCH transport block with a combination of modulation scheme and transport block size corresponding to the CQI index, and occupying a group of downlink physical resource blocks termed the CSI reference resource, could be received with a transport block error probability not exceeding 0.1*". The definition of the CSI reference resource is as follows: "*In the frequency domain, the CSI reference resource is defined by the group of downlink physical resource blocks corresponding to the band to which the derived CQI value relates. In the time domain, the CSI reference resource is defined by a single downlink subframe (...). In the layer domain, the CSI reference resource is defined by any RI and PMI on which the CQI is conditioned.*". Additionally, [84] includes a set of assumptions made for the purpose of deriving the CQI concerning the CSI reference resource. For example, it is assumed that the first 3 Orthogonal Frequency Division Multiplexing (OFDM) symbols in the subframe are occupied by control signalling, that there are not any resource elements used by synchronisation or broadcast signals, that the cyclic prefix length is normal, that a specific redundancy version of the transmission is used (version 0), and that a specific transmission mode is used.

Table 4.1 presents the correspondence between each CQI index and the modulation and coding rate.

As above-mentioned, direct estimation of the channel state is also possible in the evolved Node B (eNodeB), for example using uplink sounding signals. Typically, the channel matrix itself is measured. This represents a short term –because it is the result of an instantaneous or short period measure– narrowband –because it is done over a subcarrier or a small subband– channel state measure. This CSI could be processed to obtain other channel representations:

- Transmit channel covariance matrix, as another short-term narrowband measure.

- Wideband transmit channel covariance matrix, averaging the covariance matrices of the multiple subbands over the whole system bandwidth.

- Long-term wideband transmit channel covariance matrix, averaging the covariance matrices in time and frequency domains.

Table 4.1: CQI table [78]

CQI index	**modulation**	**code rate ×1024**	**code rate**	**efficiency**
0	–	–	–	–
1	QPSK	78	0.0762	0.1523
2	QPSK	120	0.1172	0.2344
3	QPSK	193	0.1885	0.3770
4	QPSK	308	0.3008	0.6016
5	QPSK	449	0.4385	0.8770
6	QPSK	602	0.5879	1.1758
7	16QAM	378	0.3691	1.4766
8	16QAM	490	0.4785	1.9141
9	16QAM	616	0.6016	2.4063
10	64QAM	466	0.4551	2.7305
11	64QAM	567	0.5537	3.3223
12	64QAM	666	0.6504	3.9023
13	64QAM	772	0.7539	4.5234
14	64QAM	873	0.8525	5.1152
15	64QAM	948	0.9258	5.5547

4.2.2 CSI reporting modes

Both periodic and aperiodic reporting are supported by the LTE standard. The aperiodic reports are more detailed than the periodic reports, which are meant to provide a coarse channel knowledge.

Aperiodic reports are conducted over the Physical Uplink Shared Channel (PUSCH) channel. Concerning the CQI reporting, three types of aperiodic feedback are distinguished:

- wideband feedback, when a unique CQI is calculated for the whole bandwidth,
- higher layer configured (or eNodeB-configured) subband feedback, when both a wideband CQI and CQI for each subband (whose size depends on the system bandwidth) are reported, and
- UE-selected subband feedback, when the UE selects M subbands (whose size depends on the system bandwidth), and reports a wideband CQI and a CQI calculated over the M subbands.

Concerning the PMI, some reporting modes do not include PMI feedback, others consider a single PMI calculated over a whole set of frequencies of interest, and the remainder consider the feedback of multiple PMIs, each one calculated for a specific subband.

Not all the combinations of CQI feedback and PMI feedback types are allowed. Table 4.2 presents the allowed configurations and their names. Finally, it is worth noting that each transmission mode uses a subset of the available reporting modes. For example, transmission mode 4, i.e. closed-loop spatial multiplexing, is able to use modes 1–2, 2–2, and 3–1.

Table 4.2: Aperiodic CSI reporting modes.

		PMI Feedback Type		
		NO PMI	Single PMI	Multiple PMI
CQI Feedback Type	Wideband (wideband CQI)			Mode 1–2
	UE selected (subband CQI)	Mode 2–0		Mode 2–2
	Higher layer config. (subband CQI)	Mode 3–0	Mode 3–1	

Periodic reports are conducted over the Physical Uplink Control Channel (PUCCH) channel. Concerning CQI reporting, only wideband feedback and UE-selected subband feedback are supported. Table 4.3 presents the allowed configurations and their names. Transmission mode 4, i.e. closed-loop spatial multiplexing, is able to use modes 1–1, and 2–1.

Table 4.3: Periodic CSI reporting modes.

		PMI Feedback Type	
		NO PMI	Single PMI
CQI Feedback Type	Wideband (wideband CQI)	Mode 1–0	Mode 1–1
	UE selected (subband CQI)	Mode 2–0	Mode 2–1

4.2.3 CSI reporting testing

Definition of tests to check the correct CSI reporting is provided in section 9 of [85]. It is very important to take into account that in the defined tests there are references to Signal to Noise Ratio (SNR), but not to Signal to Interference plus Noise Ratio (SINR). That is to say, interference is not considered.

Concerning the CQI reporting, both reporting under Additive White Gaussian Noise (AWGN) conditions and under fading conditions are considered.

In the AWGN testing, noise level and useful signal levels are fixed to have a specific SNR. In this kind of testing, the object of testing is wideband CQI reporting. As an example, let us explain the first test included in the specifications. In this test, the UE measures the channel and reports a CQI with 5 ms periodicity measured over the whole bandwidth (wideband CQI). In a first phase, the eNodeB uses a fixed MCS. After this phase the median CQI measured by the UE is calculated. Then, a second phase starts with the eNodeB using a fixed MCS corresponding to the median CQI (MCS_{med}), median CQI + 1 (MCS_{med+1}), or median CQI -1 (MCS_{med-1}). The test is passed if the BLER with MCS_{med} is lower than 0.1 and the BLER with MCS_{med+1} is higher than 0.1, or if the BLER with MCS_{med} is higher than 0.1 and the BLER with MCS_{med-1} is lower than 0.1.

In the testing with fading conditions three scenarios are considered: frequency selective scheduling, non-frequency selective scheduling and frequency-selective interference. It could be striking the consideration of a scenario with frequency-selective interference given that it has been highlighted that there is no interference in the CSI tests. In fact, frequency-selective interference in this kind of tests is emulated with an AWGN noise with different power in each one of the subbands considered in the testing (and hence the frequency-selectivity). Therefore, it is not a realistic interference, but a simplification. The first test for the fading conditions with frequency-selective scheduling is quite similar to the above-described test for AWGN. In fact, the first phase is the same up to the calculation of the median CQI. The second phase is divided into two parts. In one part, the eNodeB makes transmissions over randomly selected subbands using the MCS_{med}. In the second part, the eNodeB makes transmissions over the best subband selected according to the subband CQIs sent by the UE, and with the corresponding MCS. The test is passed if the second part obtains a throughput that is at least a predefined number of times higher than the obtained with the random subband selection (without subband CQI).

Concerning the PMI reporting, it is based on the analysis of the relative increase in throughput when the transmitter uses the recommended precoder compared to when a random precoder is used.

Finally, the RI reporting testing analyses the relative increase in throughput when transmissions are based on the reported rank compared to when a fixed rank is used.

For more details about how the testing is performed, the interested reader is referred to [85].

4.3 CSI calculation

Despite the definitions given for CSI reports, and the description of the tests to check the correct reporting, it is yet not completely described how the CSI must be calculated. Next subsections present the unspecified issues and options to implement them in practice.

4.3.1 PMI and RI calculation

PMI and RI definitions and tests are especially open to different calculation implementations.

In this regard, Love presented in [86] multiple criteria to choose the optimal precoding matrix from a given codebook: a criterion to minimize the error rate for the Maximum Likelihood (ML) decoder, two criteria for linear receivers (both Zero Forcing (ZF) and MMSE receivers), and a capacity selection criterion that maximizes the mutual information.

Schwarz extended and particularized for LTE the Love's capacity criterion in [87]. Schwarz stated that the joint optimization of RI, PMI and CQI is not computationally feasible. The solution, according to Schwarz, is to separate the optimization process into different independent problems providing local optimal values. Schwarz criterion is to select the best wideband PMI based on the maximization of the sum of the pre-equalization mutual information over the set of subcarriers of interest. Moreover, Schwarz analysed the effect of considering a subset of subcarriers instead of the whole bandwidth in the maximization.

Schwarz extended his previously commented work in [88], where the maximization of sum mutual information over subcarriers is particularized for linear receivers and used to evaluate the best RI and PMI. Specialization makes Schwarz to consider the post-equalization mutual information, which is more appropriate for linear receivers than the pre-equalization mutual information. The post-equalization mutual information for a specific resource r and a layer l is defined as:

$$I_{r,l} = \log_2(\gamma_{r,l}), \tag{4.1}$$

where $\gamma_{r,l}$ is the post-equalization SINR for resource k and layer l. Joint optimization of PMI and RI is based on the exhaustive search of the combination of precoder and rank providing the highest sum mutual information over the resources of interest. Once identified the optimal combination, the corresponding precoder is used as PMI, and the corresponding rank is used as RI. Concerning the CQI determination, Schwarz bases its process on the calculation of the effective post-receiver SINR. For each MCS in the CQI table, an effective SINR is calculated from the individual SINR values measured by the UE in the resources of interest. This value is mapped to a BLER value, and the highest index with a BLER lower than 0.1 is signalled as the CQI.

One concern related to the Schwarz approach is that mutual information per resource (calculated for bit-interleaved coded modulation) depends not only on the SINR, but also on the modulation used in that resource. Moreover, in LTE the same modulation (and coding) must be used in all the resources occupied by the transmission. Therefore, the maximization of the sum of the mutual information is not necessarily an optimal approach. However, since the mutual information curves of higher order modulations are higher or equal than the modulations of lower order for *almost all* the SINR range, this is in fact a close-to-optimal approach. Consequently, this approach has been used in this Thesis.

4.3.2 Interference knowledge

Schwarz works consider only the interference produced between streams of a multi-stream transmission, i.e. intra-stream interference, but it does not consider the presence of interference coming from cells different to the serving cell, i.e. inter-cell interference.

It is simple to estimate the intra-stream interference if both the channel to the serving cell and the precoder used by this cell are known. But, how to estimate the inter-cell interference is not a so simple task. Concerning the knowledge that the receiver has about inter-cell interference, we could distinguish between measurement of a full interference covariance matrix, and knowledge of the aggregate of power received from interferers.

The first option would be the one providing optimal results given that the measured covariance matrix is a good estimate of the covariance to be experienced in a future transmission adapted according to this CSI knowledge. The rationale is that, in order to maximize the SINR, it is not valid to just maximize the signal component, but a trade-off between useful signal maximization and interference minimization must be found. The knowledge of the spatial colouring of the interference provided by the interference covariance matrix allows finding this trade-off.

Nevertheless, one may ask what would happen if the interference spatial distribution changed fast. That is to say, if the interference covariance matrix measurement done in a specific instant was not a good estimate of the covariance experienced in future transmissions. In this case, the explained trade-off would not be valid. Probably, a more robust approach would be not considering the interference spatial distribution but only maximize the useful signal.

4.3.3 Averaging and interference issues

There is a part of the CQI definition whose importance is not usually well weighed: "*based on an unrestricted observation interval in time and frequency*". These words mean that the frequency and time span of the measurements used to derive the CQI are not constrained to the specific CSI reference resource of the CQI.

Obviously, the restriction of the measurement to the specific CSI reference resource of the report (in frequency and time), is an allowed embodiment of the CQI calculation, but not the only one.

Think about the tests described in subsection 4.2.3. In those tests, it does not make much sense to extend the measurements beyond the span of the CSI reference resource. Neither does it in the tests performed by Schwarz in his works. However, the sense appears if we consider scenarios with inter-cell interference, especially with highly variable interference. In this kind of scenarios, it could be interesting to perform some averaging of the interference using an observation interval larger than the CSI reference resource.

Note that the target of link adaptation is to adapt transmission formats to the channel quality. Therefore, the optimal approach, in the sense of providing the best possible link adaptation, would be that whose channel quality estimation/prediction was more similar to the actual channel quality in a future transmission. The problem with inter-cell interference is that its dynamics in case of rapid change of precoders in the transmitters is not easily predictable by a receiver, or not predictable at all. Hence, an instantaneous measurement may not be a good estimate of the future interference. The use of an averaged interference estimation could provide better adaptation statistics. On the other hand, if the interference variability is due to the mobility averaging it could be a bad idea.

4.4 CSI use in the eNodeB

The specifications do not include an algorithm to adapt the downlink transmissions based on the available CSI. Therefore, each base station may apply a different algorithm.

Concerning the explicit reporting of CSI, and excluding the non-codebook-based transmission modes, the role of each CSI report is clear. The RI is used to select the maximum number of transmission layers in a specific transmission, the PMI is used to select the precoder from a codebook, and the CQI is used to select the modulation and coding scheme.

Some modifications can be needed before using the CQI to select the modulation and coding scheme. First, the received CQI must be corrected before it is used to adapt the MCS of a transmission when the percentage of resources devoted to this transmission is different to the value assumed in the calculation of the CQI (3 OFDM symbols used to transmit control signalling are assumed in the calculation).

An additional correction could be done based on the Hybrid ARQ (HARQ) feedback sent by the UE to the eNodeB. For instance, if a negative acknowledgement is received by the eNodeB, it indicates a likely bad transmission format selection in a previous transmission, specifically a not robust enough format. The process that takes into account HARQ feedbacks to correct LA information is referred to as Outer-Loop Link Adaptation (OLLA) [48], in contrast with the Inner Loop Link Adaptation (ILLA) that just takes the CQI and selects the corresponding MCS.

The OLLA uses a CQI offset, Δ_{CQI}, to correct CQI values directly received from UE in order to control the BLER measured for the first transmission of blocks. The corrected CQI, $\widehat{CQI}$, depends on the reported CQI, CQI_r, according to the next formula:

$$\widehat{CQI} = CQI_r - \Delta_{CQI}. \tag{4.2}$$

Each time a Negative Acknowledgement (NACK) is received from the UE corresponding to a transport block that was transmitted for the first time, the CQI correction is incremented in Δ_{UP}, whereas the CQI correction is decremented in Δ_{DOWN} when a Positive Acknowledgement (ACK) is received from the UE.

The $BLER$ for first transmissions with this algorithm tends to be:

$$BLER = \frac{1}{1 + \frac{\Delta_{UP}}{\Delta_{DOWN}}}, \tag{4.3}$$

or equivalently:

$$BLER = \frac{\Delta_{DOWN}}{\Delta_{UP} + \Delta_{DOWN}}. \tag{4.4}$$

This $BLER$ value is said to be the target $BLER$. This value, together with the specific Δ_{UP} and Δ_{DOWN} values, can be adjusted to optimize the system performance.

A special case of OLLA is found when there are multiple streams. In this case, several CQI offsets are configured, one per each multi-stream configuration and per stream. That is to say, in case that 1 or 2 streams could be used, there will be 3 CQI offsets, one for single-stream transmission and the other two for each one of the streams of the two-stream transmission.

Figure 4.1 presents graphically the above-mentioned ILLA and OLLA processes and which are the inputs to these processes. Together they form the

LA module in the Medium Access Control (MAC) layer, used by the packet scheduler to request the best transmission format for a specific user over a set of resources (shown as "LA request", while the answer is the "decision").

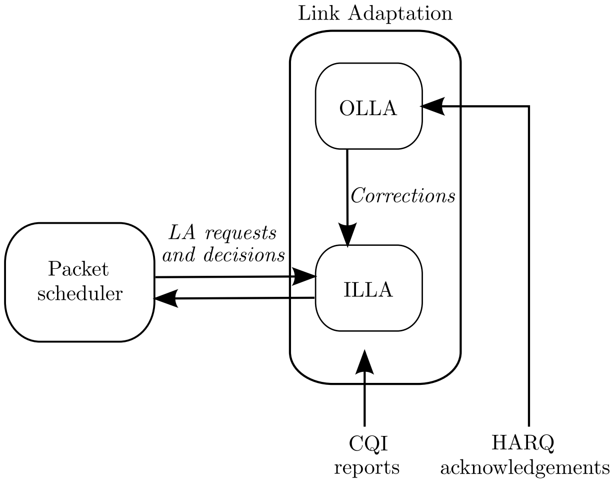


Figure 4.1: CSI processing at eNodeB

In order to conclude with the corrections to be applied to the CQI received at the base station, also filtering of CQI reports in time, or prediction could be used. These features are, nevertheless, not explored in this Thesis.

In relation to link adaptation decisions at the eNodeB, it would be desirable to know the data rate that each UE could experience given whatever resource allocation. Each CSI report expands a specific frequency domain CSI reference resource, indicating the achievable data rate in a transmission over this reference resource. Therefore, in order to know the achievable data rate over a set of resources different to the set used by a CSI report, any processing must be done. Two cases can be considered. First, the set of resources under test may be partially included in a frequency domain CSI reference resource. In this case, it is assumed that the resource allocation under test may be characterized by the same CSI of the reference resource. Second, the set of resources under test may expand over multiple frequency domain CSI reference resources. In this case, each Resource Block (RB) of the set of resources under test is assigned a post-processing SINR equal to the effective SINR characteristic of the CSI report corresponding to the CSI reference resource in which this RB was included. Then, a CSI is calculated over the set of resources under test as normal.

If non-codebook based transmission modes are considered, direct estimation of CSI is more useful than in codebook based transmission. Direct estimation allows more flexibility in the selection of the precoders, what is very important both in SU-MIMO and in MU-MIMO. The problem with this estimation is that interference information can not be measured. Therefore, in practice, direct estimation is completed with information from explicit CSI reports.

In non-codebook based SU-MIMO, the optimum precoder is the dominant right singular-vector of $\mathbf{R_{in}}^{-1/2}\mathbf{H}$ or, equivalently, the main eigenvector of $\mathbf{H}^H\mathbf{R_{in}}^{-1}\mathbf{H}$. It is worth noting that if $\mathbf{R_{in}}$ is a diagonal matrix with all its elements being equal, then the optimum precoder is the right singular-vector of $\mathbf{H}$ or, equivalently, the eigenvector of $\mathbf{R_H} = \mathbf{H}^H\mathbf{H}$. This vector is also the best precoder when $\mathbf{R_{in}}$ is not known. Direct estimation of CSI provides the information needed to calculate these non-codebook precoders except, precisely, $\mathbf{R_{in}}$. The reason is that this matrix is measured only in downlink.

The interference covariance matrix can be estimated from CSI feedbacks, i.e., CQI and PMI. In fact, the channel covariance or the channel itself could also be reconstructed from the CSI feedbacks.

In addition, we will consider a non-standard mode of operation of LTE in which the interference covariance is reported by the user.

In a non-codebook based precoding with MU-MIMO, given a set of users, if their channels are known (or their transmit channel covariance), precoders can be selected according to different algorithms. In this Thesis, two schemes are used:

- Block diagonalization

  Block Diagonalization (BD) is a linear precoding technique that allows spatial multiplexing between different users and the transmission of multiple data streams per user. Its main characteristic is the nulling of the multi-user interference. In this sense, BD is similar to the ZF algorithms. BD was first proposed in [89] and further investigated in [90]. The original algorithm assumes the knowledge of the short term CSI in form of the channel matrix itself. A less optimal version of the algorithm can still be used if only long-term spatial channel covariance is known at the transmitter. In the multi-user transmissions considered in this Thesis, only rank 1 transmissions are allowed to each user. This fact enables a simplification of the BD calculations which are explained in the following lines.

  We consider a set of J users with Q_j antennas each user j, where $j \in [0, J-1]$. The cell that serves those users has P antennas. We assume that $P > Q_j$ for all j.

Let us denote $\mathbf{H}_j \in \mathbb{C}^{Q_j \times P}$ as the channel between the serving cell and the user j.

For each user j, assume the knowledge of $\mathbf{R}_j \in \mathbb{C}^{P \times P}$, where $\mathbf{R}_j = \mathbf{H}_j^H \mathbf{H}_j$. This matrix can be decomposed as:

$$\mathbf{R}_j = \mathbf{V}_j \mathbf{S}_j^2 \mathbf{V}_j^H, \tag{4.5}$$

where $\mathbf{V}_j \in \mathbb{C}^{P \times P}$ is the matrix with eigenvectors of $\mathbf{R}_j$, and $\mathbf{S}^2 \in \mathbb{C}^{P \times P}$ is the matrix of eigenvalues. Using this decomposition, an equivalent channel (this is the equivalent channel when the receiver uses as receiver matrix the hermitian of the left singular vector matrix of $\mathbf{H}_j$) can be obtained with:

$$\widehat{\mathbf{H}}_j = \mathbf{S}_j \mathbf{V}_j^H, \tag{4.6}$$

where $\widehat{\mathbf{H}}_j \in \mathbb{C}^{P \times P}$.

Next, obtain a reduced rank equivalent channel. With this aim, it is necessary to calculate the singular value decomposition of the equivalent channel:

$$\widehat{\mathbf{H}}_j = \widehat{\mathbf{U}}_j \widehat{\mathbf{S}}_j \widehat{\mathbf{V}}_j^H, \tag{4.7}$$

where $\widehat{\mathbf{U}}_j$, $\widehat{\mathbf{S}}_j$ and $\widehat{\mathbf{V}}_j^H \in \mathbb{C}^{P \times P}$. Using this decomposition, pre-multiply the equivalent channel by the hermitian of the first left singular vector of $\widehat{\mathbf{H}}_j$, that is to say, $\left(\left[\widehat{\mathbf{U}}_j\right]_{:,1}\right)^H$. The new channel variable obtained is $\overline{\mathbf{H}}_j$:

$$\overline{\mathbf{H}}_j = \left(\left[\widehat{\mathbf{U}}_j\right]_{:,1}\right)^H \widehat{\mathbf{H}}_j, \tag{4.8}$$

where $\overline{\mathbf{H}}_j \in \mathbb{C}^{1 \times P}$.

For each user j, build $\widetilde{\mathbf{H}}_j \in \mathbb{C}^{(J-1) \times P}$ aggregating row-by-row the $\overline{\mathbf{H}}_j$ of the other users:

$$\widetilde{\mathbf{H}}_j = \left[\overline{\mathbf{H}}_0^H ... \overline{\mathbf{H}}_{j-1}^H \overline{\mathbf{H}}_{j+1}^H ... \overline{\mathbf{H}}_{J-1}^H\right]^H. \tag{4.9}$$

Next, for each user j, obtain the singular decomposition of $\widetilde{\mathbf{H}}_j$:

$$\widetilde{\mathbf{H}}_j = \widetilde{\mathbf{U}}_j \widetilde{\mathbf{S}}_j \widetilde{\mathbf{V}}_j^H \tag{4.10}$$

Let us define the null space dimension of the channel matrix $\widetilde{\mathbf{H}}_j$ as $N_\emptyset$. This value is, at most, equal to the absolute difference of the number of transmit antennas minus the number of multiplexed entities plus one:

$$N_{\emptyset} \leq |P - J + 1|. \tag{4.11}$$

Build $\widetilde{\mathbf{V}}_{j\emptyset} \in \mathbb{C}^{P \times N_{\emptyset}}$ as the submatrix of $\widetilde{\mathbf{V}}_j$ containing its last $N_{\emptyset}$ columns. That is to say, a matrix whose columns form an (orthogonal) basis for the null space of $\widetilde{\mathbf{H}}_j$.

Multiply the equivalent channel matrix, $\widehat{\mathbf{H}}_j$, by the matrix $\widetilde{\mathbf{V}}_{j\emptyset}$:

$$\widetilde{\widetilde{\mathbf{H}}}_{j\emptyset} = \widehat{\mathbf{H}}_j \widetilde{\mathbf{V}}_{j\emptyset}, \tag{4.12}$$

where $\widetilde{\widetilde{\mathbf{H}}}_{j\emptyset} \in \mathbb{C}^{P \times N_{\emptyset}}$ lies in the null space of $\widetilde{\mathbf{H}}_j$.

This new matrix is again decomposed with singular value decomposition:

$$\widetilde{\widetilde{\mathbf{H}}}_{j\emptyset} = \widetilde{\widetilde{\mathbf{U}}}_{j\emptyset} \widetilde{\widetilde{\mathbf{S}}}_{j\emptyset} \widetilde{\widetilde{\mathbf{V}}}_{j\emptyset}^{H}, \tag{4.13}$$

where $\widetilde{\widetilde{\mathbf{V}}}_{j\emptyset} \in \mathbb{C}^{N_{\emptyset} \times N_{\emptyset}}$.

The precoder for user j is finally obtained as:

$$\mathbf{w}_j = \left[\widetilde{\mathbf{V}}_{j\emptyset} \widetilde{\widetilde{\mathbf{V}}}_{j\emptyset} \right]_{:,1}, \tag{4.14}$$

with $\mathbf{w}_j \in \mathbb{C}^{P \times 1}$.

It is worth noting that the previous process can be also applied if $\mathbf{H}_j$ is known instead of $\mathbf{R}_j$. The first part of the process would be skipped up to the calculation of the equivalent channel that would be $\widehat{\mathbf{H}}_j = \mathbf{H}_j$.

- Signal to Leakage plus Noise Ratio (SLNR) Precoding

  SLNR precoding is a linear technique that maximizes the ratio between the useful signal power transmitted to a user and the aggregation of noise power plus interference power received by other users [91].

  The precoder for user j is obtained as:

$$\mathbf{w}_j = \operatorname{eig}\left(\left(Q_j \sigma_{\mathbf{in}}^2 \mathbf{I} + \sum_{\substack{k \neq j \\ k=0}}^{J-1} \mathbf{R}_k \right)^{-1} \mathbf{R}_j \right), \tag{4.15}$$

  where the operator eig$(\cdot)$ returns the principal eigenvector of the operand, $\sigma_{\mathbf{in}}^2$ is the noise plus inter-cell interference power per antenna, and $\mathbf{I} \in \mathbb{C}^{P \times P}$ is the identity matrix.

Once the precoders for a set of users have been calculated, it is trivial to predict the SINR that those users will experience in a transmission, if the interference covariance matrix is known. The part of the interference due to multi-user transmission is easily known. But the rest, due to interference from non-serving cells, will be obtained processing the implicit measures, or using explicit measures in a non-standard operation mode.

4.5 Conclusions

This chapter has addressed the topic of Link Adaptation in LTE. In particular, the differences between implicit and explicit CSI feedback have been presented. Moreover, LTE implicit feedback has been analysed showing that its calculation is an opened issue, especially the aspects related to how interference is considered. Several questions have been raised, including the interest of the interference spatial distribution in the calculation of the PMI, and the need for time and frequency interference averaging in this calculation.

Then, it has been detailed how the eNodeB can make use of various indicators sent by the user to perform the Link Adaptation. Correction processes employed in this Thesis, including ILLA and OLLA loops, have been explained. Those processes use other sources of information to correct the indications provided by CSI.

Finally, two precoder schemes have been detailed for non-codebook based MU-MIMO transmission: block diagonalization and SLNR precoding. The former tries to maximize the useful signal while nulling the interference. The latter tries to maximize the signal while minimizing the interference to others. Both approaches are appealing and will be evaluated in the following chapters.

Block diagonalization has been particularized for transmission of 1 layer per user. This particular configuration was widely used in the framework of International Mobile Telecommunications Advanced (IMT-Advanced) evaluation where it was very useful to fulfil the performance requirements as the following chapters will show.

Chapter 5

LTE Scheduling Analysis

This chapter provides a deeper understanding of the scheduling process in the downlink of Long Term Evolution (LTE). The chapter focuses on a specific family of algorithms that is commonly used in LTE based on the proportional fairness concept. Proportional Fair Scheduling (PFS) has been thoroughly used in LTE due to its ability to provide a good trade-off between cell spectral efficiency and user fairness. Current algorithms provide sub-optimum solutions at a low computational cost. Nevertheless, an analysis of current algorithms reveals several drawbacks. Some of them are related to the incomplete awareness of LTE specific features while others are related to the division of the algorithm into independent tasks. This chapter presents some modifications of these algorithms to alleviate their problems. Additionally, enhancement of interference variability due to the use of PFS is evaluated and an algorithm to minimize this variability is studied. This set of modifications does not consider Multi-User MIMO (MU-MIMO) transmission, which is studied in a specific section where a detailed MU-MIMO capable algorithm is provided and evaluated.

The chapter has been divided into the next sections:

- Section 5.1 introduces the scheduling process and its different objectives.
- Section 5.2 analyses in detail the concept of proportional fairness.
- Section 5.3 describes a typical implementation of a proportional fair scheduler for Single Input Multiple Output (SIMO) and Single User MIMO (SU-MIMO). A deep analysis of its advantages and drawbacks is provided.
- Section 5.4 proposes and evaluates two modifications to the typical implementation of the proportional fair scheduler aiming at increasing fairness.

- Section 5.5 proposes and evaluates a modification to the typical implementation of the proportional fair scheduler aiming at reducing the interference variability and the consequent failures in link adaptation.
- Section 5.6 details a MU-MIMO capable adaptation of the typical implementation of the proportional fair scheduler, which follows the ideas of the evaluators of International Mobile Telecommunications Advanced (IMT-Advanced) candidate technologies, and paves the way to the integration with Coordinated Multi-Point (CoMP) in the following chapter.
- Section 5.7 outlines the main conclusions of this chapter.

5.1 Introduction

Packet scheduling is the LTE resource management function whose main aim is to allocate radio resources to users [92]. Many scheduling algorithms have been proposed to optimize system performance. Among them, PFS provides a good trade-off between cell spectral efficiency and user fairness maximization due to a distribution of resources among users that takes into account both the channel quality and the previously experienced throughput [93–95]. Nowadays, most of the real implementations of schedulers are based on this PFS philosophy.

In Third Generation Partnership Project (3GPP) LTE standard, as a Multiple-Input Multiple-Output (MIMO)-Orthogonal Frequency Division Multiple Access (OFDMA) technology, resources to be distributed by the packet scheduler are spread over time, frequency, space and power domains [92]. Concerning the time and frequency domains, in LTE physical layer, it is defined the Resource Block (RB) that consists of a number of contiguous subcarriers in the frequency domain during a slot of 0.5 ms. The smallest resource allocation unit in the time-frequency domain is a pair of RBs contiguous in time during a subframe (1 ms) but not necessarily occupying the same set of subcarriers. Space domain management capability depends on the configured multi-antenna transmission. If space domain is available, several spatial streams can be allocated to a user and also several users can be spatially multiplexed. The last allocable domain is the power domain. The LTE scheduler manages the transmission power available at the base station. In this chapter, it is considered that power allocation is uniform over all RBs.

One peculiarity of LTE is that, in this technology, the scheduler decides not only on the resource allocation but also on the transmission characteristics. Specifically, in downlink, the scheduler chooses the modulation, coding, number of streams, spatial processing, priority between transmissions and/or retransmission and even the distribution of flows into Hybrid ARQ (HARQ) processes. LTE presents two particular characteristics related to the selection of the transmission format [78]. First, the standard indicates that all RBs allocated to a user in a subframe must use the same Modulation and Coding Scheme (MCS). This MCS-constraint affects multi-carrier transmission performance. Second, it is a common assumption in LTE-related research that the same number of RBs and even the same MCS used in the first transmission are used in case of retransmission since this reduces the receiver complexity. This assumption will be referred to as the HARQ-constraint.

The high complexity of optimal PFS has encouraged the development of sub-optimal scheduling algorithms. In LTE, most algorithms are based on decoupling the scheduling problem into two phases as proposed in [96]. In the first phase, the Time Domain Scheduling (TDS), a subset of users is selected following a given criterion. In the second phase, the Frequency Domain Scheduling (FDS), resources are allocated to the previously selected users. One positive feature of the algorithm described in [96] is that it is HARQ-aware, since it takes into account the HARQ-constraint and the resulting decoding gain due to retransmissions. One disadvantage of this algorithm is that, although TDS reduces FDS complexity, it also decreases multi-user diversity order in a non-optimized way since TDS and FDS are almost independent processes. Besides, FDS is unaware of multi-carrier transmission implications.

The following two sections present the general framework of PFS and the most common sub-optimal implementation. Based on the analysis conducted in both sections, new algorithms are proposed to increase system performance and to enable MU-MIMO transmissions in the subsequent sections.

5.2 Proportional fair scheduling

This section presents a deep analysis of PFS explaining what is the proportional fairness and how utility maximization helps to achieve proportional fairness.

5.2.1 Mathematical model of scheduling

We consider a system with a set $\mathcal{K}$ of users and a set, $\mathcal{Z}$, of RBs. Let us define Ω_k as the set of RBs allocated to user k in a scheduling interval, fulfilling $\Omega_k \subseteq \mathcal{Z}$. Consider $\mathcal{P}$ as the collection of all subsets of $\mathcal{Z}$, that is to say, $\mathcal{J} \in \mathcal{P} \leftrightarrow \mathcal{J} \subseteq \mathcal{Z}$. Then $\Omega_k \in P$.

We assume that at the time of decision making, t, the scheduler knows the average data rate (hereinafter named throughput) experienced by user k until this moment, $T_k(t)$. We also assume that a Link Adaptation (LA) module provides $R_k(\mathcal{J}, t)$, defined as the estimation of the data rate that user k can achieve in a transmission over the set of resources $\mathcal{J}$ during the scheduling interval that starts at time t.

Once the scheduling decision is made, a specific data rate is assigned to each user. Let us define $\widehat{R}_k(t)$ as the assigned data rate to user k in the scheduling interval that starts at time t. Note that $\widehat{R}_k(t) = 0$ if user k is not selected for transmission.

Hereinafter, variable t is omitted if an undefined scheduling time is considered. Also, the set of resources $\mathcal{J}$ is omitted if a system consists of only one resource.

5.2.2 Proportional fairness

The concept of proportional fairness was originally proposed by Kelly [97]. According to Kelly, the rates assigned to a set of users $\mathcal{K}$, $\left\{\widehat{R}_1, ..., \widehat{R}_{|\mathcal{K}|}\right\}$, is proportionally fair if it is feasible and if for any other feasible set, $\left\{\widehat{R}^*_1, ..., \widehat{R}^*_{|\mathcal{K}|}\right\}$, the aggregate of proportional changes is zero or negative:

$$\sum_{k=1}^{|\mathcal{K}|} \frac{\widehat{R}^*_k - \widehat{R}_k}{\widehat{R}_k} \leq 0. \tag{5.1}$$

Kelly demonstrated that this criterion is equivalent to the solution of the next problem:

$$\text{maximize} \sum_{k=1}^{|\mathcal{K}|} \log\left(\widehat{R}_k\right), \tag{5.2}$$

which is a maximization of the aggregate of user utilities defined as the logarithm of assigned user rates. It is remarkable that this utility function provides a trade-off between maximization of the aggregate of rates and maximization of the fairness among users.

In order to understand the effect of this criterion on the resource allocation of a wireless system, let us consider a system with $|\mathcal{K}|$ users in which resource allocation allows the system bandwidth to be divided into parts of whatever size. We assume that the system is not frequency selective and that it is available an estimate of the rate that each user can achieve performing a transmission over the whole system bandwidth, R_k. We also assume that if the fraction of resources allocated to user k is α_k, the assigned rate, $\widehat{R}_k$ is equal to

$R_k\alpha_k$. Then, the previously presented problem can be reformulated as:

$$\begin{aligned} \underset{\alpha_k}{\text{maximize}} \quad & \sum_{k=1}^{|\mathcal{K}|} \log\left(R_k\alpha_k\right) \\ \text{subject to} \quad & R_k \geq 0, \\ & 0 \leq \alpha_k \leq 1, \\ & \sum_{k=1}^{|\mathcal{K}|} \alpha_k = 1. \end{aligned} \tag{5.3}$$

This proportional fairness problem leads to a decision in which the system bandwidth is uniformly distributed among users, that is to say, the optimal fraction of resources is $\alpha_k = \frac{1}{|\mathcal{K}|}$. In this situation, the vector of assigned rates is proportional to the vector of estimated rates.

Consider now a wireless system whose bandwidth is divided into a fixed number of resources that is not proportional to the number of users. In such a system, it is not possible to perform a uniform allocation of resources in each scheduling interval and then the assigned rates can not be proportionally fair. Nevertheless, it is possible to achieve proportional fairness among the long term throughput of the users.

Adapting the original concept, we could say that a vector of experienced throughputs $\left\{T_1(t), ..., T_{|\mathcal{K}|}(t)\right\}$ is proportionally fair if it maximizes the following utility function:

$$U_{PF}\left(\left\{T_1(t), ..., T_{|\mathcal{K}|}(t)\right\}\right) = \sum_{k=1}^{|\mathcal{K}|} \log\left(T_k(t)\right), \tag{5.4}$$

for all feasible vectors of experienced throughputs, where it is worth noting that only bounded $T_k(t)$ values are feasible. Proportional fairness utility function provides a trade-off between maximization of the aggregate of user throughputs and maximization of the fairness among users.

In this sense, a PFS was implemented for Qualcomm's High Data Rate (HDR) system in [93]. This system consists of only one transmission channel and then it is not possible to perform a proportionally fair allocation in each scheduling interval. The PFS implementation assigns in each time t the channel to the k user that maximizes the next utility function:

$$U_{HDR}(k,t) = \frac{R_k(t)}{T_k(t)}. \tag{5.5}$$

In practice, an exponential moving average is used to obtain $T_k(t)$, instead of using an exact averaging of all the past throughput values [98]. At time $t+1$, the throughput is updated according to the following equation [93]:

$$T_k(t+1) = \frac{1}{W} \cdot T_k(t) + \left(1 - \frac{1}{W}\right) \cdot \widehat{R}_k(t), \tag{5.6}$$

where W is a parameter equivalent to the length of a sliding average window.

It is shown in [95] that with sufficiently large value of t and W, $T_k(t)$ weakly converges to a constant value for a certain user k. Moreover, it is demonstrated that the vector of average throughput values converges to a proportionally fair solution that solves the problem presented in (5.4). In order to drawn such conclusion, fairly accurate rate predictions are assumed in [95]. Concerning the value of W, it has been shown that, in practice, it should be chosen to offer a good estimation of the average throughput, with the ability to track changes in the channel characteristics [95] (e.g. 100 slots in [94], 1000 slots in [93]).

Previously it has been noted that the effect of a proportionally fair allocation over the vector of assigned rates is to obtain a vector proportional to the vector of achievable rates. Note that the strategy defined by (5.5) tends to achieve a similar equilibrium, that is to say, a vector of experienced throughput values proportional to the vector of achievable rates. Consider a situation with two users that present the same achievable rate, then the user with lower experienced throughput will be prioritized. This user will tend to increase its experienced throughput, while the other user will experience a decrease in its throughput. As a consequence, the proportion among achievable rate and experienced throughput will tend to be equal achieving the mentioned equilibrium. Besides, it can be deduced that a user will be scheduled with more probability when its achievable throughput presents high peaks compared with the experienced throughput.

In [95] an extension of the HDR proportional fair scheduling to multi-carrier systems is discussed. A simple strategy is proposed in which the HDR PFS is applied independently to each RB. Therefore, each resource r is allocated to the user k maximizing the next utility function:

$$U_{MC_a}(k, r, t) = \frac{R_k(r,t)}{T_k(t)}. \tag{5.7}$$

Additionally, in [95] a more complete strategy is also presented in which it is considered that the data rate of a joint transmission over multiple resources can be different from the aggregate of the data rates of multiple transmissions performed over independent resources. The strategy can be translated to an algorithm that assigns at each time t the channel according to an allocation Ω_k

to maximize the next utility function:

$$U_{MC_b}(t) = \sum_{k=1}^{|\mathcal{K}|} \frac{R_k(\Omega_k, t)}{T_k(t)}. \tag{5.8}$$

Note that if the number of channels and/or the number of schedulable users is 1, both strategies are equivalent to that represented by Equation (5.5). It is worth noting that, according to [95], the latter strategy presents the same convergence properties as the original HDR PFS. However, the first strategy is clearly the worst since it does not consider properly multicarrier transmission performance.

In [99], the proportional fair scheduling proposed for HDR is also extended to multi-carrier systems. The authors take into consideration that proportional fair scheduling tends to maximize the expression shown in Equation (5.2). The authors try to find an algorithm whose main aim is to maximize Equation (5.2) at each scheduling interval. The new algorithm implementation assigns at each time t the channel according to an allocation Ω to maximize the next utility function:

$$U_{MC_c}(t) = \prod_{k=1}^{|\mathcal{K}|} \left(1 + \frac{R_k(\Omega_k, t)}{(W-1)T_k(t)}\right). \tag{5.9}$$

When the number of channels and/or the number of schedulable users is 1, this expression is equivalent to that of Equation (5.5). Expressions (5.9) and (5.8) are different, but it can be demonstrated that both tend to be equivalent if $t \to \infty$ and $W \to \infty$.

5.3 The D-PFS algorithm

This chapter proposes new scheduling algorithms based on an LTE scheduler implementation that has become a *de facto* standard, i.e. the Decoupled PFS (D-PFS) algorithm. This algorithm is based on the work of Pokhariyal *et al.* [96]. Other studies have followed a similar approach (see [92], e.g.). The main characteristic of D-PFS operation is that it is divided into two phases: the TDS and the FDS.

First, the TDS selects a subset of up to $|\mathcal{C}|_{\max}$ users from the set of users $\mathcal{K}$ requiring resources. This is achieved after a prioritization of users according to a proportionally fair utility. The aim of this TDS selection is to decrease the FDS complexity and the signalling overhead required by user multiplexing. Additionally, TDS allows a high Quality of Service (QoS) control. A peculiarity of the D-PFS is that it is assumed that TDS does not know how FDS allocates resources (they are independent processes). Therefore, user utilities are calculated assuming full-bandwidth transmissions. Besides, in TDS, prioritization of users does not care about pending retransmissions.

TDS phase is described in detail in Algorithm 1. Recall that $\mathcal{K}$ is the set of $|\mathcal{K}|$ users of the system. After TDS, $\mathcal{C}$, the set of candidate users for resource allocation, is formed. This set is divided into two sets: $\mathcal{C}_N$ and $\mathcal{C}_R$, such as $\mathcal{C} = \mathcal{C}_N \cup \mathcal{C}_R$, being $\mathcal{C}_N$ the set of users that are candidates to perform new transmissions and $\mathcal{C}_R$ the set of users that are candidates to perform retransmissions. Let us denote Q_k and $Q_{\mathcal{C}_R}$ as the number of resources required by the user k and the user set $\mathcal{C}_R$, respectively, to perform retransmissions. In Algorithm 1, a function $\text{argpos}_k(f(k), i, \mathcal{J})$ has been used, as the function returning the k^* index fulfilling that $f(k^*)$ would be the i-th value if the values of $f(k)$ with $k \in \mathcal{J}$ were sorted in descending order. In order to make FDS feasible, operation of TDS presents some restrictions, for example, $Q_{\mathcal{C}_R}$ can not be greater than Q. Besides, $|\mathcal{C}| = |\mathcal{C}_N| + |\mathcal{C}_R|$ can not be greater than $|\mathcal{C}|_{\max}$, the maximum number of candidate users. These restrictions are fulfilled by the algorithm described in Algorithm 1.

After TDS, the FDS is executed. In this phase, resources are allocated to the $|\mathcal{C}|$ candidate users. Users with retransmissions are prioritized (arguing the minimization of transmission delay) and their required resources are guaranteed. Specifically, assuming the presence of $|\mathcal{C}_R|$ users with pending retransmissions that jointly require $Q_{\mathcal{C}_R}$ of a total of Q RBs, $Q_{\mathcal{C}_N} = Q - Q_{\mathcal{C}_R}$ RBs are firstly allocated to the $|\mathcal{C}_N| = |\mathcal{C}| - |\mathcal{C}_R|$ users with new data, guaranteeing the availability of Q_R blocks for retransmissions. Resources for new transmissions are allocated according to proportionally fair utilities calculated per user and RB as in Equation (5.7). On the other hand, resources for retransmissions are distributed according to only the channel quality of each RB. Although this mechanism (that allocates resources for new transmission first) disfavours retransmissions, it is stated that this fact is counteracted by the decoding gain due to the soft combining of HARQ retransmissions.

Algorithm 2 describes the FDS phase. Recall that $\mathcal{Z}$ denotes the set of system resources, and let us define Θ as the set of unallocated resources. Initially, Θ is equal to $\mathcal{Z}$ since no resources are allocated. Function $\text{argpos}_{(k,r)}(f(k,r), i, \mathcal{J}1, \mathcal{J}2)$ has been used denoting the function returning the (k^*, r^*) pair fulfilling that $f(k^*, r^*)$ would be the i-th value if the values of $f(k,r)$ with $k \in \mathcal{J}1$ and $r \in \mathcal{J}2$ were sorted in descending order. During FDS, $\mathcal{S}$, the set of scheduled users, is formed. This set is divided into two sets: $\mathcal{S}_N$ and $\mathcal{S}_R$, being $\mathcal{S}_N$ the set of users scheduled to perform new transmissions and $\mathcal{S}_R$ the set of users scheduled to perform retransmissions. It is worth noting that, with the algorithm described in Algorithm 2, $\mathcal{S}_R = \mathcal{C}_R$, that is to say, candidates for retransmission have always enough resources to be scheduled.

Algorithm 1 TDS phase

for each user k in $\mathcal{K}$ **do**
  Calculate $U_w(k)$
end for
$\mathcal{C}_R = \emptyset, Q_{\mathcal{C}_R} = 0$
$\mathcal{C}_N = \emptyset$
for $i = 1$ to $|\mathcal{K}|$ **do**
  $(k^*) = \underset{k}{\text{argpos}}(U_w, i, \mathcal{K})$
  if user k^* has pending retransmissions **then**
    if $Q_{\mathcal{C}_R} + Q_{k^*} <= Q$ **then**
      $C_R \leftarrow k^*$
      $Q_{\mathcal{C}_R} = Q_{\mathcal{C}_R} + Q_{k^*}$
    else
      $\mathcal{C}_N \leftarrow k^*$
    end if
  else
    $\mathcal{C}_N \leftarrow k^*$
  end if
  if $|\mathcal{C}_N| + |\mathcal{C}_R| = |\mathcal{C}|_{\max}$ **then**
    break
  end if
end for

One drawback of D-PFS is related to the utility function used in the FDS phase. Note that the used utility function, that is, Equation (5.7), is not the most complete one, captured in Equation (5.8). Both equations would be equivalent if and only if the achievable data rate in a multi-carrier transmission was just the addition of the achievable data rates in multiple transmissions over single resources. Nevertheless, this is not true in LTE, among other reasons, because the same MCS must be used in all RBs allocated to a user. In practice, it means that, when the data rate estimate $R_k(\mathcal{J}, t)$ is calculated over a set of resources $\mathcal{J}$, the estimated data rate per resource $R_k(\mathcal{J}, t)/|\mathcal{J}|$ may be lower than the data rate estimated for some resources (underutilized resources) and higher for others (limiting resources). Two ideas arise from this analysis. First, if the value of $R_k(\mathcal{J}, t)$ was highly conditioned by the resource with the worst quality, then it would be useful for the user to get rid of this RB. Moreover, another user could get the unallocated RB to improve its own utility value. The second idea is that, even if a user cannot improve its utility by getting rid of an RB, it is yet possible to achieve a better scheduling decision through

Algorithm 2 FDS phase

$\mathcal{S}_N = \emptyset, \mathcal{S}_R = \emptyset$
$\Theta = \mathcal{Z}$
{Allocation of resources to new transmissions}
for each k in $\mathcal{C}_N$ **do**
for each r in $\mathcal{Z}$ **do**
Calculate $U_N(k,r) = R_k(r)/T_k(r)$
end for
end for
$i = 1$
while $|\Theta| > Q_{\mathcal{C}_R}$ **do**
$(k^*, r^*) = \underset{k,r}{\text{argpos}}(U_N(k,r), i, \mathcal{C}_N, \mathcal{Z})$
if $r^* \in \Theta$ **then**
$\Omega_{k^*} \leftarrow r^*$
$\Theta = \Theta - r^*$
$\mathcal{S}_N = \mathcal{S}_N \cup k^*$
end if
$i = i + 1$
end while
{Allocation of resources to retransmissions}
for each k in $\mathcal{C}_R$ **do**
for each r in $\mathcal{Z}$ **do**
Calculate $U_R(k,r) = R_k(r)$
end for
end for
$i = 1$
while $Q_{\mathcal{S}_R} < Q_{\mathcal{C}_R}$ **do**
$(k^*, r^*) = \underset{k,r}{\text{argpos}}(U_R(k,r), i, \mathcal{C}_R, \mathcal{Z})$
if $r^* \in \Theta$ and $|\Omega_{k^*}| < Q_{k^*}$ **then**
$\Omega_{k^*} \leftarrow r^*$
$\Theta = \Theta - r^*$
$\mathcal{S}_R = \mathcal{S}_R \cup k^*$
end if
$i = i + 1$
end while
$\mathcal{S} = \mathcal{S}_N \cup \mathcal{S}_R$

the transfer of an RB from one user to another. This can be achieved via the transfer of a bad RB that produces an underutilization of capacity in other RBs, or through the transfer of a good RB with underutilized capacity that could be better used by another user.

The second drawback of the D-PFS is that TDS operation produces an obvious reduction in the multi-user gain of the final scheduling decision. The effect of this drawback would be minimized if TDS prioritized users in such a way that the final decision was similar to that of the strategy presented in Equation (5.8). In order to get the most of the opportunistic scheduling performed by FDS, it could be positive that TDS prioritized users with very high utility peaks, as FDS tends to do. Nevertheless, the TDS prioritizes users according to full bandwidth metrics. In conclusion, it is envisaged that D-PFS could be improved by increasing the coupling between TDS and FDS, making TDS prioritize users in a way similar to how FDS allocates resources.

HARQ-awareness

The HARQ manager of a base station manages several HARQ entities, one per served user, each one of them comprising multiple HARQ processes. This manager knows if a user has new transmissions and/or any pending retransmission and can establish priorities among them. At the end, the HARQ protocol has a set of features that can be considered in the scheduling process to obtain feasible and efficient decisions. Therefore, it is important to go beyond the limited D-PFS HARQ-awareness.

In LTE downlink, HARQ is an asynchronous protocol. This means that the scheduler has total freedom to schedule any HARQ process at any time. This freedom enables the definition of multiple policies to manage the HARQ processes. For example, the scheduler can prioritize HARQ processes with pending retransmissions as in D-PFS. However, this solution is not oriented to maximize the proportional fairness but to the minimization of packet delay. Other policies to prioritize HARQ processes more related to the data rate could be investigated.

On the other hand, LTE downlink HARQ is adaptive, in the sense that retransmissions may change the transmission format previously used. Resource block allocation –including the number and the set of resource blocks,– modulation, coding or redundancy version can be changed in retransmissions. Nevertheless, for simplicity reasons the number of blocks and the MCS are kept constant in subsequent retransmissions [96]. This fact poses tight constraints to the allocation of resources to retransmission. D-PFS considers this by limiting the resources for new transmissions so as to make room for pending retransmissions.

Finally, the HARQ protocol allows for soft combining of retransmissions with previous transmissions to increase the probability of correct decoding at the receiver. In D-PFS this fact is taken into account allocating best resources to new transmissions provided the increased quality of retransmitted data. In [100] authors stated that this kind of prioritization performs well under good signal level conditions. Otherwise, it is suggested to allocate first the best resources to retransmissions.

5.4 D-PFS modifications for increased proportional fairness

Based on the analysis of the D-PFS, this Thesis proposes two algorithms that incrementally include new features to increase the proportional fairness of the decisions.

5.4.1 Decoupled and MCS constraint Aware PFS

The first assessed algorithm is the Decoupled and MCS constraint Aware PFS (DMA-PFS). This algorithm is based on the D-PFS but includes the MCS-constraint awareness, and hence, multi-carrier awareness. This algorithm is applied to the users with new transmissions. In summary, after the normal operation of the D-PFS FDS an iterative process is conducted to exchange resources among users, and even to leave RBs unallocated, with the aim of maximizing the scheduling utility.

The exact procedure is described in Algorithm 3. For each selected user (from $\mathcal{S}_N$), it is found the RB r_w with worst (highest negative) impact on its utility, in the sense that if this RB was unallocated the utility would present the highest increase (even if this increase is negative). This increase is referred to as 'donor user utility increase', ΔU_d. Next, it is found the user k_r, that being the recipient of r_w, would have the highest utility increase. This increase is referred to as 'recipient user utility increase', ΔU_r. It is worth noting that this user may not belong to user set $\mathcal{S}_N$. Then, if total utility can be increased, the RB is allocated to the best recipient user. Otherwise, it would remain unallocated. Finally, for each unallocated RB it is found the user that getting this resource maximizes the user utility increase. If the maximum user utility increase is positive, then the resource is allocated. The process is repeated for all users if, in a previous execution, any allocation change is done.

Algorithm 3 MCS constraint aware enhancement

```
Initialize $continue = true$
while $continue = true$ do
    $continue = false$
    for each $k_d$ in $\mathcal{S}_N$ do
        $r_w = \Omega_{k_d}(\arg\max_j U(k_d, \Omega_{k_d} - \Omega_{k_d}(j)))$
        $\Delta U_d = U(k_d, \Omega_{k_d} - r_w) - U(k_d, \Omega_{k_d})$
        $k_r = \arg\max_{k_t} (U(k_t, \Omega_{k_t} + r_w) - U(k_t, \Omega_{k_t}))$
        $\Delta U_r = U(k_r, \Omega_{k_r} + r_w) - U(k_r, \Omega_{k_r})$
        if $\Delta U_d > 0$ or $\Delta U_d + \Delta U_r > 0$ then
            unallocate $r_w$: $\Omega_{k_d} \leftarrow \Omega_{k_d} - r_w$ and $\Theta \leftarrow \Theta + r_w$
            if $\Delta U_r > 0$ then
                allocate $r_w$: $\Omega_{k_r} \leftarrow \Omega_{k_r} + r_w$ and $\Theta \leftarrow \Theta - r_w$
            end if
            if $|\Theta| > 0$ then
                for each $r$ in $|\Theta|$ do
                    $k_r = \arg\max_{k_t} (U(k_t, \Omega_{k_t} + r) - U(k_t, \Omega_{k_t}))$
                    $\Delta U_r = U(k_r, \Omega_{k_r} + r) - U(k_r, \Omega_{k_r})$
                    if $\Delta U_r > 0$ then
                        allocate $r$: $\Omega_{k_r} \leftarrow \Omega_{k_r} + r$ and $\Theta \leftarrow \Theta - r$
                    end if
                end for
            end if
            $continue = true$
        end if
    end for
end while
```

5.4.2 Coupled and MCS constraint Aware PFS

The Coupled and MCS constraint Aware PFS (CMA-PFS) algorithm is based on the DMA-PFS but includes a tighter coupling between TDS and FDS. It is said that TDS and FDS are coupled because some knowledge is incorporated to TDS about FDS allocation in order to calculate TDS priorities.

The exact procedure is described in Algorithm 4. First, utilities are calculated for each RB r and for each user k, assuming that a new transmission will be performed independently over each RB. Then, each resource is tentatively allocated to the user with highest utility for this resource, identified by k_{best}.

Users with tentatively allocated resources are grouped in a high priority group and their priorities, P_k^H, are calculated as the aggregate of the utilities of the resources tentatively allocated. Users without tentatively allocated resources are included in a low priority group and their priorities, P_k^L, are calculated as the maximum resource utility of each user. Finally, up to $|\mathcal{C}|_{\max}$ users with the highest priorities are selected from the high priority group and if there is still room for more users those come from the low priority group.

Note that HARQ-awareness is not considered in CMA-PFS.

Algorithm 4 TDS-FDS coupling enhancement

Calculate $U(k,r) = R_k(r)/T_k(r)$
for each r **do**
  $k_{best} = \underset{k}{\operatorname{argmax}}\, U(k,r)$
  $\Omega_{k_{best}} \leftarrow \Omega_{k_{best}} + r$
end for
for each user k **do**
  if $|\Omega_k| > 1$ **then**
    $P_k^H = \sum_{i=1}^{|\Omega_k|} U(k, \Omega_k(i))$
  else
    $P_k^L = \max(U(k,\Omega_k(1)), \ldots, U(k, \Omega_k(|\Omega_k|)))$
  end if
end for
Select up to $|\mathcal{C}|_{\max}$ users

5.4.3 Assessment methodology

Assessment methodology of this work was based on system level simulations. The proposed algorithms were tested in the Indoor hotspot (InH) environment extracted from [7]. It considers an indoor scenario with a long hall with adjacent offices. Users are pedestrians and two base stations operating at 3.4 GHz with an omnidirectional antenna setup are mounted on the ceiling of the corridor. The IMT-Advanced stochastic and geometric channel model was used in the simulations.

Concerning LTE configuration, Channel Quality Indicator (CQI), Precoding Matrix Indicator (PMI) and Rank Indicator (RI) were reported with 5 ms period and with a frequency granularity of 5 RBs. Using this feedback, a LA algorithm was conducted that included an Outer-Loop Link Adaptation (OLLA) to control CQI estimation errors maintaining the Block Error Rate (BLER) of new transmissions at a target value of 20% [48].

Base stations had 4 transmitting antennas, while users had 2 receiving antennas. Antennas were considered to be vertically polarized and uniformly spaced 0.5 wavelengths. Two MIMO schemes were considered: SIMO and SU-MIMO using codebook-based closed-loop spatial multiplexing with dynamic rank adaptation based on user's feedback.

Additional assumptions and models of the simulation methodology are summarized in Table 5.1.

Table 5.1: Simulation assumptions and models.

Simulation length / runs	2 s / 100 runs
Cell selection	1 dB handover margin
Traffic model	full-buffer
Interference model	explicit model
CSI feedback	realistic
SINR estimation error	1 dB lognormal error per RB
Control channel overhead	3 OFDM symbols per subframe
PFS W	1000
PFS K_S	5

Several performance indicators are used in this evaluation. The Cell Scheduling Utility (CSU) is used to measure how proportionally fair is an algorithm. It is calculated as:

$$\mathrm{CSU} = \frac{\sum_{c=1}^{C} \sum_{k=1}^{K_c} \log(T_{c,k})}{C}, \tag{5.10}$$

where C is the total number of cells simulated (in all simulation runs), K_c is the number of users served by cell c and $T_{c,k}$ is the average throughput experienced by user k of cell c during a simulation run. Cell Spectral Efficiency (CSE) measures the spectral efficiency of the whole system. This performance indicator is defined as:

$$\mathrm{CSE} = \frac{\sum_{c=1}^{C} \sum_{k=1}^{K} T_{c,k}}{C \cdot \omega}, \tag{5.11}$$

where ω is the channel bandwidth used per cell. Cell-Edge User Spectral Efficiency (CEUSE) is defined as the 5% point of the Cumulative Distribution Function (CDF) of the User Spectral Efficiency (USE), being the USE defined as the throughput of a user divided by the channel bandwidth.

The optimum proportional fair scheduling provides the highest CSU value. Concerning the rest of performance indicators, it would be desirable an increase of them since it would mean an increase of spectral efficiency and fairness.

5.4.4 Results

In Figure 5.1 it is represented the gain achieved by DMA-PFS and CMA-PFS over the CSU of the D-PFS algorithm, both for SIMO and SU-MIMO. Results show an improvement in CSU with both new algorithms. This means that the throughput distribution produced by both algorithms is more proportionally fair than that of the D-PFS. It also means that both multi-carrier-awareness and coupling are positive features. CMA-PFS algorithm provides the best performance since it includes both features.

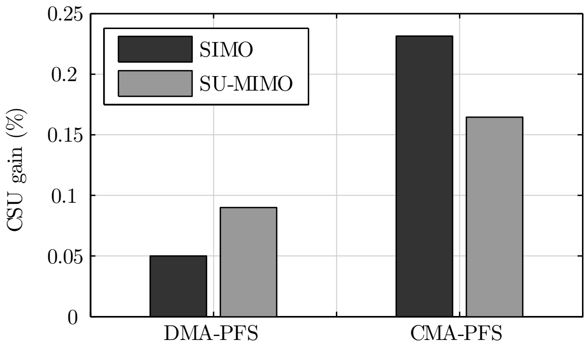


Figure 5.1: CSU gain obtained with the proposed algorithms. IEEE © 2015.

Concerning the rest of performance indicators, it would be desirable an increase of them. In fact, results depicted in Figures 5.2 and 5.3 show that CEUSE and CSE are improved with the new algorithms. Indeed, CEUSE presents very high gains, from 9% to 20% for the CMA-PFS. At the same time, CSE does not decrease but presents gains between 1.4% and 2%. This is a very remarkable fact since it is easy to improve the CEUSE of a system with a scheduling that gives higher priority to users with low throughput values. However, it usually comes with the cost of a reduced CSE. Nevertheless, the proposed algorithms provide CEUSE and CSE gain at the same time.

Two additional facts are remarkable. First, results have been obtained in a realistic simulation setup. For example, rate estimation errors have been taken into account. Therefore, it is not strictly necessary to have perfect rate estimates to obtain performance improvements from the presented algorithm, what makes them really appealing. Second, both in SIMO and SU-MIMO performance gains have been obtained, what makes the algorithms widely useful in LTE deployments.

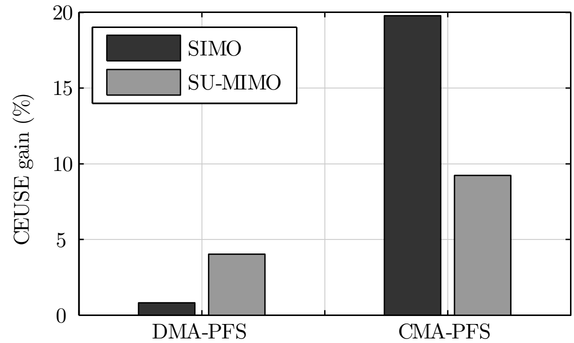


Figure 5.2: CEUSE gain obtained with the proposed algorithms. IEEE © 2015.

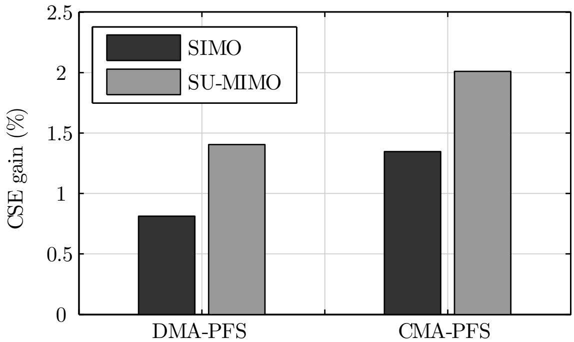


Figure 5.3: CSE gain obtained with the proposed algorithms. IEEE © 2015.

5.5 D-PFS modification for interference stabilization

This section presents a modification of the PFS algorithm that stems from the observation of an effect known as the flash-light effect. This effect manifests as a rapid variation of the interference power and spatial distribution due to fast changes in scheduling decisions. In the simulations conducted in this chapter, only full-buffer traffic is used. This means that all RBs are constantly being used. Besides, the power used in all the RBs is the same, since power control is not applied in the downlink. Therefore, in this chapter, the flash-light effect is only produced by the fast change of the precoders selected in consecutive scheduling decisions.

The flash-light effect comes together with an increase of link adaptation failures since the interference measured in one subframe is probably not equal to the interference experienced in a future transmission. These failures have a detrimental impact on system performance. Therefore, it is necessary to minimize this problem as much as possible.

With this aim, precoders fast variation must be first understood. Two actors are involved in the allocation of precoders to RBs. One actor is the link adaptation and the other is the scheduling. As commented in Chapter 4, in codebook-based transmission modes, PMI feedback is used to select the precoder for each user. In Chapter 4, it was also commented that PMI calculation may consider interference spatial distribution or not, and may consider an averaging of interference over time or just an instantaneous value. Depending on the decisions made about how to consider the interference, the reported PMI may change faster or slower. Concerning the scheduling, this algorithm may allocate a set of resources to a user in one subframe and to another user in the next subframe, based on, for example, the maximization of a utility function and need of resources for retransmissions. As a consequence, scheduling makes precoders allocations change even if PMI are stable.

Figure 5.4 illustrates the flash-light effect. In the upper part of the figure, a mobile terminal is measuring its channel quality to report it to its serving cell. The mobile terminal measures and reports a good quality because the interfering cell is using a precoder that produces low interference to the mobile terminal. In the lower part, it is represented a possible scenario found when a data transmission to the terminal is done by its serving cell. This cell may have adapted the transmission according to the good channel quality report of the mobile terminal. However, in this scenario, the interfering cell is now using a precoding that produces a high interference to the mobile terminal. Therefore, there is a probable transmission failure.

In this section, as in the previous one, we consider that the PMIs are calculated considering the spatial distribution of interference and without averaging.

One strategy that may be used to minimize the impact of the flash-light effect is to enforce the scheduling decision to be more stable. Osseiran uses in [101] an algorithm that divides the whole system bandwidth into subbands allocating one precoder per subband during a period of time. This way, interference spatial distribution is stable during, at least, the precoding allocation period. During this period of time, allocation of resources to users may change, but in order to be efficient, users should be allocated the resources whose corresponding precoder is good for them. This approach reduces the scheduling flexibility and may decrease the system performance. A possible solution would be to make the precoder allocation taking into account the proportion of users

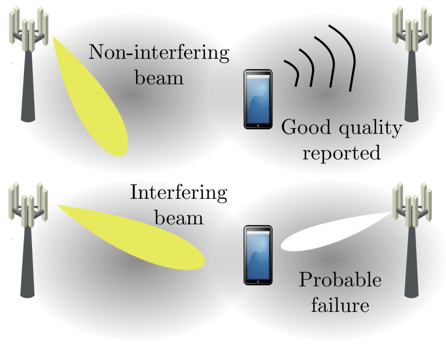


Figure 5.4: Flash-light effect illustration.

preferring each precoder, but it could be a difficult task. Another concern of this strategy is that retransmissions are not explicitly considered.

The approach proposed in this Thesis, known as Interference Stabilization Proportional Fair Scheduler (IS-PFS), has the objective of reusing the traditional PFS algorithm as much as possible. It consists of the modulation of the proportional fairness utility function with a factor that takes into account how much the preferred precoder of the user whose utility is being calculated matches the precoder used in the resource where the utility is being calculated in the previous allocation of resources.

The equation to calculate the new utility function is:

$$U_{IS-MC_a}(k, r, t) = \frac{R_k(r, t)}{T_k(t)} \times M(\mathbf{W}_{pref_k}, \mathbf{W}_{old_r}), \tag{5.12}$$

where the term $\frac{R_k(r,t)}{T_k(t)}$ is exactly the utility function expressed in Equation (5.7), and the term $M(\mathbf{W}_{pref_k}, \mathbf{W}_{old_r})$ is a matching measure between the preferred precoder of user k, $\mathbf{W}_{pref_k}$, and the precoder used in the previous scheduling in resource r, $\mathbf{W}_{old_r}$.

One possible matching function that takes into account the difference between the products $\mathbf{W}_{pref_k}\mathbf{W}_{pref_k}^{\mathrm{H}}$ and $\mathbf{W}_{old_r}\mathbf{W}_{old_r}^{\mathrm{H}}$ is:

$$M_1(\mathbf{W}_{pref_k}, \mathbf{W}_{old_r}) = \frac{1}{1 + \left\| \mathbf{W}_{pref_k}\mathbf{W}_{pref_k}^{\mathrm{H}} - \mathbf{W}_{old_r}\mathbf{W}_{old_r}^{\mathrm{H}} \right\|_2^2}, \tag{5.13}$$

where the relevance of the difference of these products is that the product of the precoder by its hermitian is a fundamental part of the interference covariance matrix used to model the interference (see, e.g., Equation (3.15)).

The use of the new utility function to make scheduling decisions promotes the selection of users that present a high proportionally fair utility at the same time that its preferred precoder is similar to the precoder used in previous scheduling decisions. The effect of this approach is a stabilization of the interference dynamics that does not preclude changes that attain high gains in the scheduling utility maximization.

This approach is very simple to implement as compared to the previous technique from [101] for example, and can react rapidly to changes in the channel quality not having its strict periodicity in precoder changes.

5.5.1 Assessment methodology

The proposed algorithm has been tested in four IMT-Advanced scenarios: the InH, the Urban Micro-cell (UMi), the Urban Macro-cell (UMa), and the Rural Macro-cell (RMa).

Concerning LTE configuration, CQI, PMI and RI were reported with 5 ms period and with a frequency granularity of 5 RBs. Using this feedback, a LA algorithm was conducted that included an OLLA to control CQI estimation errors maintaining the BLER of new transmissions at a target value of 20% [48]. Both simulations with OLLA activated and deactivated have been conducted.

Base stations had 4 transmitting antennas, while users had 2 receiving antennas. Antennas were considered to be vertically polarized and uniformly spaced 0.5 wavelengths. Concerning the multi-antenna scheme, it is considered SU-MIMO using codebook-based closed-loop spatial multiplexing with dynamic rank adaptation based on user's feedback.

Additional assumptions and models of the simulation methodology are collected in Table 5.2.

Table 5.2: Simulation assumptions and models.

Simulation length / runs	10 s / 10 runs
Cell selection	1 dB handover margin
Traffic model	full-buffer
Interference model	explicit model
CSI feedback	realistic
SINR estimation error	1 dB lognormal error per RB
Control channel overhead	3 OFDM symbols per subframe
PFS W	1000
PFS K_S	10

5.5.2 Results

Figure 5.5 compares the absolute CSE values provided by the traditional D-PFS algorithm and by the IS-PFS, while Figures 5.6 and 5.7 present the comparison for the CEUSE and CSU, respectively. The OLLA, that corrects the CQI based on the information provided by HARQ acknowledgements, has been used in a set of simulations denoted with “w OLLA” in the figures. However, in other simulations, denoted with “w/o OLLA”, it has not been used.

The IS-PFS algorithm provides consistent gains for all the performance indicators over the D-PFS in all the tested scenarios, when OLLA is not used. This improvement is due to the best link adaptation achieved through the interference stabilization provided by the IS-PFS. The use of OLLA reduces the link adaptation failures forcing a given BLER and, therefore, reacts to the detrimental impact of the flash-light effect. However, it seems that OLLA interferes with the IS-PFS operation. At least in the InH and UMi scenarios the addition of OLLA to the IS-PFS provides worst results than the use of IS-PFS standalone. In these scenarios, IS-PFS would be the preferred option. On the other hand, in UMa and RMa, the joint use is positive for the CSE and the CSU, but not for the CEUSE. Being the CSU a more complete performance indicator, it could be said that in UMa and RMa scenarios the preferred option is the use of IS-PFS with OLLA. As a conclusion, results indicate that in more interference-limited scenarios the use of IS-PFS is a good option, while in less interference-limited scenarios the joint use of IS-PFS and OLLA would be the best option.

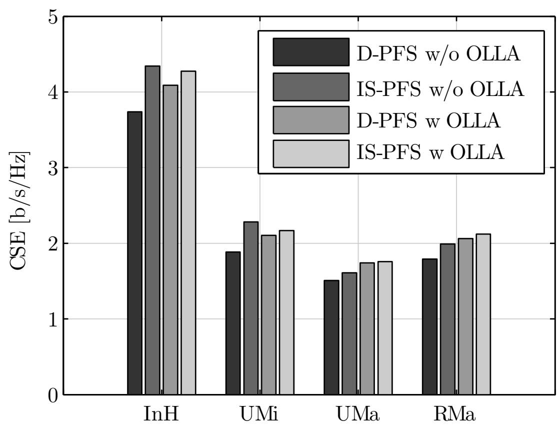


Figure 5.5: CSE of D-PFS and IS-PFS.

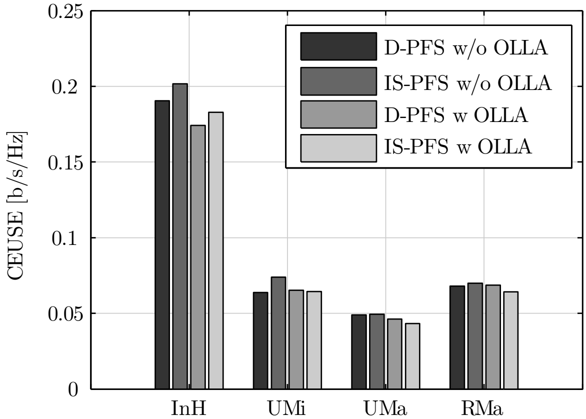


Figure 5.6: CEUSE of D-PFS and IS-PFS.

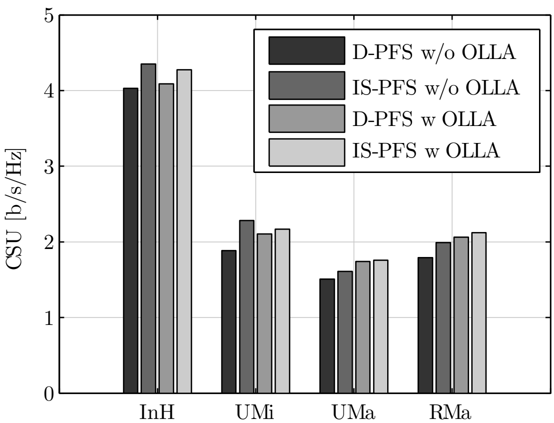


Figure 5.7: CSU of D-PFS and IS-PFS.

The joint improvement of CSE and CEUSE could be achieved without increasing the throughput experienced by most of the users of the system, i.e., increasing only the throughput of the 5% of users with lowest throughput and the throughput of users with best channel quality. Nevertheless, this is not the mode of operation of the IS-PFS. In fact, the CSU results show in all the cases an improvement in the proportional fairness when the IS-PFS is used. Indeed, the comparison of the user spectral efficiency CDFs for D-PFS and IS-PFS demonstrates that there is a efficiency improvement in most of the percentiles, as can be seen in Figures 5.8–5.11. This means that most of the users get a benefit from this technique.

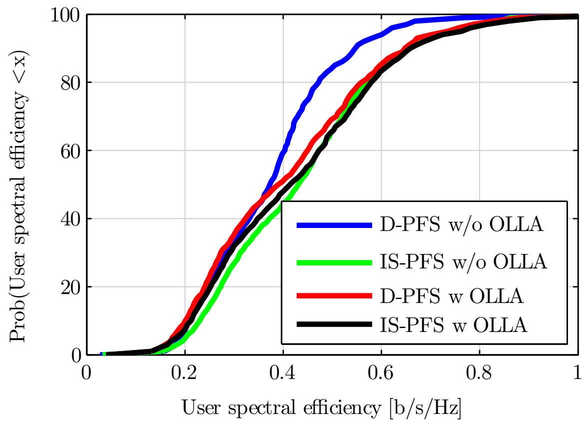


Figure 5.8: User spectral efficiency CDF in the InH scenario.

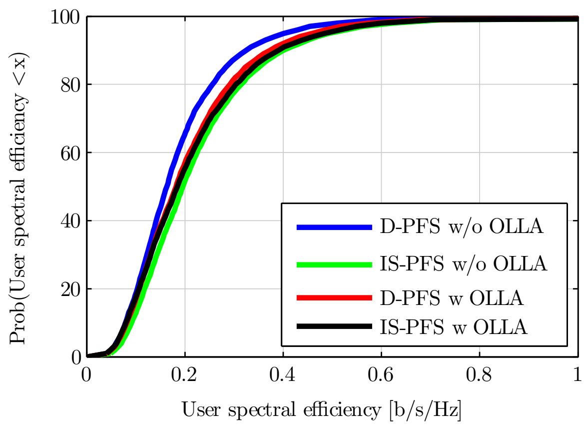


Figure 5.9: User spectral efficiency CDF in the UMi scenario.

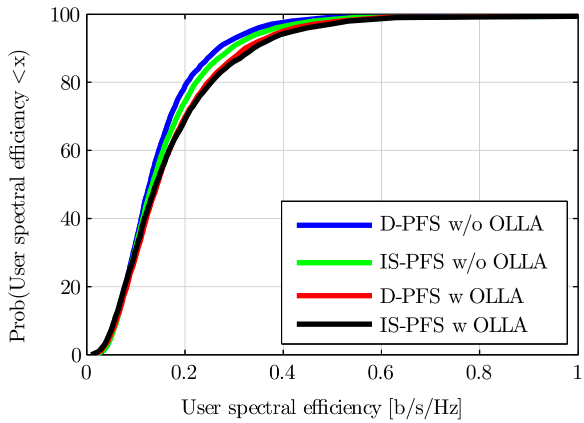


Figure 5.10: User spectral efficiency CDF in the UMa scenario.

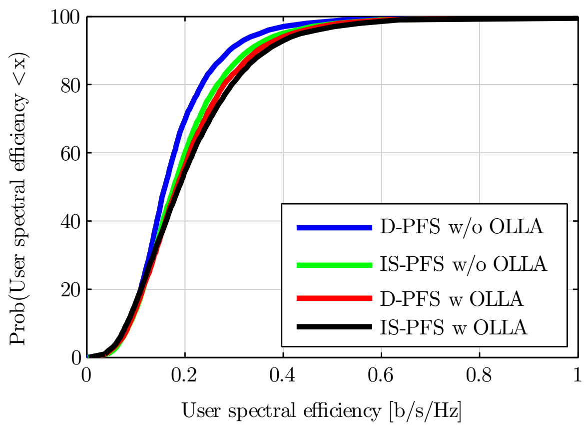


Figure 5.11: User spectral efficiency CDF in the RMa scenario.

5.6 Scheduling in MU-MIMO

In previous sections, only SIMO and SU-MIMO transmission have been considered. Therefore, MU-MIMO scheduling is the next evolutionary step of the algorithms described so far. The use of MU-MIMO provides the opportunity to increase system capacity and user throughputs, but presents new challenges such as increased complexity in the scheduling process. Concerning this process, Mondal [102] identifies the following important decisions:

- selection between SU-MIMO and MU-MIMO transmission,
- determination of the best MU-MIMO user pair in terms of utility maximization,
- selection of transmit weights, and
- MCS determination for MU-MIMO transmission (if selected).

This section provides a complete scheduling solution to make these decisions. This solution is not intended to be optimal, but aims at leveraging the previously developed algorithms to provide a simple solution that takes into account the LTE system constraints. Therefore, the D-PFS algorithm is taken as a basis to develop the multi-user scheduling.

One of the peculiarities of the proposed algorithm is that users are constrained to use 1 layer in their transmissions. This limitation simplifies the algorithm and was used by several organizations involved in IMT-Advanced evaluation process (see e.g. the assumptions of organizations 1, 4 and 5 in [103]).

In a first phase, the D-PFS is applied as usual, with its TDS and FDS stages. This phase entails a first round of RB allocations to scheduling candidates following any criterion. The scheduling decides also in this phase if the users with allocated resources are going to perform a new transmission or a retransmission. Besides, we consider that users with pending retransmissions are prioritized. In each RB, the user selected for scheduling is known as the primary user.

In a second round, each RB can be assigned to an additional user. For each RB, a set of candidate users is formed with all the users capable of doing new transmissions. In case of having a primary user with a retransmission assigned, user selection is based on an exhaustive search of the best combination of users according to any criterion. The selection of users has one restriction: the primary user can not be changed when this user is going to perform a retransmission. With this restriction, we ensure that the users with retransmissions have enough resources to perform their retransmissions. In order to select the users to be multiplexed in each RB, the exhaustive search is done over the set of candidate users with capability to perform new transmissions (in our algorithms they are the users without pending retransmissions). In the exhaustive search, a multi-user utility is calculated for each combination of users being this utility the aggregate of the utilities of the multiplexed users.

Multi-user transmission is decided if the utility of any multi-user combination is higher than the single-user utility. The combination of users with the highest multi-user utility determines the set of multiplexed users.

It is worth noting that the utility of a user in multi-user transmission is different to that of a user in single-user transmission. In order to explain this difference two facts must be considered. First, sharing implies a reduction of the amount of power available for each user as compared with single-user transmission. Second, some interference is produced between the multiplexed users in multi-user transmission. As a result, the Signal to Interference plus Noise Ratio (SINR) and hence the data rate supportable by each user in multi-user transmission is lower or equal to that supported in single user transmission.

In order to calculate the multi-user utility according to proportional fairness criterion, there is a need to calculate the rate that each multiplexed user can achieve. To do so, first the precoder to be used for each user must be calculated. For this, there are different alternatives, as discussed in Section 4.4.

Once selected a set of users with their corresponding precoders, SINR values can be calculated for each user as in [104]. These SINR values are used to evaluate the utility of scheduling the set of selected users. A search over the whole set of users (and usually also over any precoder allocation) leads to the selection of the best scheduling user set.

5.7 Conclusions

A novel scheduling algorithm, named CMA-PFS, has been proposed for the downlink of LTE, based on a well-known algorithm, the D-PFS. First, the advantages and disadvantages of D-PFS have been analysed. Then, as a first step towards CMA-PFS, multi-carrier awareness has been incorporated to D-PFS obtaining the DMA-PFS algorithm. Next, the coupling between the D-PFS phases has been considered to obtain the CMA-PFS. Simulations conducted in an indoor scenario have shown that the proposed algorithms provide an increase of proportional fairness to the system. Furthermore, this increase comes with a higher cell-edge user spectral efficiency at the same time that cell spectral efficiency is improved. This behaviour has been shown for both SIMO and SU-MIMO schemes.

In addition to the previous modifications of the typical D-PFS, a change in the proportionally fair utility function has been proposed to obtain a scheduler with proportional fairness awareness and ability to stabilize the interference, called IS-PFS. The impact of the use of IS-PFS instead of the D-PFS over the system performance has been evaluated by simulation. Results indicate that, in more interference-limited scenarios, the use of IS-PFS is a good option to maximize the proportional fairness, while in less interference-limited scenarios the joint use of IS-PFS and OLLA would be the best option.

Finally, the D-PFS scheduler has been extended to MU-MIMO schemes and a detailed implementation has been explained. This implementation is evaluated in subsequent sections and is the reference for the scheduler used in the CoMP assessments.

It is worth noting that the same concepts applied to proportional fairness scheduling could be extended to other kind of schedulers such as maximum throughput schedulers.

Chapter 6

Analysis of a CS/CB CoMP scheme for LTE-Advanced

This chapter introduces Coordinated Multi-Point (CoMP) transmission/reception as the third part of the analysis of Radio Resource Management (RRM) techniques of Long Term Evolution (LTE)/LTE Advanced (LTE-Advanced) performed in this Thesis. Specifically, this chapter focuses on the CoMP transmission in the downlink. After a general description of the CoMP feature and its evolutionary path in the Third Generation Partnership Project (3GPP), the focus is on a specific implementation that was partly presented in the 3GPP by some partners. This implementation falls in a specific category of CoMP known as Coordinated Scheduling and Beamforming (CS/CB). The last objective of the chapter is to present a detailed description of its implementation, while the evaluation results will be discussed in the next two chapters, dealing with the general evaluation of LTE and the Fifth Generation (5G).

The chapter has been divided into the next sections:

- Section 6.1 explains the evolution of the CoMP support in the 3GPP and its current state.
- Section 6.2 presents the specific CS/CB CoMP algorithm used in this Thesis.
- Section 6.3 outlines the main conclusions of this chapter.

6.1 CoMP in the 3GPP

The evolutionary path of CoMP in the 3GPP started in 2008. CoMP was in its origin closely related to the LTE-Advanced study item and the activities conducted in the 3GPP to develop its specifications towards the fulfilment of the International Mobile Telecommunications Advanced (IMT-Advanced) requirements set by the International Telecommunication Union (ITU). The objective of the first development phase was the inclusion of CoMP in the first version of LTE-Advanced, i.e., the Release 10 of the 3GPP specifications, which would be the 3GPP proposal for IMT-Advanced. The summary of the CoMP-related activities conducted in the 3GPP in this first phase is found in the 3GPP technical report 36.814 [24], whose final version was published in March 2010.

This document is very interesting to understand the main concepts behind CoMP. First, it presents a clear terminology. This terminology defines three important sets of points (base stations in downlink):

- the cooperating set, as the points directly or indirectly participating in data transmission to the user,
- the transmission points set, as set of points actively transmitting data to the user, and
- the measurement set, as the set of points about which channel state information related to their link to the user is reported.

According to these definitions, two main CoMP categories are distinguished:

- Joint processing, in which data is available at each point of the cooperating set. If the transmission is conducted from multiple points at a time, it is categorized as joint transmission, while if the transmission is done from one point at a time, it is call dynamic cell selection.
- CS/CB, in which data is only available at one cell, the serving cell. Nevertheless, user scheduling and/or beamforming decisions are coordinated among the points of the cooperating set.

The CoMP section in [24] identified a set of potential changes to the radio-interface specifications to support CoMP. First, feedback and measurement mechanisms from the users should be enhanced to allow reporting from multiple points. In this regard, it was not precluded the exploitation of channel reciprocity inherent to Time Division Duplexing (TDD). In second place, it was envisaged the need for new reference signals depending on the transmission scheme used, and at the same time it was required that reception and demodulation was as in non-CoMP modes. As a consequence, the reference signals designed for non-codebook Single User MIMO (SU-MIMO)/Multi-User MIMO (MU-MIMO) schemes developed in parallel to CoMP took into account the potential needs of CoMP.

Concerning the feedback in support of downlink CoMP, Section 4.1 already described the possibilities commented in [24]. The list was quite extensive since at the moment of publication of this document there was not a decision about the feedback to be included in the specifications. Concerning this, it is quite enlightening the requirement that any additional feedback designed for CoMP should be consistent with the feedback framework for SU-MIMO/MU-MIMO.

Regarding uplink CoMP reception, it was expected to have limited impact on the system specifications. Both, joint reception of the transmitted signal at multiple reception points and/or coordinated scheduling decisions among cells to control interference were envisaged.

Another great contribution of [24] is the evaluation of the performance of different CoMP schemes in the IMT-Advanced scenarios, namely Indoor hotspot (InH), Urban Micro-cell (UMi), Urban Macro-cell (UMa), and Rural Macro-cell (RMa), and a couple of 3GPP specific scenarios. All those scenarios are characterized by an homogeneous deployment of base stations. Multiple 3GPP members contributed to this evaluation, providing the results of their implementations. The major part of the contributions were based on CS/CB CoMP implementations, but there were also joint processing implementations. Unfortunately, most details of the implementations were not provided by the authors of these contributions.

The rest of the story, that is to say, the rest of the work concerning CoMP in the 3GPP has already been mentioned in Chapter 1. However, it is repeated here for the sake of completeness in the following five paragraphs. If the reader is already familiar with this work, those paragraphs can be skipped.

Although CoMP was a feature studied in LTE-Advanced study item, it was not finally included in the Release 10 specifications due to the huge specification work, and it was postponed to Release 11. In fact, in March 2010, a new study item on CoMP Operation for LTE started. But again, it was considered of low priority and work started in fact in December 2010. In comparison with the previous studies, the main change was the consideration of deployment types different to the homogeneous macro networks in which the performance evaluation of LTE-Advanced was specially focused. Deployment types with distributed Remote Radio Heads (RRHs) and heterogeneous networks were considered in this study item. A technical report [12] presents the outcomes of the study item. The work mainly focused on CoMP with high capacity and low latency communication between transmission points. However, a preliminary evaluation of constraints from lower capacity/higher latency communication between transmission points on the performance gain of CoMP was conducted. As a conclusion, the study item recommended a way forward on actual design principle with its related scenarios for the high capacity/low latency inter-point communication.

The study item finalized in September 2011 and, immediately, a work item started to modify the specifications according to the study item findings. The work focused on providing the means to implement CoMP techniques among transmission points of the same evolved Node B (eNodeB) or among points of different eNodeB but assuming an ideal backhaul (high capacity, low latency). With this aim, means to estimate appropriately the channel quality between a user and multiple transmission points were included in the standard. The core part of the downlink specifications (not accounting for performance requirements nor conformance tests) finalized in December 2012, while the rest of the work ended in December 2013.

In parallel to the Release 11 specification work, a Release 12 study item on "CoMP for LTE with Non-Ideal Backhaul" started in June 2013. The justification of this study item was that CoMP in Rel-11 did not address the support of CoMP with multiple eNodeBs with non-ideal backhaul. Due to this limitation, the operators having non-ideal backhaul may not be able to take performance benefit from CoMP operation. The study considered realistic backhaul delays. Moreover, estimation errors, downlink overhead, complexity, feedback overhead, backward compatibility and practical User Equipment (UE) implementations were taken into consideration. CoMP between macro eNodeBs, between macro eNodeBs and small cells, and between small cells was also studied. In December 2013, the work was completed and gathered in a technical report [13].

Following the study item, a work item on enhanced inter-eNodeB signalling support for Multi-Cell Coordination in LTE started in the 3GPP. Based on the identification of the cases in which CoMP could provide performance gains done by the study item, enhancement on network interface and signalling messages was carried out by the work item. The specification should allow implementing both centralized and distributed coordination, focusing primarily on macro-pico heterogeneous networks but also considering macro-macro homogeneous networks. As a result, existing messages in an interface between eNodeBs were extended to allow the exchange of information useful for CoMP such as hypotheses on activity or muting of specific transmission points, and benefit of such transmission/muting. Besides, the measurement of wideband power received from multiple transmission points was allowed. Release 12 version of technical specification [14] includes explicit comments about inter-eNodeB CoMP and the information that can be exchanged between nodes. The Release 12 CoMP-related work item finalized in December 2014.

In parallel to the Release 12 work item, in June 2014 another work item started in the Release 13, called “Enhanced Signalling for Inter-eNodeB CoMP”. The purpose of this work item is quite similar to the Release 12 one: to extend the interface between nodes in order to allow the exchange of useful information. This work item is being finalized at the moment of writing this Thesis.

In summary, the work conducted in the first phase of development was mostly focused on homogeneous deployments and high-speed links between nodes, or even on the assumption of collocated nodes. On the contrary, the work in the last years has mainly focused on the use of CoMP in heterogeneous scenarios, and deployments with high delay links between nodes.

The CoMP scheme used and assessed in this Thesis falls within the first group. Although the work officially done in the 3GPP has not evolved in this field since 2010, we think that it is still interesting to evaluate such kind of schemes. First, because they are very simple schemes, some of them do not even need any special feedback, and its use is just an implementation issue in the base station. Second, because in the definition of the 5G, mobile communication radio access networks with cloud deployments are being actively investigated, and this kind of configuration is usually characterized by the presence of high-speed links between the transmission points.

6.2 Channel covariance based CS/CB CoMP

This section presents the details of a CS/CB CoMP scheme used and evaluated in this Thesis.

The first important characteristic of the proposed scheme is that coordination is limited to the cells of the same site, whatever number of cells are in the site. This fact ensures a rapid communications between cells. In case of a cloud configuration of the radio access network, this coordinated set could be extended to the whole set of transmitters connected through high-speed links. Besides, the measurement set is assumed to be equal to the coordinated set. Concerning the transmission set, only one coordinated point performs data transmission to each UE. More specifically, for each UE, a cell is configured as its serving cell, being this cell the one transmitting data to the user. Therefore, the scheme falls in the category of CS/CB CoMP.

The second important characteristic is the wideband nature of the Channel State Information (CSI) assumed to be known at the coordinated set to make scheduling decisions. The fact of using a wideband CSI makes the scheme robust to short-term variations of the channels and to the delays in the CSI acquisition. The specific assumptions about the CSI knowledge are presented in Section 6.2.1.

The third characteristic of the scheme is that it is based on a selfish iterative approach, in which each cell makes its own scheduling decisions to maximize its benefit. However, after each scheduling execution, the decision is shared with the coordinated points to help them make better decisions. The exact procedure is explained in Section 6.2.2.

6.2.1 CSI knowledge

The CSI topic has been already explored in Section 4.1. In order to achieve the theoretical gains of CoMP, it would be necessary to have a perfect downlink CSI on each subcarrier. In the scheme studied in this Thesis, a specific CSI measurement is specially relevant, which is the wideband transmit channel covariance, also known as spatial correlation in the literature [105]. The definition of this measurement, $\widehat{\mathbf{R}}$, is expressed in the next equation:

$$\widehat{\mathbf{R}} = \frac{1}{|\mathcal{S}|} \sum_{k \in \mathcal{S}} \mathbf{H}_k^H \mathbf{H}_k, \tag{6.1}$$

where $\mathcal{S}$ is a set of subcarriers, corresponding to a subband where the covariance metric is obtained, $\mathbf{H}_k$ is the channel matrix in subcarrier k. This measure is in fact a wideband version of the transmit channel correlation matrix defined in Section 3.3.1. $\widehat{\mathbf{R}}$ is an instantaneous value, obtained for a specific channel measurement. If this value is averaged over a long period of time, it will converge to statistical correlation.

It is well known that in TDD it is possible to use uplink sounding reference signals to obtain uplink CSI in form of transmit channel covariance in the base station. Using channel reciprocity, uplink CSI can be transformed to downlink CSI. In addition, it has been reported the feasibility of estimating the spatial covariance also in Frequency Division Duplexing (FDD) from uplink signals. See for example the work of Chalise [106], or the reports of Ericsson in the 3GPP such as [105]. In fact, Ericsson and other companies assumed this possibility in its evaluation of LTE-Advanced with CoMP. In this Thesis the same assumption is done. Therefore, we assume that the coordinated base stations are able to estimate $\widehat{\mathbf{R}}$ for all the users connected to any of the cells in the coordinated set.

The CSI could also come to the serving base station as feedback sent from the users after downlink measurements. In this case, the whole matrix or main eigencomponents could be feedback. The current specifications of LTE allow only implicit feedback. This fact does not preclude an estimation of the spatial correlation at the base station, but this feedback-and-reconstruction process would imply some error. The study of this kind of CSI acquisition is out of the scope of this Thesis. Instead, in scenarios with many coordinated points, where it could be difficult the direct estimation of all the CSIs by the coordinated points, we assume that a highly accurate representation of the channel covariance can be reported. This will be the assumption made in some 5G-related scenarios, where we have taken the liberty of making assumptions about the features supported in the 5G. It is worth noting that, in this case, we assume a very short delay in the sharing of CSI between coordinated points.

To sum up, we assume that each base station posses enough knowledge about the spatial correlation from itself and its coordinated cells to each UE. This assumptions is valid for intra-eNodeB coordination, and for the coordination among base stations with high-speed links connecting them.

6.2.2 Iterative process

We propose an iterative scheduling process, where in each iteration, each cell performs an autonomous scheduling of its resources based on previous decisions of its coordinated cells. It is based on a set of works presented by several companies in the 3GPP in the 2009-2010 period. See [107–112] for the details about those works.

One of the main characteristics of this scheme is the selfish behaviour of the coordinated points, which make their decisions with the only goal of maximizing the benefit of their scheduling, not taking care of the effects of their decisions over other points. The coordination comes from the sharing of scheduling decisions between coordinated points to make scheduling decisions that take into account the interference produced by the scheduling of the coordinated points.

Two approaches have been considered in the 3GPP to conduct the per-cell scheduling:

- sequential per-cell scheduling, when in each iteration one cell starts the scheduling when the previous one has ended,
- parallel per-cell scheduling, when all the coordinated cells perform the scheduling simultaneously

For example, Motorola [107], LG [108] and CATT [109] used a sequential approach. On the contrary, Qualcomm [110] and Samsung [111] used a parallel approach. Texas Instruments did not specify its selection [112].

The first iteration is special, because no previous decisions from coordinated cells are available. In this case, scheduling is made as in a non-coordinated implementation.

After each scheduling decision, all decided parameters are broadcasted to the coordinated cells that can use them to estimate the interference coming from other cells and to reduce the effect of those interference to its scheduling decision.

The stopping criterion for the iterative process could be the stability of scheduling decisions. That is to say, when decision are equal in one iteration and the next one. This fact would indicate the reach of a convergence point. However, in general, we cannot ensure the convergence of the algorithm and we will fix a maximum number of iterations based on some testing.

This stopping mode raises one concern. If the scheduling decision of a cell is based on the previous decisions of the coordinated cells, but previous decisions are not guaranteed to be equal to the actual decisions, the final decisions may be really bad decisions.

This problem is more important in the parallel implementation. The right part of figure 6.1 shows an example of a parallel implementation of the scheduling process with three coordinated cells and two iterations. In the parallel implementation, the three cells make decisions (coloured boxes) at the same time, and the decision is sent to the other cells as an input for the next iteration (coloured arrows). The problem is that each cell uses the decisions made by the other cells in the previous iteration, which are not necessarily equal to the final decision.

In the sequential implementation, this problem is somewhat alleviated. See in the left part of Figure 6.1 that the final decision of cell #0 is made using non-final decisions of the other cells. However, final decision of cell #1 takes into account the final decision of cell #0, although non-final decision of cell #2 is available. Moreover, cell #2 makes its decision using final decisions of the other cells. Therefore, sequential implementation ensures that only one cell makes its final decision based on non-final decisions from other cells, while the others have a mix of final and non-final inputs.

The other means proposed in this Thesis to improve the scheduling decisions is a final step added to the algorithms used in the 3GPP. Specifically, as a final step, we have included a phase in which each cell uses the final scheduling decisions of other cells as inputs to change only the Modulation and Coding Scheme (MCS) allocated to each user. This way, we ensure that at least MCS selection is based on the final decisions.

It is worth noting that this iterative approach supports any transmission mode used in the scheduling decisions. It is valid both for SU-MIMO and MU-MIMO.

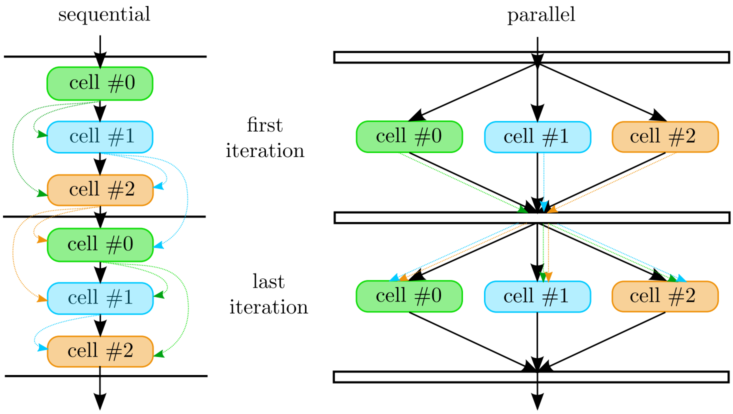


Figure 6.1: Sequential and parallel implementations of the iterative scheduler.

6.3 Conclusions

In this chapter, the evolution of CoMP support in the 3GPP has been first presented. This evolution was firstly focused on homogeneous deployments and high capacity links between coordinated points. In the final phase, it has been more focused on heterogeneous deployments with low capacity links.

This Thesis considers the scenario of high capacity links between coordinated points or collocation of transmission points in the same network nodes (cells of the same site, for example). This scenario is still appealing, especially considering the cloud radio access network concept which is a trend of 5G. Indeed, the fronthaul fibre-based scenario contemplated in several projects like METIS-II is perfectly suited to the assumptions in this chapter.

The CoMP scheme presented in this chapter was partially proposed by several 3GPP members. This chapter describes a particular embodiment of this scheme that considers a selfish behaviour of each coordinated point that makes scheduling decisions based on the knowledge of the decisions of the other coordinated points to improve its own decisions.

Chapter 7

LTE/LTE-Advanced Baseline Performance Evaluation

This chapter provides a performance evaluation of the downlink of the Long Term Evolution (LTE) radio access network. It is worth noting that any system performance evaluation of a system as complex as LTE will be only a partial view of the system possibilities. In fact, this performance evaluation was performed in the framework of the evaluation of the International Mobile Telecommunications Advanced (IMT-Advanced) candidate technologies, and uses the system configurations defined in this process. Therefore, the conclusions of the assessment will be strictly valid for such framework. Nevertheless, some general conclusions can be extracted from this specific set-up. The system configuration includes non-codebook based spatial multiplexing and Coordinated Multi-Point (CoMP) to enhance the basic LTE features, as these features were included in the IMT-Advanced evaluation process. Therefore, the evaluation conducted in this chapter is in fact an LTE/LTE Advanced (LTE-Advanced) evaluation, since the system features assumed are beyond the possibilities of LTE.

With this aim, the chapter has been divided into the next sections:

- Section 7.1 presents the LTE system configuration assumed in the assessment, including the main system parameters and the multiple implementations of key Radio Resource Management (RRM) algorithms described in previous chapters.

- Section 7.2 assesses the performance of an LTE system with multiple users and base stations, for the configurations presented in the previous section.

- Section 7.3 outlines the main conclusions of this chapter.

7.1 System configuration

A common set of scenarios and system configurations have been chosen in all the performed simulations. Their main parameters are summarized in Table 7.1. Note that the traffic model used in this section is the full-buffer model. Therefore, the system will be fully loaded. Conclusions of this chapter would be different if other load conditions were tested. Nevertheless, this testing is out of the scope of the IMT-Advanced evaluations and, therefore, is out of the scope of the evaluation done in this chapter.

The number of users per cell is an important parameter that has an effect on the performance indicators evaluated. A higher number of users would increase multi-user diversity and hence the cell spectral efficiency, but the cell-edge user spectral efficiency would be reduced. We use these values because they were the values most commonly used in the evaluation of IMT-Advanced technology candidates in the International Telecommunication Union (ITU).

It is worth noting that the control channel overhead is fixed, and 3 Orthogonal Frequency Division Multiplexing (OFDM) symbols per subframe are assumed to be devoted to control data transmission.

The schedulers are based on those presented in previous chapter. All of them are proportional fairness aware schedulers. In case of Single User MIMO (SU-MIMO) transmission, the Interference Stabilization Proportional Fair Scheduler (IS-PFS) scheduler, presented in Section 5.5, has been used due to its ability to improve system performance.

Concerning link adaptation algorithms, the IMT-Advanced evaluators did not report the use of Outer-Loop Link Adaptation (OLLA). Therefore, it has not been used in this section. Instead, the IS-PFS scheduler is used as a means to reduce link adaptation failures.

As stated in chapters 4 and 5, in this Thesis it is assumed that retransmissions of a transport block are non-adaptive and use the same number of Resource Blocks (RBs) that the first transmission of this transport block.

The User Equipment (UE) receiver implements a linear Minimum Mean Square Error (MMSE) receiver with interference suppression capabilities as presented in the receiver modelling of Section 3.3.1.

Two antenna configurations are considered in this assessment. In both configurations it is assumed that the UE has a uniform linear array of vertical polarized antennas with 0.5λ spacing. The difference between the configurations is found in the evolved Node B (eNodeB) antenna array. In the Correlated Co-polarized (CC) configuration, the eNodeB has uniform linear array of vertical polarized antennas with 0.5λ spacing, while in the Uncorrelated Co-polarized (UC) configuration the spacing is of 4λ. This information is summarized in Table 7.2. In both configurations we will fix the number of antennas in the eNodeB to 4, and the number of UE antennas to 2, being this a typical configuration.

The electrical tilting of eNodeB antennas has been fixed to the set of values used by most of the IMT-Advanced evaluators.

Table 7.1: System simulation assumptions

Parameter	**Value**
Traffic model	Full-buffer
User density	10 users per cell
Bandwidth	10 MHz in Indoor hotspot (InH), 20 MHz in the rest of scenarios
Control channels overhead	L=3 (OFDM symbols per subframe)
Handover margin	1 dB
Network synchronization	Synchronized
Scheduler	Proportional fair in frequency and time
Receiver type	MMSE with interference suppression
HARQ scheme	Incremental redundancy, asynchronous, non-adaptive
eNodeB antenna tilt. electrical tilt	0° in InH, 12° in UMi and UMa, 6° in RMa
UE antenna config.	Vertically polarized, 0.5 wavelength spacing

Table 7.2: Antenna configurations at the base station

Configuration	**Characteristics**
UC (uncorrelated co-polarized)	vertically polarized 4 wavelength spacing
CC (correlated co-polarized)	vertically polarized 0.5 wavelength spacing

Table 7.3 presents the requirements set by ITU for IMT-Advanced technologies concerning the two performance indicators used in this evaluation: the cell spectral efficiency and the cell-edge user spectral efficiency. In the next

sections, the performance of LTE/LTE-Advanced will be compared with this table.

Table 7.3: IMT-Advanced spectral efficiency requirements.

Metric	**Scenario**			
	InH	**UMi**	**UMa**	**RMa**
Cell spectral efficiency (b/s/Hz)	3	2.6	2.2	1.1
Cell-edge user spectral efficiency (b/s/Hz)	0.1	0.075	0.06	0.04

7.2 System level evaluation

In this section it is evaluated the performance of a complete LTE radio network. Therefore, it is assumed the existence of a cellular deployment with multiple eNodeBs and UEs.

7.2.1 Geometry factors

In order to compare the different deployment scenarios, in an initial phase it is used an indicator known as the geometry factor. This is a wideband Signal to Interference plus Noise Ratio (SINR) obtained considering only channel losses due to large scale variations, that is to say, neglecting multipath fading. This indicator shows a strong dependence on the scenario geometry, hence its name.

Figure 7.1 presents the geometry factor distribution for the different deployment scenarios assuming a transmission with a unique transmit antenna and two receive antennas. The InH scenario presents the higher geometry factor values, mainly between 0 and 50 dB. The rest of scenarios present similar curves with geometry values mainly between 0 and 20 dB.

It is worth noting that in this Thesis modulation errors have not been taken into account. The consideration of such errors would limit the SINR available in any scenarios, typically under $20 - 30$ dB.

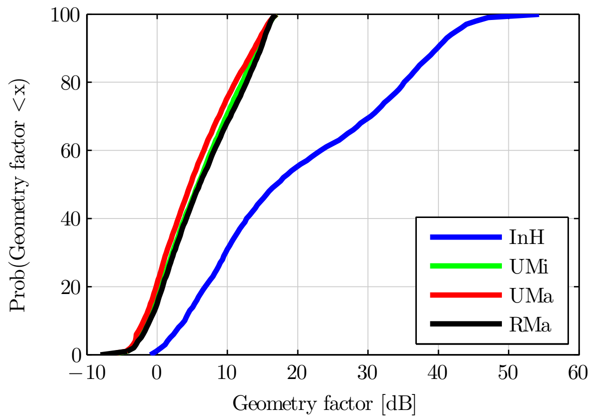


Figure 7.1: Geometry factor CDFs for the different deployment scenarios with SIMO 1×2.

7.2.2 LTE Release 8 evaluation

In this section only the closed-loop spatial multiplexing mode of LTE Release 8 is evaluated. The ability of this mode to adapt the transmission rank makes it a good option in many scenarios. It provides diversity through the use of rank 1 transmissions for users with bad channel quality, and spatial multiplexing through the use of higher ranks for users with good channel. Therefore, the use of transmit diversity would not provide significant gains for the users with worst channel quality, at least in the kind of semi-static simulations conducted in this Thesis. In addition, the use of an open-loop spatial multiplexing, which would be useful for high speeds, would not provide significant gains for the range of speeds considered in this assessment.

With regard to the specific simulation assumptions in this scenario, they have been summarized in Table 7.4.

It is worth noting that non-ideal Channel State Information (CSI) knowledge based on the feedback of implicit indicators is assumed. Subband Channel Quality Indicator (CQI) with a frequency granularity of 5 RBs per subband, and 5 ms time periodicity is assumed. Realistic delays are taken into account in the reporting process.

Table 7.4: Release 8 system simulation assumptions

Parameter	**Value**
Transmission mode	Closed-loop spatial multiplexing
Scheduler	IS-PFS algorithm
CSI feedback	Non-ideal CSI measurement Wideband PMI, RI, subband CQI SINR measurement error per RB is N(0,1)dB 6 ms reporting delay, 5 ms reporting period

SU-MIMO

In this configuration closed-loop spatial multiplexing with dynamic rank selection based on CSI reports is the multi-antenna scheme used.

Figures 7.2 and 7.3 present the cell spectral efficiency and the cell-edge user spectral efficiency, respectively, for the different deployment scenarios. For the sake of clarity, and to facilitate the comparison of the results with the table of requirements, these results have been replicated in table 7.5. In fact, the comparison shows that LTE Relese 8 SU-MIMO mode fulfils the requirements in InH and RMa scenarios, but is far from fulfilling them in UMi and UMa. Performance in RMa is quite similar to that of UMa, but the less demanding requirements in the RMa scenario explain the requirements fulfilment in RMa and not in UMa.

The highest spectral efficiency values are found in the InH scenario. This is due to the better geometry factor distribution of this scenario. Minor differences are observed among the rest of scenarios due to its geometry factor distribution similarity.

Concerning the antenna configurations, the CC configuration provides the best results in all scenarios. Figure 7.4 presents the Cumulative Distribution Functions (CDFs) of the data SINR and the user spectral efficiency. The data SINR is the value in dB of the linear average of the post-processing SINR value calculated for each transport block received by a user. The post-processing SINR is equal to the effective SINR explained in section 3.3.2. It can be observed in the figures that the highest throughputs are due to higher SINR values in all scenarios, except in the InH scenario where it is the result of better spatial isolation of the transmissions. Hereinafter, only CC configuration will be considered in the evaluation. With the current results, the use of CC configuration should be preferred when using similar deployments to those used in this chapter.

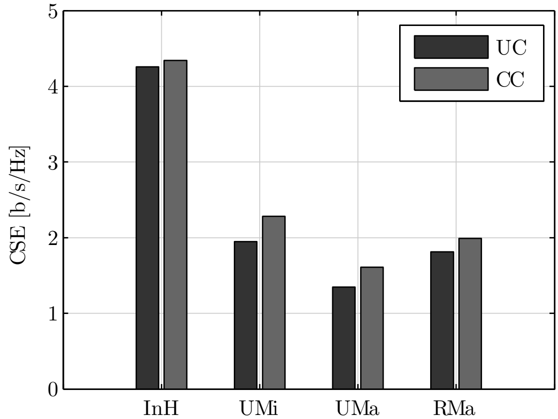


Figure 7.2: LTE Release 8 cell spectral efficiency for the different deployment scenarios.

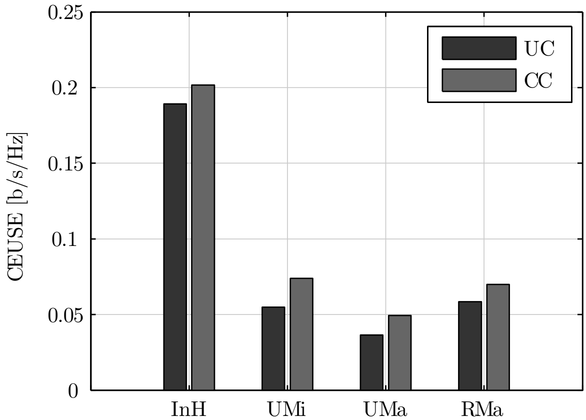


Figure 7.3: LTE Release 8 cell-edge user spectral efficiency for the different deployment scenarios.

Table 7.5: Release 8 spectral efficiencies.

Metric	Scenario			
	InH UC/CC	**UMi UC/CC**	**UMa UC/CC**	**RMa UC/CC**
Cell spectral efficiency (b/s/Hz)	4.26/4.34	1.95/2.28	1.35/1.61	1.81/1.99
Cell-edge user spectral efficiency (b/s/Hz)	0.189/0.202	0.055/0.074	0.037/0.049	0.058/0.070

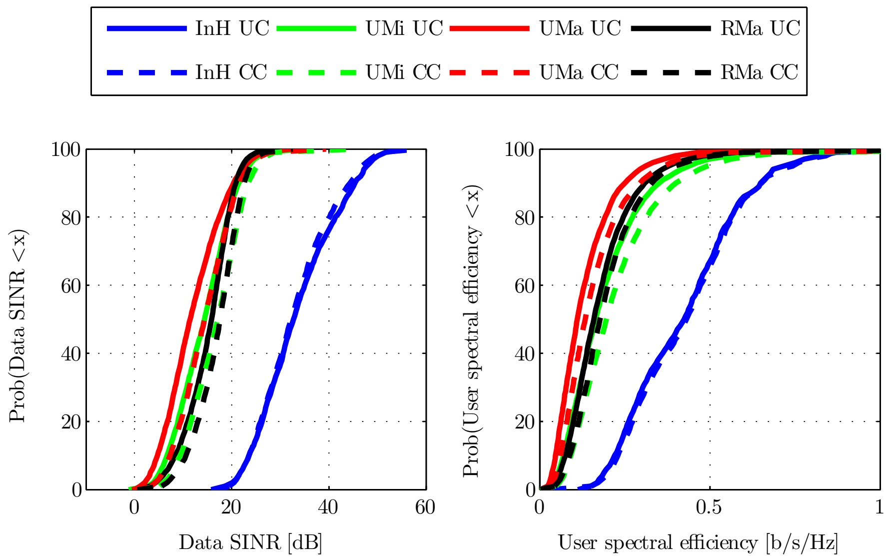


Figure 7.4: CDFs of the data SINR (left) and user spectral efficiency (right) for LTE Release 8 SU-MIMO.

7.2.3 LTE Release 10 without CoMP

LTE Release 10 presents an enhanced multi-antenna mode that allows dynamic switching between SU-MIMO and Multi-User MIMO (MU-MIMO) transmission, non-codebook-based precoding, and the use of up to 8 layers per user.

This section considers the non-codebook based precoding and dynamic switching, but the number of antennas is limited to 4.

The specific simulation assumptions in this scenario have been summarized in Table 7.6. It is worth noting that in this section the multi-user capable extension of the Decoupled PFS (D-PFS) scheduling algorithm presented in Section 5.6 has been used for MU-MIMO simulations. In case of SU-MIMO, the IS-PFS is the used option.

CSI estimation, in form of wideband transmit channel covariance matrix, is assumed to be performed by the eNodeBs in uplink using sounding signals. Implicit feedback, in form of wideband or subband feedback, could be also used. This study does not consider this option following the trend of IMT-Advanced evaluators.

The following evaluations omit the RMa scenario as its behaviour is quite similar to the UMa, and it fulfils all the IMT-Advanced requirements.

Table 7.6: Release 10 system simulation assumptions

Parameter	**Value**
Scheduler	Proportional fair scheduler in frequency and time domains
eNodeB antenna config.	CC configuration
Transmission scheme	Non-codebook based spatial multiplexing
Link adaptation	Dynamic rank selection based on CSI
CSI estimation	Wideband transmit channel covariance matrix estimated from uplink sounding

SU-MIMO

Figure 7.5 depicts the cell spectral efficiency (left) and the cell-edge user spectral efficiency (right) using Release 8 SU-MIMO codebook based (CB) transmission and Release 10 SU-MIMO non-codebook based (NCB) transmission. If the new results are compared to the requirements, it turns out that UMa barely fulfils them, while UMi is still under requirements. In fact, a slight degradation of cell-edge user spectral efficiency is obtained in the UMi scenario compared to the Release 8 results.

Figure 7.6 presents the CDFs of the data SINR and the user spectral efficiency. In general, better efficiencies come together with a degradation of the SINR. Nevertheless, the higher ability to multiplex more streams per user in non-codebook based transmission explains this fact. That is to say, more higher rank transmissions are performed and this means that each transmission uses less power. Additionally, each transmission receives, in average, more intra-user interference due to the use of more parallel streams.

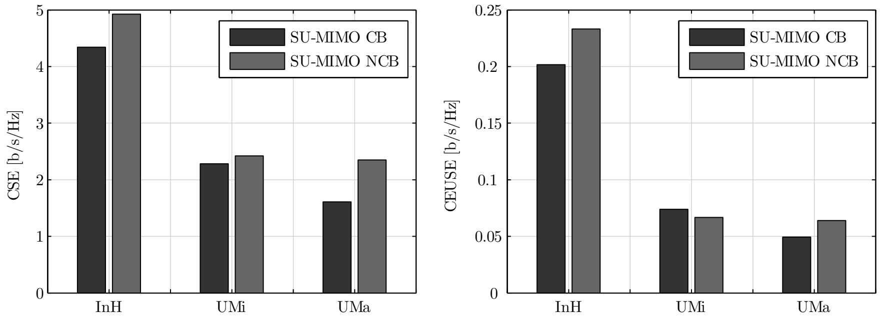


Figure 7.5: Cell spectral efficiency (left) and cell-edge user spectral efficiency (right) using Release 8 SU-MIMO codebook based (CB) transmission and Release 10 SU-MIMO non-codebook based (NCB) transmission.

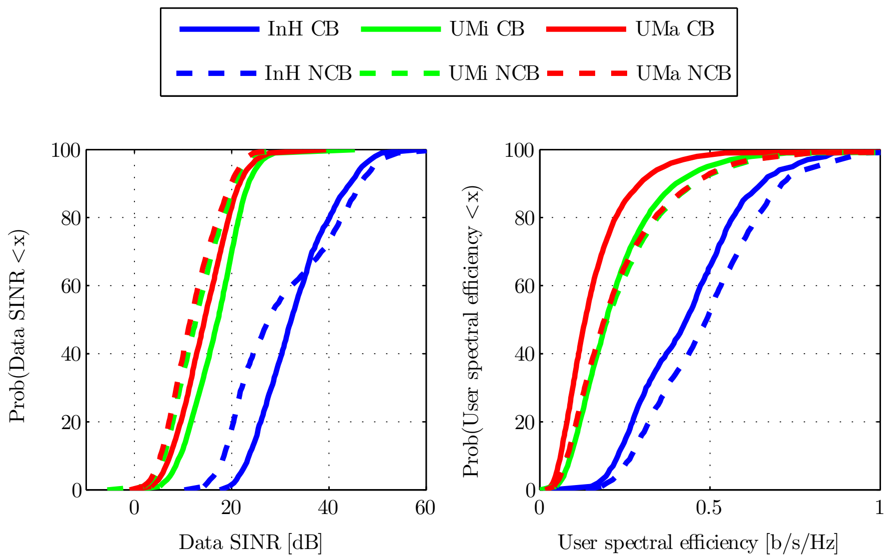


Figure 7.6: CDFs of the data SINR (left) and user spectral efficiency (right) for Release 8 SU-MIMO codebook based (CB) tranmission and for Release 10 SU-MIMO non-codebook based (NCB) transmission.

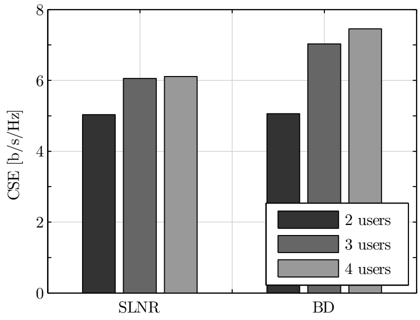
MU-MIMO

Performance of LTE Release 10 with non-codebook based MU-MIMO is assessed in this section. In this evaluation, we have limited the number of layers that each use is able to use to 1. Besides, remember that our CSI feedback assumption is the knowledge of the wideband transmit channel covariance matrix. This means that the layer 1 transmissions are in fact the result of a wideband beamforming.

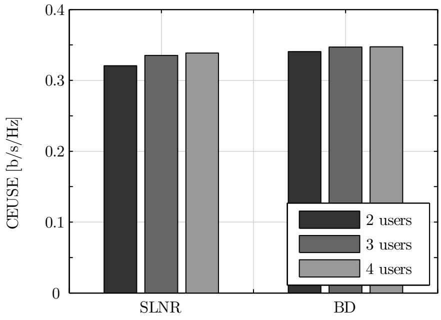
Results in Figure 7.7 show that, with this configuration, it is possible to obtain both cell spectral efficiencies and cell-edge user spectral efficiencies higher than those obtained with SU-MIMO. In this figure, two different precoding schemes are compared, namely Signal to Leakage plus Noise Ratio (SLNR) precoding and a Block Diagonalization (BD)-based precoding. Both were explained in Section 4.4. Moreover, three different maximum numbers of spatially multiplexed users are compared. This numbers are 2, 3 and 4 users. Concerning the precoding, results show that the BD precoding provides the best results in all tests. Therefore, it is more interesting to null the interference produced to the multiplexed users as BD does, instead of maximizing a SLNR metric. With regards to the maximum number of multiplexed users, the gain is clear when going from 2 to 3 users, but increasing this value to 4 is only useful, and not much useful, in the InH scenario. Therefore, limiting the number of users to 3 is, in the evaluated scenarios, a good option to reduce computational complexity experiencing a very low performance degradation (in InH Cell Spectral Efficiency (CSE)) or not any degradation.

Figure 7.8 collects the cell spectral efficiency (left) and the cell-edge user spectral efficiency (right) for Release 8 SU-MIMO ('SU CB'), Release 10 non-codebook based SU-MIMO ('SU NCB') and for Release 10 non-codebook based MU-MIMO ('MU NCB'). For Release 10, only BD precoding is represented, and for MU-MIMO the maximum number of multiplexed users is 4. If the results are compared with the requirements, UMi is now clearly fulfilling them, for both CSE and Cell-Edge User Spectral Efficiency (CEUSE). In general, all the scenarios experience a performance improvement when using MU-MIMO.

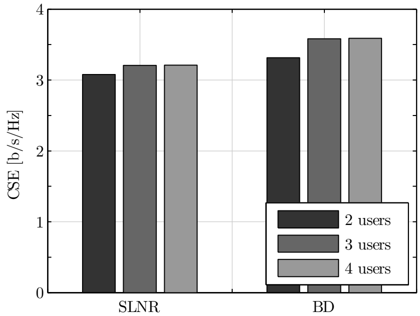
Figure 7.9 presents the CDFs of the data SINR and the user spectral efficiency. The better efficiencies come together with a degradation of the SINR. The multi-user mode enables the multiplexing of more streams per subframe than in single-user mode. The reason is found in the easy spatial separation of streams when each stream is sent to users that are spatially distant from each other. Although each stream experiences a poorer SINR, the increment in the number of simultaneously transmitted streams increases the system efficiencies.



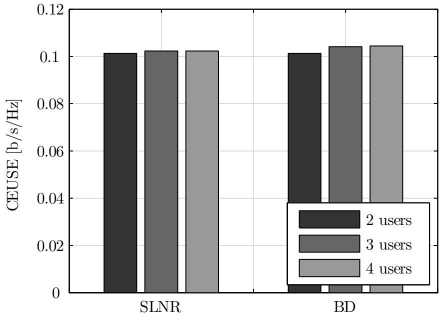
(a) CSE in InH.



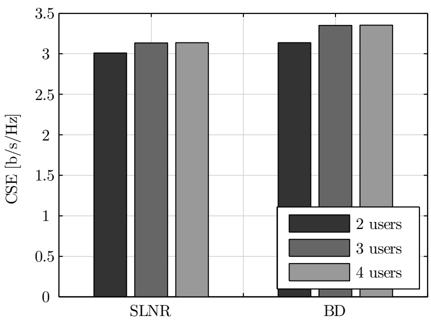
(b) CEUSE in InH.



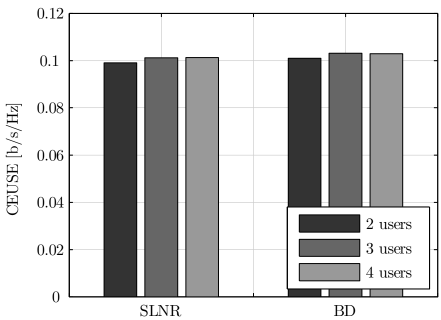
(c) CSE in UMi.



(d) CEUSE in UMi.



(e) CSE in UMa.



(f) CEUSE in UMa.

Figure 7.7: Cell spectral efficiency and cell-edge user spectral efficiency in different scenarios using non-codebook based MU-MIMO transmision with BD and SLNR precoding, and a different maximum number of multiplexed users (2, 3 or 4).

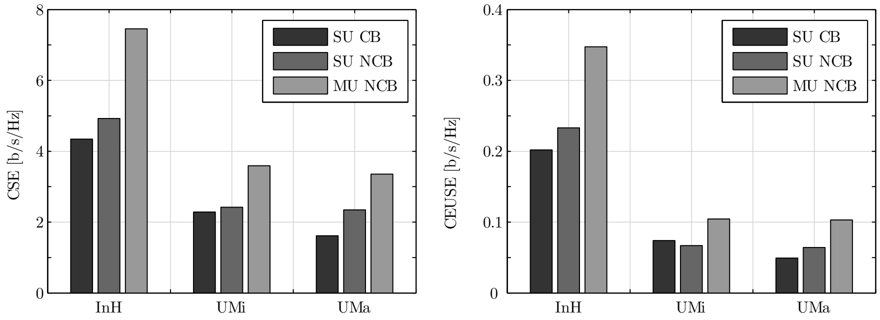


Figure 7.8: Cell spectral efficiency and cell-edge user spectral efficiency for different scenarios using Release 10 non-codebook based transmission in SU-MIMO and MU-MIMO with up to 4 multiplexed users and BD precoding.

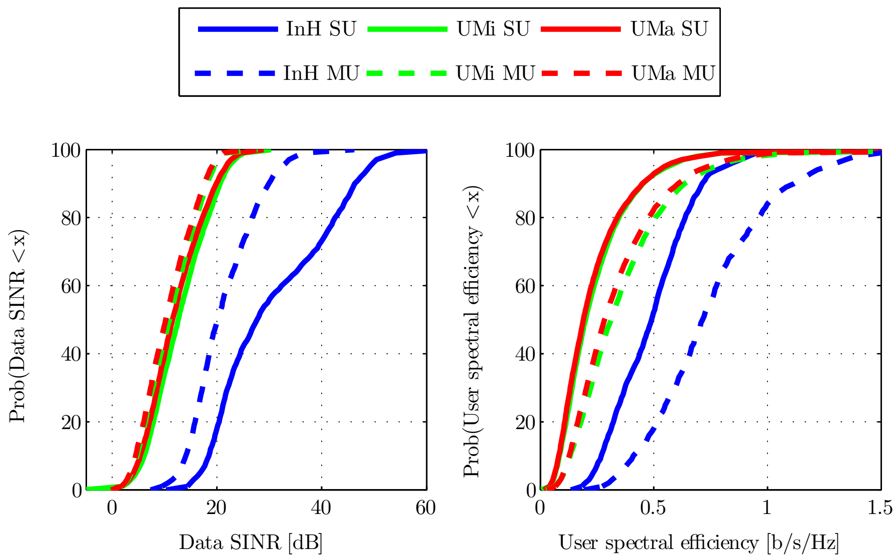


Figure 7.9: CDFs of the data SINR (left) and user spectral efficiency (right) for different scenarios using Release 10 non-codebook based transmission in SU-MIMO and MU-MIMO with up to 4 multiplexed users and BD precoding.

7.2.4 LTE Release 10 with CoMP

LTE Release 10 implicitly supports the Coordinated Scheduling and Beamforming (CS/CB) CoMP scheme presented in Chapter 6 for intra-eNodeB CoMP. In this section, the effect of the use of this specific CoMP implementation on the system performance is assessed.

It is worth noting that the CSI knowledge assumed in the previous section is also assumed in this one, that is to say, the knowledge of wideband transmit channel covariance matrices at the transmitter side.

Figure 7.10 depicts the cell spectral efficiency (left) and the cell-edge user spectral efficiency (right) of Release 10 non-codebook based MU-MIMO ('MU NCB') and the same scheme with CoMP ('MU NCB + CoMP'). Only BD precoding has been used, and the maximum number of multiplexed users is 4. As can be observed in the figures, the performance gain is quite limited.

In fact, in Figure 7.11, which presents the CDFs of the data SINR and the user spectral efficiency, a very small improvement of the data SINR is observed. This increment is due to the best precoder selection in CoMP based on the sharing of scheduling decision between coordinated transmission points.

Figure 7.12 summarizes the gains, as compared with the LTE Release 8 configuration, seen in the cell spectral efficiency and cell-edge user spectral efficiency when the LTE Release 10 features are considered. It can be concluded that the most important step forward is the consideration of non-codebook based MU-MIMO ('MU NCB'). Therefore, the implementation of this technique should be prioritized with the aim of maximizing the system performance when deployments are similar to those used in this assessment.

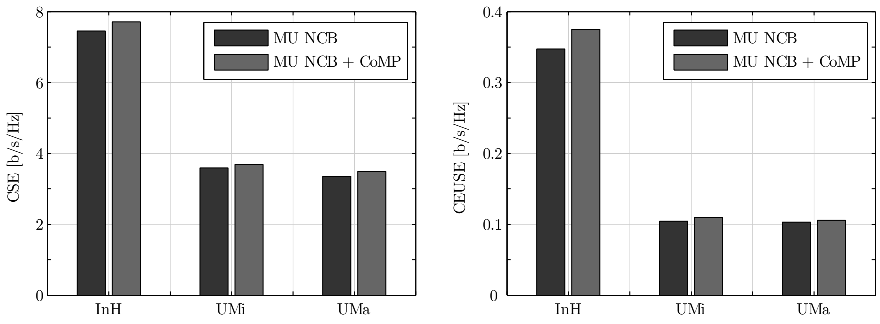


Figure 7.10: Cell spectral efficiency and cell-edge user spectral efficiency for different scenarios using Release 10 non-codebook based MU-MIMO transmission with and without CoMP.

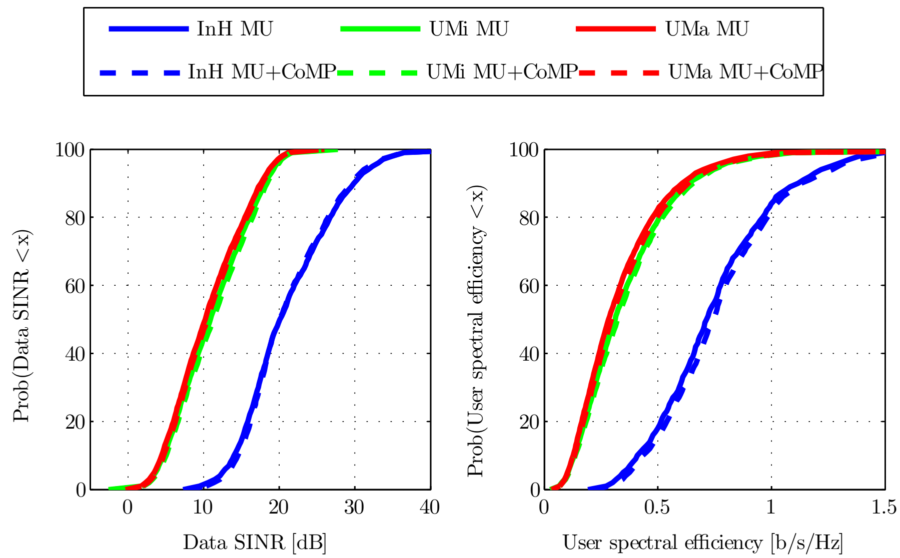
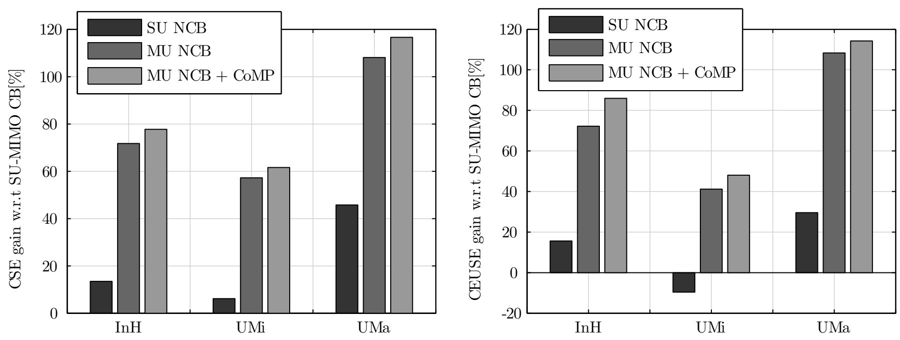


Figure 7.11: CDFs of the data SINR (left) and user spectral efficiency (right) for different scenarios using Release 10 non-codebook based MU-MIMO transmission with and without CoMP.



(a) CSE gain. (b) CEUSE gain.

Figure 7.12: Gains in cell spectral efficiency (left) and cell-edge user spectral efficiency (right) for the Release 10 configurations with regard to Release 8 SU-MIMO.

Figure 7.13 depicts the gains produced by the inclusion of CoMP in comparison to LTE Release 10 MU-MIMO in cell spectral efficiency (left) and cell-edge user spectral efficiency (right). Although the gains are consistent, which is very positive, they are very limited. We envisage the need for a stronger coordination than that assumed in the proposed scheme to obtain higher gains. However, the gains obtained in the cell-edge user spectral efficiency for the InH scenario suggest that this technique is useful in some scenarios, with likely a reduced amount of predominant interferers, being easy in these scenarios to coordinate a small amount of transmission points to obtain non-negligible gains. This case could be, for example, found in some heterogeneous networks were interference may come to some users from a reduced set of high-powered interferers. The work of Lee and others in [113] reinforces these ideas.

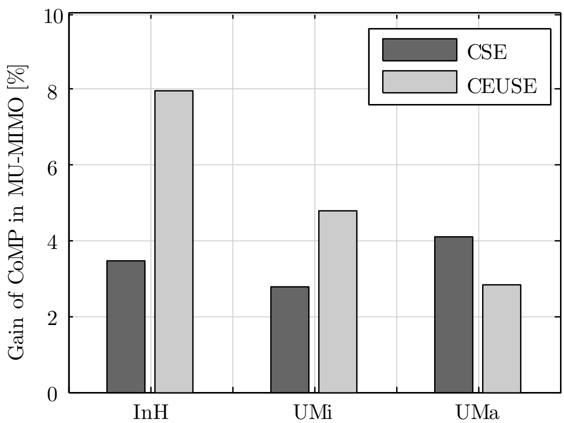


Figure 7.13: Gains in cell spectral efficiency (left) and cell-edge user spectral efficiency (right) for the MU-MIMO CoMP with regard to MU-MIMO without CoMP.

7.3 Conclusions

This chapter has provided a performance evaluation in the downlink of the LTE radio access network performed in the framework of the evaluation of the IMT-Advanced candidate technologies.

In this framework, the evaluation focused on the impact that several multi-antenna techniques have on the system performance, and its utility to reach the goals set by the ITU Radiocommunication Sector (ITU-R) for the IMT-Advanced technologies.

It is remarkable that link adaptation in this assessment does not include the OLLA mechanism. Therefore, the IS-PFS scheduler designed to stabilize the resource allocations has been used to achieve a good link adaptation. Without this algorithm, the system performance would be worse and the requirements of IMT-Advanced would be hardly fulfilled.

The main conclusion of the chapter is that the multi-antenna technique with the highest impact on the system performance, when the indicators are the cell spectral efficiency and the cell-edge user spectral efficiency, is the non-codebook based MU-MIMO of LTE-Advanced. Additionally, we have concluded that a block diagonalization based precoding provides better results than the SLNR precoding, even when the CSI knowledge of the transmitter is limited to a wideband transmit channel covariance matrix. In fact, the utility of this kind of feedback to increase the system performance is a very interesting conclusion of the assessment.

Finally, and concerning the CoMP scheme proposed in a previous chapter, it has proven to be useful mainly in the InH scenario, while its benefit in other scenarios is quite limited. Nevertheless, the usefulness in InH is considered a hint of the ability of this scheme to provide significant results in more heterogeneous scenarios, or in scenarios with a reduced amount of interferers concentrating most of the interference. This idea will be confirmed in the next chapter.

Chapter 8

5G Performance Evaluation

This chapter provides the performance evaluation of the Long Term Evolution (LTE) and LTE Advanced (LTE-Advanced) with Coordinated Multi--Point (CoMP) transmission in a set of scenarios selected by the European project Mobile and wireless communications Enablers for Twenty-twenty Information Society (METIS). Given that LTE-Advanced is extended with additional functionalities not contemplated in the current standard, it can be considered that this is somehow a seminal work on the Fifth Generation (5G) performance evaluation, although the 5G technology is still not defined.

With this aim, the chapter has been divided into five sections:

- Section 8.1 describes the 5G METIS objectives and test cases.
- Section 8.2 explains the 5G METIS simulation methodology.
- Section 8.3 presents a performance evaluation conducted in a virtual reality office scenario.
- Section 8.4 details a performance evaluation conducted in a sports stadium scenario.
- Section 8.5 outlines the main conclusions of this chapter.

8.1 METIS objectives and test cases

The overall purpose of the METIS project was to develop a 5G system concept that fulfilled the requirements of the beyond-2020 connected information society and to extend wireless communication systems for new usage areas.

The METIS 5G concept was developed into a system supporting three generic services envisaged for 2020 and beyond: extreme mobile broadband, massive machine-type communications, and ultra-reliable machine-type communications. These services map to a set of technical METIS goals relative to the legacy systems identified in [114]:

- 1000 times higher mobile data volume per area,
- 10 to 100 times higher typical user data rate,
- 10 to 100 times higher number of connected devices,
- 10 times longer battery life for low power devices,
- 5 times reduced End-to-End (E2E) latency.

However, all these technical objectives do not need to be met at the same time. In [114] twelve test cases were defined to investigate the technical objectives in detail, with each test case focusing on a subset of the requirements and specific goals.

METIS deliverable D6.1 [115] provided simulation guidelines to align assumptions, methodology and simulation reference cases in order to allow for a direct comparison of different technology components. Following the simulation guidelines specified in [115], the METIS technical objectives were investigated in deliverable D6.5 [116] for each of the twelve METIS test cases. In fact, four of these test cases were defined in terms of simulation modelling and assessed by the author of this Thesis:

- test case 1, virtual reality office.
- test case 4, stadium.
- test case 5, teleprotection in smart grid network.
- test case 7, blind spots.

Additionally, preliminary analysis of performance in test case 2, dense urban information society, was also conducted by the author of this Thesis, and included in deliverable D6.3 [117]. In this chapter, results concerning METIS test cases 1 and 4 are detailed with a special focus on the role of CoMP on the final performance of the system. Only these two test cases were selected because of their particular relevance to the proposals of this Thesis. Test case 1 and test case 4 are requiring massive capacity, for which advanced scheduling and CoMP schemes were applied. On the other hand, test case 5 was especially suited for the investigations on new medium access mechanisms and Device-to-Device (D2D) communications, whereas test case 7 was more interested in coverage extension methods.

8.2 METIS simulation methodology

METIS simulation methodology is described in [115]. Note that this Thesis made a significant contribution to this methodology proposal. In this section, two aspects of this methodology are presented: the propagation models and the traffic models.

METIS contemplates three approaches to the modelling of propagation. One is based on legacy simulators where channel is in general based on geometry-stochastic channel models. This kind of models shows some unrealistic effects, such as the fact that line of sight situation is not spatially correlated. Another approach is the use of ray tracing models that, in principle, are much more realistic. Both, large-scale and small-scale propagation effects could be easily considered in this approach. However, most simulators are not ready to do so. In order to leverage the available simulators while approaching to a more realistic modelling, a hybrid approach is used. In this approach, the determination of the line of sight condition is based on pure geometrical considerations and large-scale fading is based on ray tracing. Other large-scale parameters are cross-correlated among them, and with the large-scale fading, according to the parameters of a specific geometry-stochastic model. Finally, small-scale modelling is done using a geometry-stochastic model. This modelling approach was used and validated in a paper co-authored by the author of this Thesis [118].

In [115] a set of propagation scenarios covering a wide range of propagation conditions were defined. The most relevant propagation scenarios for this Thesis are:

- propagation scenario #7, which models the link between a base station and a mobile station in an indoor office. Proposes the use of International Telecommunication Union (ITU) Radiocommunication Sector (ITU-R) Indoor hotspot (InH) model for small-scale modelling.

- propagation scenario #13, which models a D2D link in an indoor office. Same modelling as propagation scenario #7.

- propagation scenario #1, which models the link between a base station and a mobile station in a urban micro-cellular environment when both entities are outdoors. Proposes the use of ITU-R Urban Micro-cell (UMi) model for small-scale modelling.

- propagation scenario #9, which models a D2D link in a urban micro-cellular environment with both link ends outdoors. Same modelling as propagation scenario #1.

Again, it is worth noting that the definition of those propagation scenarios was made by the author of this dissertation and resulted from the investigations made in this Thesis.

Concerning the traffic modelling, and for the sake of simplicity, in the final evaluation of the METIS system, a constant bit rate model with fixed time between packet generations and fixed packet size was generally used. In this chapter this model is also considered.

8.3 Performance evaluation in a virtual reality office

This section includes the results of the performance evaluation in an indoor office as specified by the test case 1 of the METIS project, also referred to as Test Case 1 (TC1). This test case consists of an indoor scenario in which the work involves interaction with high-resolution Three Dimensions (3D) scenes, where a team of 20 individuals is simultaneously interacting using virtual reality. The challenge is to support a huge traffic flow of 5 Gbps per user interacting in the scene, both in uplink and downlink, while keeping packet Round Trip Time (RTT) at the application layer below 10 ms in average.

8.3.1 Description of the test case

This test case consists of a top-modern office space located in a refurbished 19th century building classified as cultural heritage. The building is rented by a company working with 3D tele-presence and virtual reality. The work involves interaction with high resolution 3D scenes and is typically performed in teams of some 5 to 10 individuals simultaneously interacting with a scene. Some team members are sited within the building; others are working remotely from other office buildings. Each scene may include the virtual representation of the team members or computer generated characters and items. The high-resolution quality of the scene provides an as-if-you-were here feeling. Since each team member may affect the scene, all must continuously update the scene by streaming data to the others. In order to provide the real-time interaction, the work is supported by bi-directional streams with very high data-rates and low latencies.

8.3.2 Main KPIs and requirements

Main Key Performance Indicators (KPIs) of TC1 are summarized in the table below. More details on the definition of these KPIs can be found in [114].

Table 8.1: Main KPIs defined for TC1

KPI	Requirement
Traffic volume per subscriber	36 [Tbyte/month/subscriber]
Average user data rate (busy period)	0.5 [Gbps]
Traffic volume per area	100 [Mbps/m2]
Experienced user data rate	1 [Gbps] with 95% availability 5 [Gbps] with 20% availability
Latency	10 [ms] round trip time

8.3.3 Simulation models

Simulations follow the models included in [115]. In the following, a short description of those models and some assumptions used in this Thesis are presented.

Environmental model

A realistic office environmental model for this test case is attained by explicitly considering walls, screens, desks, chairs and people. Figure 8.1 shows a 3D visualization of the reference environmental model defined in [115]. For more details, the reader is referred to the Annex 9.1 of [115].

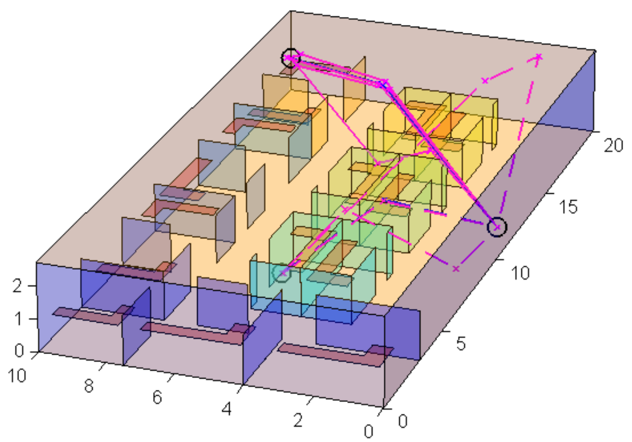


Figure 8.1: 3D visualization of the virtual reality office environmental model reference case. [115]

Deployment considerations

The deployment baseline is one main base station ceiling-mounted with fibre backhaul and with User Equipments (UEs) in either a sitting or standing position. This coverage base station works at frequencies below 6 GHz, in particular at 3.5 GHz, and is located in the centre of the office, whereas other five Access Points (APs) are deployed and operate at the mmW band, in particular at 60 GHz, to give further capacity to the system. This scenario is assumed to be isolated from outside interferences. It is worth noting that, for the performance assessment, the existence of fibre optics connection to all access points is assumed, which does not strictly follow recommendations in [115].

Propagation model

For the coverage small cell, it was used the METIS model described in [115] as propagation scenario #7, where the position of the walls was explicitly considered to determine the line of sight condition. For the propagation at mmW band, the ray-tracing material provided by other METIS partners and available at [119] was employed.

Traffic model

The File Transfer Protocol (FTP)-traffic model described in [115], corresponding with a constant bit rate traffic source, was used, adapting the reading time to achieve the desired data rates.

Mobility model

The users were stationary for the duration of the simulation. Although no mobility of users is assumed, a velocity of 3 km/h is used to account for small-scale effects.

8.3.4 Assumptions

This subsection describes some assumptions, not originally described in [115], defined and used in this Thesis to complete the simulation framework.

As depicted in Figure 8.2, in the performance evaluation of TC1 it is assumed that the main base station and the access points are connected to a local routing entity through fibre optic links. It is also assumed the presence of a Central Image Processing Entity (CIPE) with D2D capabilities connected to the local routing entity through fibre optic. The local routing entity is connected again through a fibre optic link to a security gateway that provides access to the mobile network operator core network.

Data flows in TC1 are shown in Figure 8.3. Every user (users in office and users out of office) sends/receives a traffic flow to/from a CIPE located in the same office. This flow can be sent/received to/from the central processing entity through the local routing entity or through D2D links. The CIPE collects all uplink flows, which include the video of each particular user, and bounces back the combined scene in the downlink.

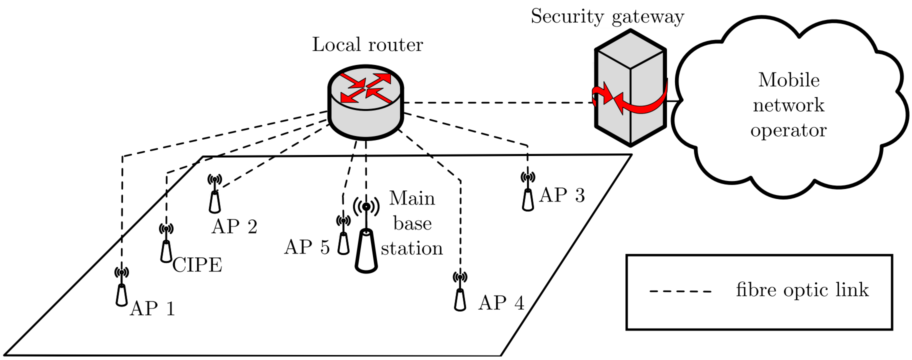


Figure 8.2: Architecture of the radio access network in TC1 evaluations.

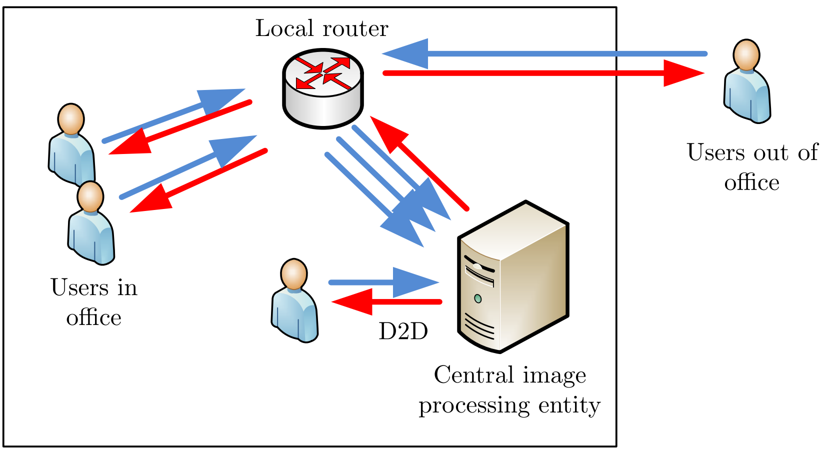


Figure 8.3: Data flows in TC1 and the role of the CIPE.

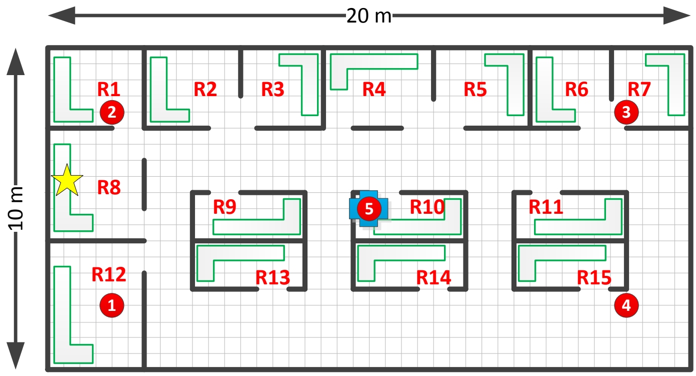


Figure 8.4: Office layout and deployment.

Due to the need for a specific layout in the ray-tracing calculation, a set of offices, corridors and desks has been defined, following the main descriptors in [114]. In particular, the scenario includes fifteen offices, one small cell station (blue cross) and five mmW access points (red circles). The central image processing entity is located in room 8 (yellow star). See Figure 8.4 for more details on the room distribution.

All transmitters and users are equipped with 8 antennas (4 cross-polarized antennas). At least 60 MHz are allocated to the main base station at 3.5 GHz, whereas 140 MHz are given to access points operating at 60 GHz. Simulations will include multiples of this allocated bandwidth (200 MHz) to assess the final requirements in terms of spectrum to support this challenging scenario.

Regarding users' location, according to mean user density and area, 20 users in average should be generated in the office. In our simulations, 18 users were placed in the office rooms and 2 users were located in outer locations. All rooms were occupied by one user except rooms R4 (2 users), R8 (4 users), and R12 (2 users). There were no users explicitly placed in outer positions but 2 users in the office (not those in R8) were in a video call different to that of the other 18 users. They were consuming radio resources and latency was artificially increased to account for the real performance of those actual outer users.

Traffic was modelled with FTP at 1 Gbps in each direction. We assumed that packet size is 20 Mb, and time between packets was 20 ms (50 fps). Latency for a fibre optics link was assumed to be 1 ms. Processing in routers and gateways was also assumed to be equal to 1 ms. Latency between gateways for users in outer locations was assumed to be 1 ms. These values are added to the radio access network delays, which depend on congestion, scheduling and system capacity.

8.3.5 Technology components

This subsection describes a set of technology components, or technology solutions, added to LTE-Advanced to reach the required KPIs in this test case.

The main characteristic of this test case is the huge amount of data, which brings the system to a congested state. In this situation, RTT latency of 10 ms forces the system to be highly efficient in the management of retransmissions, since otherwise, the delay in the radio interface would prevent users from having an appropriate experience. However, interferences are huge, and therefore coordination among access points is mandatory. Given that there is a central entity in the area, centralized coordination will be investigated here. In this sense, evaluations will be progressively investigating the performance of the system with Single User MIMO (SU-MIMO), Multi-User MIMO (MU-MIMO) and CoMP. Among the different proposed CoMP schemes, the evaluations in this chapter focus on the proposal made in this Thesis based on Coordinated Scheduling and Beamforming (CS/CB).

On the other hand, today's Transmission Time Interval (TTI) of 1 ms seems not to be appropriate for this scenario of tight latency requirements. Therefore, the frame structure to reduced TTI was adapted, also paying attention to the specific requirements on the signalling and frame structure when increasing the carrier frequency up to the mmW band. In this sense, it was assumed the use of a proposal in METIS that gives support for robust and fast control plane, fast network synchronization, short physical layer latency (data RTT latency), flexible data direction switching (for wireless backhauling, network controlled small cells), Physical (PHY) layer enablers for interference handling by cross-link interference mitigation capabilities and support of Orthogonal Frequency Division Multiplexing (OFDM)-based Multiple-Input Multiple-Output (MIMO) solutions including high-gain beam-forming [120]. To fully exploit available spectrum, this proposal enables the dynamic Time Division Duplexing (TDD) mode, which allows flexible Uplink (UL) and Downlink (DL) slots depending on the instantaneous traffic conditions. On the other hand, the adapted frame structure enables short latencies and efficient PHY signalling (control messages for UL and DL can be exchanged in every slot regardless of the direction of the user data transmission). Using this METIS proposal, TTI was reduced from 1 ms down until 0.25 ms. Results show the impact on this reduced TTI on the actual performance of the system. Other subsidiary techniques were used, to reduce interferences (through a Decentralized Interference Aware Scheduling) and to facilitate D2D communications (a Distributed Channel State Information (CSI)-based Mode Selection for D2D Communications is assumed). All these techniques are out of the scope of this Thesis, but the reader may refer to [120] for more details.

8.3.6 Results

Figure 8.5 depicts the pathgain in dB for the main station, the five mmW access points (cells 1–5), and the D2D transmitter used by the CIPE (cell 6). The lack of complete walls reaching the ceiling increases the influence range of each access point.

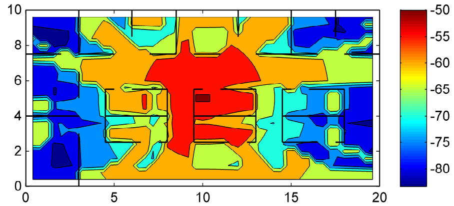
We consider a finite set of possible locations for the users in the scenario. For each location, the serving cell is selected as the cell providing the highest received power to the user. Figure 8.6 shows the possible locations and their serving cell. In this calculation we have excluded the main station.

Consequently with the wide range of each access point, wideband Signal to Interference plus Noise Ratio (SINR) is in general low, mostly in the rooms without an access point. It can be clearly observed in Figure 8.7. As represented in Figure 8.8, median SINR is below 10 dB being cell-edge users below 0 dB. It is worth noting that with a median below 10 dB, the expected bandwidth required to fulfil TC1 requirements is 4.5 GHz according to METIS spectrum analysis [121]. This interference-limited scenario will limit the set of techniques to be employed and force to make use of appropriate tools to reduce the impact of these interferences.

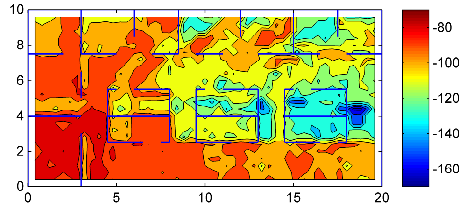
The goal is to experience 1 Gbps in 95% locations (for cell-edge users) and 5 Gbps in 20% locations, which is equivalent to have this value in percentile 80-th of the user-throughput Cumulative Distribution Function (CDF). Moreover, we need to have less than 10 ms of mean packet latency, and less than 5% of packet loss rate. In this scenario, with 200 MHz allocated to the system it is impossible to handle the amount of data to be delivered between the different recording units and the central processing entity. In fact, even with the best configuration, 200 MHz system reach congestion and the quality of service is far from the requirements specified for TC1 in [114]. Therefore, simulations have included a range of simulations, from 200 MHz up to 4.5 GHz, to check the required bandwidth for the different configurations of the system.

Different TTI values have been used in this scenario including legacy assumption of 1 ms, 0.5 ms and 0.25 ms. Moreover, several transmission schemes have been tested: Single Input Multiple Output (SIMO), Codebook Based (CB)-based SU-MIMO, non-CB MU-MIMO and non-CB MU-MIMO with CoMP. For CoMP, all access points are supposed to participate in the same cluster, in such a way that scheduling and resource allocation is made in collaboration with the other transmitters using the simple scheme proposed in previous chapters.

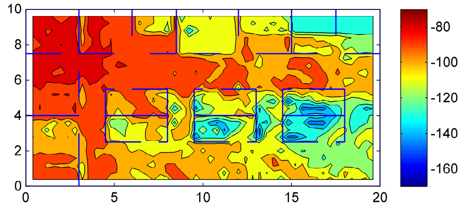
We first analyse the influence of bandwidth in the system performance for SU-MIMO (8x8). As represented in Figure 8.9, with a bandwidth of 4.5 GHz all requirements are met unless maximum packet latency of 10 ms, which is



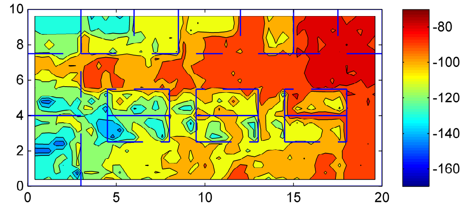
(a) main base station



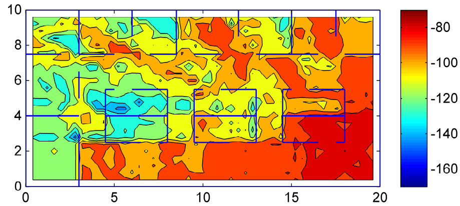
(b) AP 1



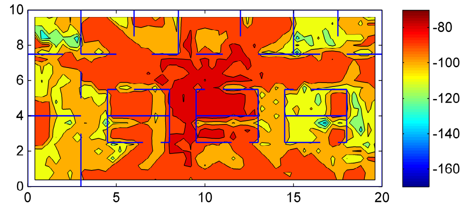
(c) AP 2



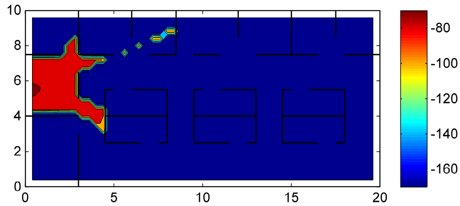
(d) AP 3



(e) AP 4



(f) AP 5



(g) CIPE (D2D station)

Figure 8.5: Path gain for the different transmission points in TC1.

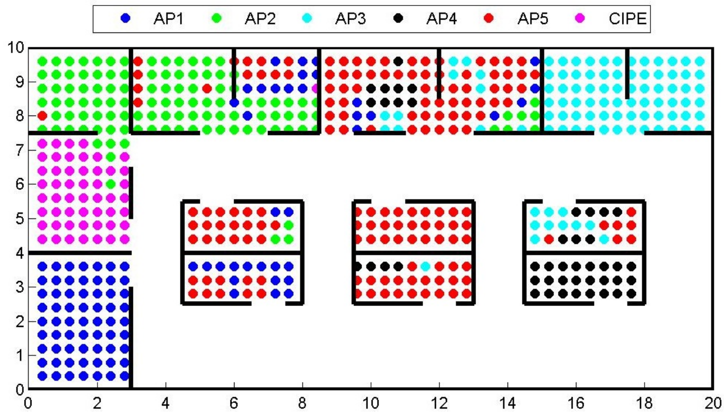


Figure 8.6: Serving transmission point in each location (excluding main base station).

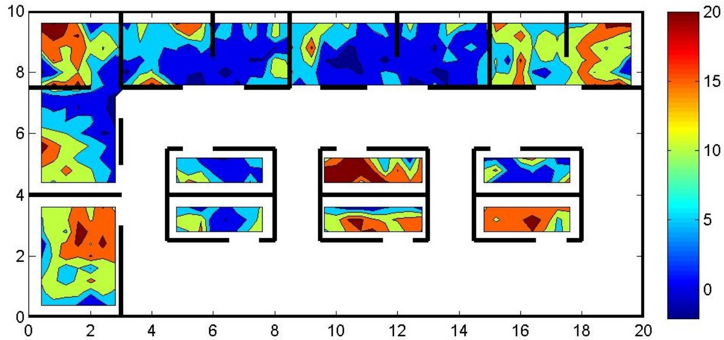


Figure 8.7: Wideband SINR in the scenario under study.

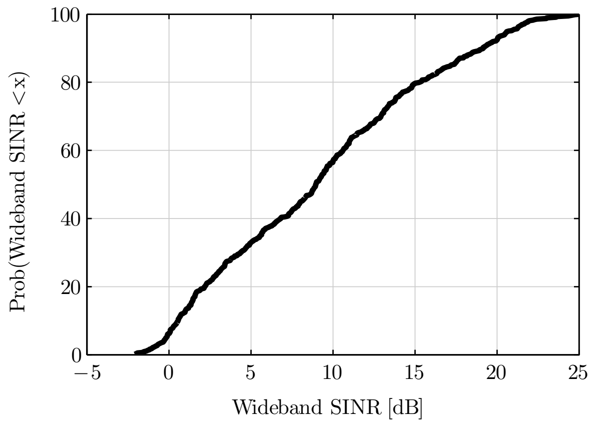


Figure 8.8: CDF of the wideband SINR in the scenario under study.

satisfied in around 85%. However, other techniques can be added to increase efficiency and reduce packet latency. Note that the figure represents to which extend the required KPI is fulfilled. In this sense, 100% means that the target is totally achieved.

We first evaluate, for 1.5 GHz, the effect of increasing complexity of MIMO transmission and coordination. As shown in Figure 8.10, the use of MU-MIMO and coordinated transmission and reception, as compared with SU-MIMO, increase by a factor of almost three cell-edge user performance but degrades (-24%) average performance. When reducing TTI time to half a millisecond, the cell-edge user performance is affected, but we improve the use of spectrum, which finally results in a better average cell performance. This trend is again observed for a TTI value of 0.25 ms. In this case, average throughput improves (+21%) and target is almost fulfilled. However, mean packet round trip time is far from reaching the objective and cannot be improved even reducing the TTI value.

After several trials increasing bandwidth we found out that the fibre optics and processing time is around 6 ms (time per fibre optic link is assumed to be 1 ms, and processing time in routers is also assumed to be 1 ms) and radio time cannot be less than 2.2 ms per direction, that is 4.4 ms in total. Therefore, the only way of reaching the goal is to improve fibre optics or routing time around 10% to have less than 10 ms in RTT time. If this is possible, with 1.875 GHz the system will be able to fulfil all system requirements.

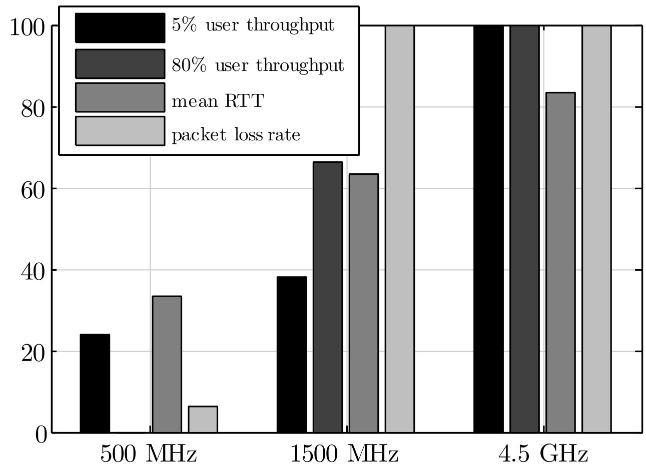


Figure 8.9: Level of satisfaction of KPI for TC1.

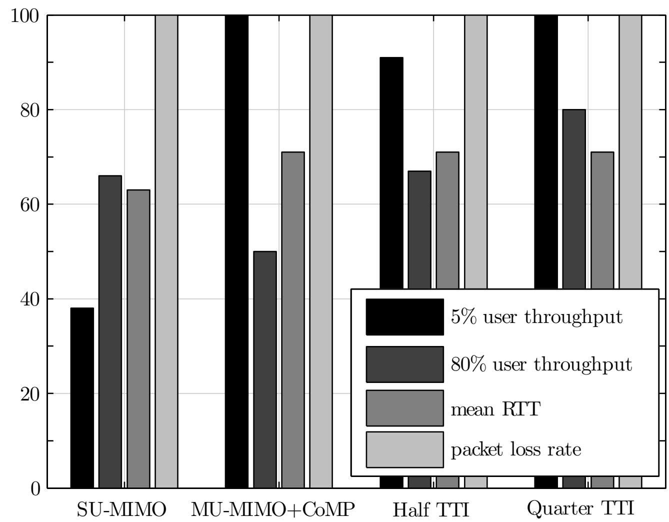


Figure 8.10: Level of satisfaction for different transmission schemes and 1500 MHz bandwidth.

8.4 Performance evaluation in a stadium

This section includes the results of the performance evaluation in a stadium as specified by the test case 4 of the METIS project.

8.4.1 Description of the test case

The stadium use case relies on an existing market, where operators experience today a difficulty in providing a service with a good quality of experience; providing then service with high level of quality of experience could be considered as a thorough new market. The mentioned difficulty is mainly related to the extreme crowdedness of the stadium that requires very specific deployments.

In this sense, this can be considered as mostly an operator-centric use case. From a technical point of view, the general challenge is to offer a reliable and extremely huge bandwidth service to a multitude of users temporarily located in a small area.

The situation is a football match that gathers many people interested in watching and exchanging high-quality video contents. People can exchange multimedia content inside the stadium or transmit them outside, particularly during the intervals of the main event in the stadium. Although two kinds of categories are described in [114], simulations only focus on the "pull" category, where spectators consume traffic.

8.4.2 Main KPIs and requirements

Main KPIs of Test Case 4 (TC4) are summarized in Table 8.2. More details on the definition of these KPIs can be found in [114]. The focus of this study is mainly on the latency in the radio access network, which must be lower than 5 ms.

Table 8.2: Main KPIs defined for TC4.

KPI	**Requirement**
Traffic volume per subscriber	9 [Gbyte/hour/subscriber] (DL+UL in busy period)
Average user data rate (busy period)	0.3-3 [Mbps] (DL+UL)
Traffic volume per area	0.1-10 [Mbps/m2]
Experienced user data rate	0.2-20 [Mbps] (DL+UL)
Latency (radio access network)	<5 [ms]

8.4.3 Simulation models

Simulations follow the models included in [115]. In the following, a short description of those models and some assumptions used in this Thesis are presented.

Environmental model

The environment of TC4 is limited to the stadium area. Platforms for spectators are roofed and tilted in order to provide a good visibility to the audience. Hence appropriate modelling of stadium requires 3D dimensioning. All area of stadium except the playground is covered with a deck at the height of 33 m. The angle of the tribunes is 30° with respect to the ground. For the sake of simplicity, the focus is only on one side of the bleacher, more specifically, on an area of 30 m x 50 m. Figure 8.11 presents this environment model. The area under study is the blue-coloured bleacher while the red coloured area is simulated in higher detail.

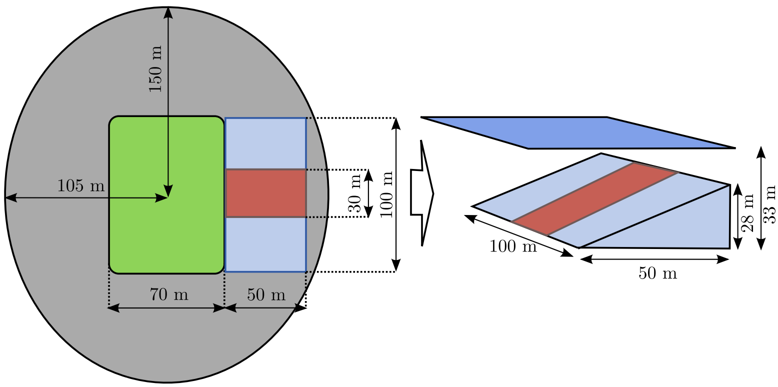


Figure 8.11: Detailed stadium environment model and area of study. [115]

Deployment considerations

There is a dense network of small cells deployed at the rooftop of the stadium and directed toward the audience. Thirty small cells access points are connected with optical fibre to a common baseband unit. Both sub 6 GHz and mmW deployments are allowed for stadium, although the focus will be on below 6 GHz assumption.

This scenario is assumed to be isolated from outside interferences. By default, small cells are deployed on out band frequency with respect to macro layer.

In the bleacher under study there are 9751 users distributed uniformly in the considered area. The distance between the users is 1 m along the minor stadium axis and 0.5 m along major stadium axis. There are 49 rows with 199 UEs each. Additionally, users separated along minor axis have different height, linear to slant of the stadium (30°).

Antennas of different cells are all deployed at the height of 33 m, with horizontal plane separation of 10 m along the major stadium axis and 15 m along minor stadium axis (cells at x=10 m, x=25 m, and x=50 m). To avoid inter-cell interferences, the antennas are highly directive and all of them are angled with respect to the roof plane orientation. The total output power for small cell is limited to 30 dBm.

Figure 8.12 shows a detail of the deployment of small cell antennas. This figure includes also the geometry of the different row of antennas and how distance from users to small cells depends on the row where users are located.

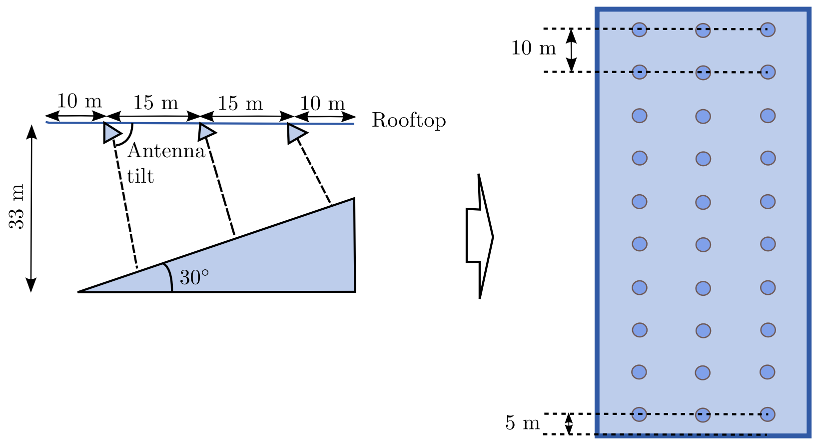


Figure 8.12: Details of the deployment of small cells in the deck of the stadium. [115]

Propagation model

The propagation model for TC4 needs to characterize Line of Sight (LoS) transmission from the small cell antennas deployed at the deck of the stadium and targeted at the audience. For this purpose, an outdoor UMi LoS model was used as defined in [7].

It is necessary to consider the next points in the modelling:

- the user relative height is 1 m above tribune level,
- distance between UE and small cell antenna is a 3D distance,
- although no mobility of users is assumed for stadium, a velocity of 3 km/h should be used to account for small-scale effects.

Traffic model

The FTP-traffic model described in [115] was used, where users were downloading 50 Mbyte files every 20 s. This results in a downlink traffic of 9 Gbyte/h.

Mobility model

The users were stationary for the duration of the simulation.

8.4.4 Assumptions

This subsection describes some assumptions, not originally described in [115], defined and used in this Thesis to complete the simulation framework.

In the simulations, the access points are connected to a central baseband unit via fibre optics. This central entity is also connected through fibre optics to a security gateway that provides access to the mobile network operator core network. This architecture is depicted in Figure 8.13. Given the availability of the central entity, the system could operate as a cloud Radio Access Network (RAN) in the sense that access points are seen by the network as a set of transmission points configuring a virtual MIMO system. Another possibility is that all access points act as conventional cells. Simulations will show the potential benefit of the cloud RAN solution.

Concerning the traffic flows, 85% of users are receiving video traffic in DL (50 Mbyte each 20 s, i.e. 20 Mbps), whereas the remaining 15% of users are exchanging data with D2D operation mode in both directions. The study of the D2D communication is out of the scope of this Thesis. Therefore, only the 85% of users is relevant to this study.

Concerning the propagation, for the AP-to-UE links, we used TC4 path loss maps available at [119].

Initially, 180 MHz are allocated to cellular communications at 2.6 GHz. Stations and users have 8 antennas (4 cross polarized pairs) and antennas at base stations have 17 dBi gain.

Given the relevance of confining transmitted signal to the area of interest, we are working on the following customized configuration of antennas:

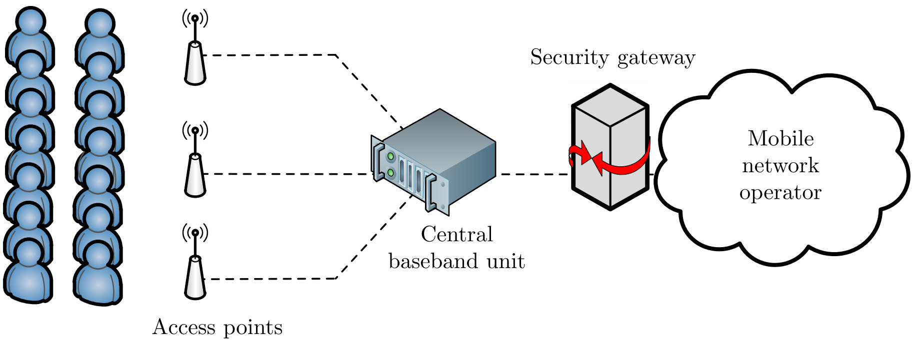


Figure 8.13: Architecture of the radio access network in TC4 evaluations.

- Antennas at x=10 m: tilt=95°, $\theta_{3dB} = 10°$, $\psi_{3dB} = 15°$
- Antennas at x=25 m: tilt=85°, $\theta_{3dB} = 15°$, $\psi_{3dB} = 20°$
- Antennas at x=40 m: tilt=65°, $\theta_{3dB} = 20°$, $\psi_{3dB} = 35°$

being θ_{3dB} the beamwidth in the horizontal plane and ψ_{3dB} the beamwidth in the vertical plane. The resulting wideband signal to interference plus noise ratio can be seen in Figure 8.14.

8.4.5 Technology components

This subsection describes a set of technology components, or technology solutions, added to LTE-Advanced to reach the required KPIs in this test case.

The main characteristics of this test case are the huge amount of data and the ultra-density of access nodes. As in TC1, there exists a central unit able of assuming the control of the whole system. This enables the cloud RAN concept that is thoroughly investigated in a progressive manner, including SU-MIMO, MU-MIMO, coordinated MU-MIMO and cloud RAN.

On the other hand, working with beyond 6 GHz bands, the frame structure must be redefined. In the case of cloud RAN with a massive availability of antennas, massive MIMO techniques will be used. Moreover, other subsidiary techniques will be used, to reduce interferences and to facilitate D2D communications. More details on the exact implementation of massive MIMO or D2D communications are out of the scope of this Thesis since they are not proposed in the framework of this dissertation, but the reader may refer to [116] for more details.

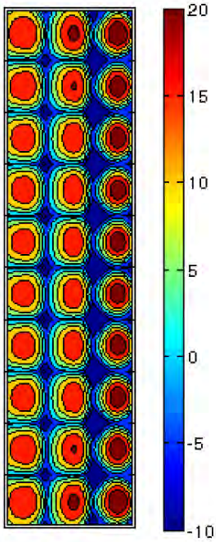


Figure 8.14: Wideband SINR [dB] with customized tilt and antenna pattern in the bleacher.

8.4.6 Results

In our simulations, a 20 Mbps traffic source has been used, with the time between packets fixed to 1 ms. This traffic source generates a flow of packets common to all the receivers, i.e., it emulates the broadcasting of a service over the stadium.

We obtain the mean radio access latency of these packets, and the mean experienced user data rate of each user. In order to calculate the latter, we average the data rate of each packet received by a user. The data rate is defined as the packet size divided by the time required to transmit the packet over the air, plus the time between retransmissions if needed. Additionally, we provide the packet loss percentage to have a complete picture of the system performance.

In the configuration labelled as "MU + CoMP" 8 antennas are placed at transmitter and receiver and coordination is implemented among preconfigured clusters of 9 cells. In configuration "MU + CoMP + MM", base station has a 64-antenna elements array. It is assumed that this 64 elements array is equivalent to an 8 elements array with 9 dB antenna gain in each element.

Results in Table 8.3 show that increasing the available bandwidth 20 times from the original assumptions in METIS [115], i.e. 180 MHz, up to 3.6 GHz is enough to reach the requirements of this test case for any of the MIMO schemes studied. They also show that there is a significant gain in using massive MIMO scheme in this scenario. This means that using multiple antennas to have the same multiplexing gain as a smaller array is useful in this scenario. In fact, with 1.8 GHz and a combination of MU-MIMO and massive MIMO, all the requirements are fulfilled, even without the use of CoMP.

Table 8.3: Summary of simulation results in TC4.

Configuration		**Experienced user data rate [Mbps]**	**Latency in radio access network [ms]**	**Packet loss [%]**
Requirement		0.3-20	< 5	N.A.
3.6 GHz	MU	5%-tile: 1.4 80%-tile: 20.0	4.69	4.8
	MU + CoMP	5%-tile: 16.1 80%-tile: 20.0	3.60	0.03
	MU + MM	5%-tile: 15.1 80%-tile: 20.0	3.61	0.03
	MU + CoMP + MM	5%-tile: 20.0 80%-tile: 20.0	3.42	0.00
1.8 GHz	MU	5%-tile: 0.8 80%-tile: 20.0	6.20	14.4
	MU + CoMP	5%-tile: 1.2 80%-tile: 20.0	5.15	5.5
	MU + MM	5%-tile: 4.4 80%-tile: 20.0	4.32	2.2
	MU + CoMP + MM	5%-tile: 14.4 80%-tile: 20.0	4.32	0.2

Figure 8.15 presents the CDF of user experienced data rate with 3.6 GHz of bandwidth. As expected, the maximum value is the data rate of the traffic

source, 20 Mbps. More than 20% of the users present this data rate for all the configurations.

Figure 8.16 shows the packet latency CDF for the same bandwidth. More than 80% of packets are received in less than 5 ms. Therefore, latency goal is clearly fulfilled as shown in Table 8.3.

Figure 8.17 presents the CDF of user experienced data rate with 1.8 GHz of bandwidth. As expected, the maximum value is the data rate of the traffic source, 20 Mbps. More than 20% of the users present this data rate for any configuration. Compared with the data rates achieved with 3.6 GHz, an increasing number of users get data rates that are lower than 20 Mbps.

Figure 8.18 shows the packet latency CDF for the same bandwidth. More than 60% of packets are received in less than 5 ms. However, latency goal is fulfilled only by the combination of MU-MIMO, massive MIMO and CoMP, as shown in Table 8.3. It is worth noting that the role of CoMP is of paramount importance to reach the defined goals. Indeed, even in a scenario with a huge pointing and confinement of signals (obtained with directive antenna patterns and additional beamforming in the hybrid architecture used for massive MIMO), using coordinated scheduling controlled by a central entity allows reducing interferences, which benefits especially cell-edge users.

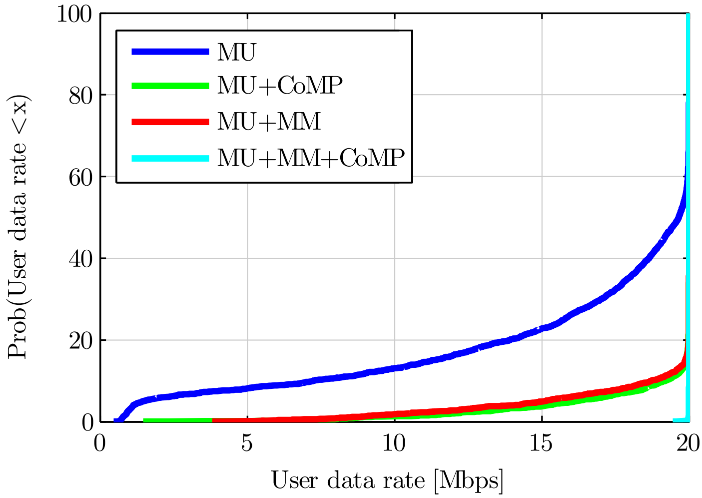


Figure 8.15: Experienced user data rate CDF with 3.6 GHz bandwidth.

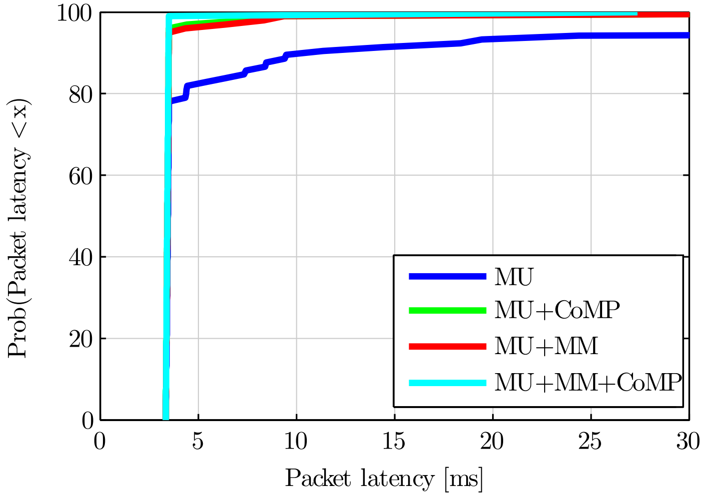


Figure 8.16: Packet latency CDF with 3.6 GHz bandwidth.

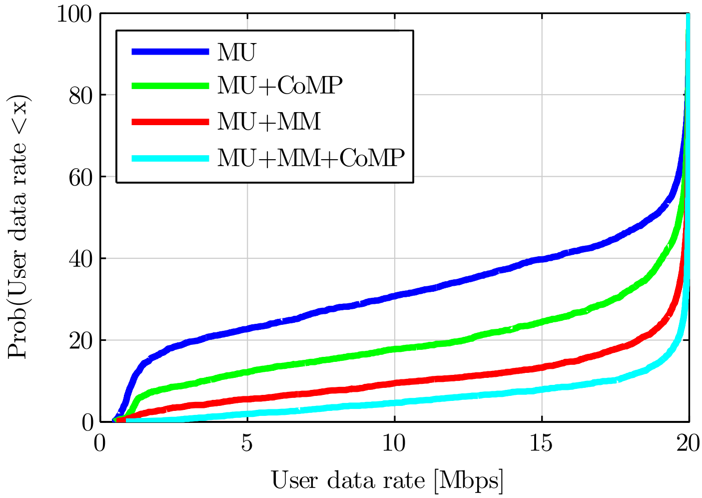


Figure 8.17: Experienced user data rate CDF with 1.8 GHz bandwidth.

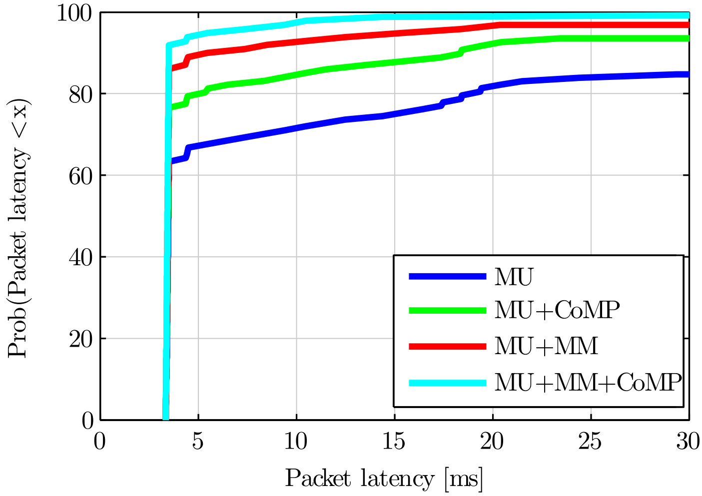


Figure 8.18: Packet latency CDF with 1.8 GHz bandwidth.

8.5 Conclusions

This chapter has presented some system-level results aimed at providing some views on the impact of CoMP in the 5G system performance. The study is focused on two specific 5G scenarios whose definition has been partially done in this Thesis.

Simulation results have proven the potential of some of the techniques discussed along the Thesis and reinforce the need for considering the implementation of CoMP techniques in the future 5G radio access technology natively.

Of significant relevance is the combined use of CoMP and short frame lengths, around a quarter of a millisecond. This reduction allows for a better management of interference in a heavily congested scenario and deserves further investigation. Additionally, results pointed out the need for a common inclusion of massive MIMO with multi-user allocation and CoMP techniques. This combination poses severe constraints in the computational burden in the base stations but has demonstrated to be the only alternative, apart from an enormous increase in the bandwidth allocation, to make cellular systems reach the challenging requirements of the 5G.

Chapter 9

Conclusions and Future Research

9.1 Concluding remarks

Two are the main problems treated in this thesis. First, there is a need to evaluate the Long Term Evolution (LTE) and LTE Advanced (LTE-Advanced) standard considering a realistic network deployment. The second problem is the need to develop realistic and practical coordination techniques among base stations, constrained to the framework of Coordinated Multi-Point (CoMP), and to evaluate the performance of such kind of coordinated schemes in a realistic deployment.

Concerning the first problem, the evaluation of LTE and LTE-Advanced, many assessments have been made in this field but those lack the required accuracy or are based on drastic simplifications. This Thesis has made an evaluation that takes into account the impact of channel implementation, channel estimation, interference model and link abstraction on the system assessment results. Moreover, the interactions between building blocks of LTE have been studied in detail, mainly in which concerns the interactions between scheduling, Link Adaptation (LA), multi-antenna schemes and CoMP. For example, performance of scheduling is affected by Hybrid ARQ (HARQ) constraints and this fact has been investigated in this Thesis. Moreover, LTE is a standard opened in some issues, then, different implementations are allowed and encouraged. This fact implies that the assessment of a particular implementation is only valid for the conditions of this assessment, including the specific implementation and deployment scenario. Although reference implementations can be used to perform LTE assessments providing reference performances figures, results obtained with these implementations can be improved if more developed algorithms or techniques are used. This Thesis has researched on implementations better suited to real conditions that maybe are not so appealing under synthetic assumptions.

Concerning the second problem, many proposals about CoMP assume non-realistic and/or non-practical assumptions. Moreover, first Third Generation Partnership Project (3GPP) proposals are restricted to intra-site coordination. This Thesis has developed a realistic, practical and low complexity inter-site capable coordination technique constrained to the limitations of the 3GPP CoMP framework. Additionally, the performance of CoMP has been evaluated both in synthetic and realistic deployments concluding the interest of feasible iterative approaches for coordinated transmission. Results point towards the need to natively take into account CoMP features in the lean design of the future Fifth Generation (5G).

It is worth to analyse now in detail the set of hypothesis that were formulated at the beginning of this Thesis work and see to which extent the Thesis answered these questions:

- Hypothesis 1: system performance is affected by system modelling precision. This hypothesis has been corroborated in several ways. For instance, it has been studied the effect of the accuracy of interference modelling on the Signal to Interference plus Noise Ratio (SINR) experienced by the users. Additionally, the effect of channel estimation error has proven to be of significant importance. Also, the results are dramatically different when transiting from synthetic to realistic results. Even the conclusions are different in some cases, since for instance the benefit derived from CoMP use is much more important under situations with imbalanced interference distribution.

- Hypothesis 2: key building blocks of the system interact and the joint effect is not just the addition of particular effects. This hypothesis has been also confirmed with the investigations about the interaction of scheduling and LA processes. As shown in Chapter 5, the performance of schedulers is heavily affected by knowledge of HARQ status. Additionally, the way in which scheduling decisions are made has an impact on the interference variation as shown by the flash-light effect.

- Hypothesis 3: link adaptation algorithms for Multiple-Input Multiple-Output (MIMO) can be improved in LTE-Advanced. LA topic has been thoroughly analysed in Chapter 4 where correction processes, including Inner Loop Link Adaptation (ILLA) and Outer-Loop Link Adaptation (OLLA) loops, have been explained, and two precoding schemes have been detailed for non-codebook based Multi-User MIMO (MU-MIMO) transmission. Indeed, interference consideration in the LA processes resulted of paramount importance to allow for a more stable behaviour of these interferences, thus reducing the flash-light effect problems.

- Hypothesis 4: low complexity inter-site CoMP algorithms can be defined in LTE-Advanced providing a considerable improvement on the system performance compared to non-coordination and intra-site coordination in realistic conditions. This Thesis has evaluated this hypothesis in the framework of the 5G Mobile and wireless communications Enablers for Twenty-twenty Information Society (METIS) scenarios, where the cloud radio access network concept is gaining momentum. Indeed, coordination resulted necessary to increase users performance in the cell borders mostly in congested situations where interference reduction and coordination is a must.

- Hypothesis 5: ideal CoMP algorithms are heavily affected by non-idealities while sub-optimal solutions can be more robust to non-idealities and hence be more suited to actual scenarios. Results demonstrated that the CoMP scheme proposed in this Thesis, feasible and based on non-ideal information, is indeed robust to these non-idealities.

9.2 Discussion on the future of the 5G

This Thesis has not fully dealt with the hot topic in current research about how the 5G will look like, or which would be the main technology enablers of the 5G. However, the activity of evaluation performed in this Thesis allows to elaborate on the potential impact of some relevant technology enablers. The observations and initial conclusions highlighted here are based on the specific setup and simulation assumptions used in the evaluations partially included in Chapter 8. Therefore, changes in the specific evaluation setup as well as the baseline systems may lead to different conclusions. Nevertheless, these initial observations provide important hints about the potential impact of different technology enablers on the future 5G system.

In general, more spectrum, higher spectral efficiency and reduced communication distance (e.g., network densification, local offload) are all important enablers to achieve technical objectives such as 1000× increment of throughput, 10 to 100× more typical user data rates, 10 to 100× higher number of connected devices and 5× reduction in End-to-End (E2E) latency. The rest of this section summarizes the potential impact of specific instances of the above enablers.

Use of spectrum

By using the simulation setup described in [116], spectrum bandwidths of at least 1500 MHz for indoor and 650 MHz for outdoor METIS 5G test cases described in [114] are required to meet the respective demands. These figures assume best case scenario, i.e., no cost constraints on the level of possible densification and the presence and utilization of additional technology components that improve spectral efficiency. Thus, much higher bandwidths will be needed when these assumptions are relaxed.

Due to the favourable propagation conditions at lower frequency bands, it is expected that such frequencies will be used for providing ubiquitous coverage in outdoor environments. Given the limited contiguous bandwidths of spectrum in the lower frequency bands, it is expected that additional frequency bands in higher frequencies will be needed and used to provide additional capacity where required. Higher frequency bands face challenging propagation conditions, but could be suitable for use in outdoor hotspots (e.g. squares, campus, stadium, business districts, etc.), indoors (e.g. airports, fairs, malls, enterprise, home, etc.), or fixed wireless links between buildings and backhaul.

System densification

Densification is needed to fulfil the 5G requirements in some specific scenario described in [116]. In the evaluation setup used, between 1 and 2.5 nodes per 100 square meters are needed indoors, depending on the available bandwidth and carrier frequency, whereas a single node for each 400 square meters suffices for outdoors under the conditions studied in [116].

Concerning the impact of higher dense networks, it is worth noting that capacity is directly proportional to the number of nodes, provided centralized interference coordination. For indoor cases, the coefficient of proportionality could be as high as 0.73 with 1 node per 100 square meters. For outdoor, as compared with LTE, increasing the available bandwidth with additional 100 MHz and using three times more nodes, capacity could be boosted by a factor of 10. As compared with the baseline deployment assumed, the levels of densification that are foreseen are from 3 to 5 times higher. With such a level of densification, new paradigms and tools for network deployment will, therefore, be needed in order to cost-effectively deploy and manage such dense networks. For instance, nomadic nodes in vehicles could be one means to temporally increase the level of network densification.

Cell coordination

As pointed out in this Thesis, the relevance of cell coordination increases with densification. In particular, our evaluations demonstrate that the use of CoMP techniques among adjacent nodes can realize significant performance gains. If system is interference limited, the performance mostly in the cell border changes drastically with the use of these techniques. Proper clustering must be used, being it dynamic and sensitive to the level of isolation between transmitters. Within each cluster, one cell could work as controller, since centralized coordination outperforms full-distributed coordination.

Finally, results show that coordination without a reduction in the transmission time interval could damage average performance. In this sense, transmission intervals must be reduced down to a quarter of millisecond to make the most of such coordination.

D2D and V2X communications

Direct communications among nodes is one of the main drivers contributing to increased capacity, reduction in latency, and support for a massive number of devices. D2D and V2X communications could increase the system capacity by a factor of two assuming an opportunistically shared spectrum [116].

Without any doubt, the introduction of this new paradigm of communication requires the network control to avoid some important issues, like interference management (hidden node problem), security and service announcement overhead. One of the main impacts of this new direct communication channel is the reduction of the average latency, since it allows for latencies in the order of 1 to 2 ms. Network infrastructure is not involved in the data plane, which obviously improves the final transmission latency.

In any case, direct mode operation in critical situations requires optimal cluster head selections and energy saving techniques.

New waveforms and multiple-access schemes

5G will likely comprise a flexible set of radio waveforms (likely separated in different carriers). Three waveform candidates are being discussed, namely Orthogonal Frequency Division Multiplexing (OFDM), Filtered Bank Multi-Carrier (FBMC), and Universal Filtered OFDM (UF-OFDM). For the classical broadband communications via a cellular network all waveform candidates achieve similar performance. Nevertheless, OFDM-based ones have a strong case due to its already wide-spread usage in current cellular networks. Moreover, new signalling procedures are to be implemented for machine type communications, including the possibility of narrowband transmissions (much less than current 180 kHz) and the use of FBMC is a potential solution to be explored in the future.

With FBMC, due to the very good frequency localization, the transmit power can be concentrated on only very few subcarriers to eventually enhance significantly the expected coverage or to reduce battery consumption. Finally, with FBMC, uplink synchronization needs can be relaxed, thus reducing access time and removing timing advance procedures.

Massive MIMO

Massive MIMO must reach the 256x256 scheme to satisfy 5G requirements. Spectral efficiency can be increased by a factor of 20 with this setup as compared with 4x4 antenna systems. Of course, we do not foresee 256 antenna elements in the handheld terminal. However, an equivalent 256 antenna receiver can be configured multiplexing with MU-MIMO 8 users, 32 antennas each. For the same spatial multiplexing capability as legacy systems (8 streams), beamforming gain reaches 15 dB.

Form factor makes the use of Massive MIMO more attractive for cmW and mmW bands. This statement fits quite well together with the proposal of using higher frequencies above 6 GHz due to the reduced antenna size. Therefore, Massive MIMO use is mainly expected for wireless backhaul, indoor scenarios and hotspots (stadiums, concerts, malls) provided the LoS condition.

The Cloud Radio Access Network (RAN) concept is also highly related with the flexible definition of Massive MIMO transmission schemes. If an optical fibre distribution system connecting several infrastructure nodes is available, this could form a virtual distributed array with massive antenna availability.

9.3 Future research topics

The more I know, the more I realize the beauty of the Socratic paradox "I know that I know nothing". This Thesis has opened the door for many other research topics that require further investigation. These new topics are well aligned with the research trends related with the development of the 5G mobile communications outlined in the previous section. In particular, the following aspects can be highlighted:

- New evaluation scenarios are being defined in the evolution towards 5G. Some of these scenarios are characterized by a more realistic reproduction of actual deployments. In fact, the work described in Chapter 8 contributed to the definition of the first scenarios developed in this process. Nevertheless, there is still the need to capture in these scenarios the requirements of the future 5G. This line is, in fact, a current research topic as it is being done in the European project METIS-II by the author of the Thesis.

- New channel models are being developed in the evolution towards 5G to capture more realistically the actual propagation effects. Development of models taking into account all the spatial dimensions (3D models) or new models for mmW bands are just a couple of examples. The more realistic modelling usually comes together with an increased computational burden of simulations. Therefore, it is envisaged the use of new methodologies to perform channel simulation. One methodology is the reuse of graphic engines used up to now in the computer games world that is a specific future research topic of this Thesis.

- Another line of research related to channel modelling is to investigate on the impact of more realistic and diverse models on MIMO schemes and CoMP. The continuous evolution of channel models towards more realistic ones done by the research community has lead to the finding that some schemes were overvalued when assessed with simple channel models. Therefore, there is a need to corroborate previous evaluation results with the new models, or, on the contrary, reject them. Moreover, new models and scenarios create new opportunities for multi-antenna schemes that should be further evaluated.

- This Thesis has considered a full-buffer traffic modelling as described in Chapter 7, and a simple constant bit rate model in Chapter 8. There is a need to further investigate the performance of the system to support more realistic traffic models. This includes the evaluation of schedulers beyond the proportional fair scheduler used in this Thesis.

- In order to develop a robust CoMP technique, the Channel State Information (CSI) knowledge has been assumed to be limited to wideband transmit channel covariance matrices estimated from uplink measurements. This embodiment of the CoMP feature would be also supported by the reporting of this information from the user side. However, this kind of report is not possible according to the current 3GPP specifications, and this does not seem to change in the near future. Therefore, a future line of research is the investigation of the impact of using real CSI reports supported by the 3GPP specifications to implement CoMP techniques.

- This Thesis includes the description and evaluation of a specific implementation of a Coordinated Scheduling and Beamforming (CS/CB)

CoMP technique. This particular implementation has provided significant benefits in some scenarios but not in others. The design of more complex CS/CB or feasible joint transmission schemes and the associated signalling is envisaged as a future line of research devoted to making the most of coordination in those scenarios where the proposed CS/CB is not useful. Additionally, it is planned to further investigate the performance of CS/CB schemes in more heterogeneous scenarios such as those envisaged in the future 5G, where it could be especially useful.

- The development process of new air interfaces for 5G has lead to the definition of some access schemes that do not rely on synchronization for the transmission. This fact motivates the definition of a new line of research to study the robustness of CoMP against this lack of synchronization, which could be particularly detrimental in joint transmission schemes.

- Under the assumption that the air interface operating above 6 GHz would operate in Time Division Duplexing (TDD) mode, there is a need to update current scheduling, link adaptation and CoMP proposals to this new mode of operation. This process should take advantage of the new possibilities that this mode brings to the channel knowledge. Additionally, it should be aware of new problems, such as increased interference variability, or the increased complexity of the scheduler.

- In the future mobile networks, end-user devices could be more than just a final receptor or initial transmitter. New capabilities such as the mobile relaying consider the mobile device as an additional part of the radio access network. Therefore, there is a need to clarify the role of mobile devices in CoMP.

Appendix A

Description of the System Level Simulator

A.1 Introduction

Guidelines and deployment scenarios to perform International Mobile Telecommunications Advanced (IMT-Advanced) evaluation are detailed in [7]. Following these guidelines a Long Term Evolution (LTE) Advanced (LTE-Advanced) system level simulation platform has been implemented in this Thesis, which is fully compliant with Third Generation Partnership Project (3GPP) specifications. Specifically, the Downlink (DL) of the system, that is, the link from the Base Station (BS) to the Mobile Station (MS) is emulated since this is the most demanding link. It is appropriate to say that implementation is focused on the Frequency Division Duplexing (FDD) version of the technology because it is envisaged that this version will be the predominant one.

After the first stage of simulation development, done in C++, a calibration process has been performed. This second stage is of paramount importance to ensure the validity of the results obtained with the simulation platform. Calibration is performed through comparison of the outcomes of the developed simulator with those of the simulation platforms of other research institutions. This work has been performed in the framework of the European project Wireless World Initiative New Radio + (WINNER+).

After calibration of the simulation platform, the capabilities of LTE and LTE-Advanced have been evaluated, and the results of this evaluation are shown along the Thesis.

In the final part of the Thesis, the simulator has been extended to include new capabilities required to perform simulations in the framework of the Mobile and wireless communications Enablers for Twenty-twenty Information Society (METIS) project. Nevertheless, these modifications are out of the scope of this appendix whose primary aim is to describe the core part of the simulation platform used in this Thesis. Therefore, the appendix will include some comments about the above mentioned modifications, but not an exhaustive explanation.

The organization of the appendix is as follows. First, in Section A.2 main building blocks of this simulator are presented, including their main features, possible implementation, and interactions. Second, the calibration process conducted to ensure the validity of the results obtained with the system-level simulation platform is explained and the results of this process are shown in Section A.3. These results confirm that the simulation platform developed is well calibrated.

A.2 Simulation platform functional description

A logical structure of the LTE-Advanced simulation platform is shown in Figure A.1. Important interactions among functional entities are shown as connections among blocks in the figure. The components shown and their interactions are described in the following subsections.

A.2.1 Network layout

The network layout is a logical entity whose main function is to store the location of each BS and each MS. Additionally, this module stores the interfering relations among base stations, that is, which BSs produce interference to the users served by each BS.

Two BS layouts are considered in IMT-Advanced simulations: a hexagonal grid for outdoor scenarios and a particular office layout for indoor scenarios.

Users are distributed uniformly over the whole area and in IMT-Advanced evaluations and their position is kept fixed during a simulation run.

A.2.2 Channel

This module is a generic entity employed by any wireless link established in the simulation. The channel module models the radio propagation conditions between a BS and a MS.

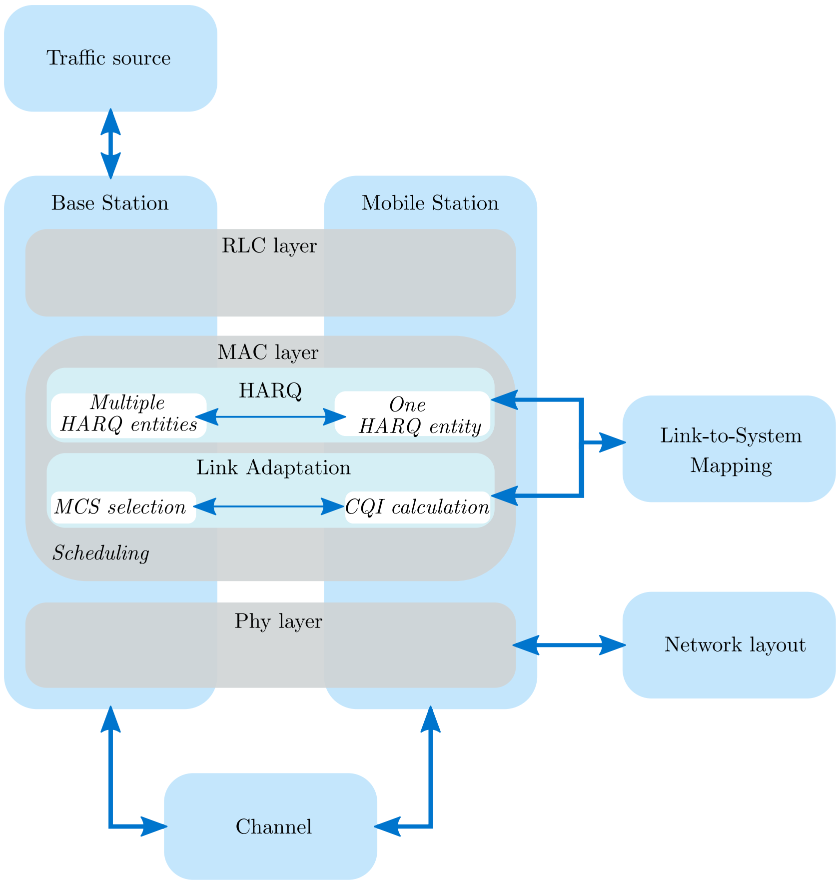


Figure A.1: Simulation platform functional scheme.

The simulation platform includes a proprietary implementation of the IMT-Advanced channel model defined by the International Telecommunication Union (ITU) Radiocommunication Sector (ITU-R) in [7]. This model is a stochastic and geometric channel model that allows creating a bidirectional radio channel consisting of a plurality of rays. Although it is a geometric based model that knows the exact location of transmitting and receiving elements, it does not specify the position of the scatterers, rather only ray directions are known.

In order to generate channel samples between one transmitter and one receiver, mobility and exact location of both ends must be known. Based on this information all large-scale parameters are generated, followed by the small scale parameters. All these parameters are kept constant during the whole simulation. Channel samples are obtained by adding the contributions of all involved rays.

There are five different scenarios and thus channel models: Indoor hotspot (InH), Urban Micro-cell (UMi), Urban Macro-cell (UMa), Suburban Macro-cell (SMa) and Rural Macro-cell (RMa). These different models are characterized by different parameters of the statistical distributions used to generate the channel samples.

A.2.3 Link to System Mapping

Link to System (L2S) mappings are developed through link level simulations to provide an abstraction of the complex processes that occur at the link level. In a system level simulator, link abstraction models are used to translate channel quality measures, such as Signal to Interference plus Noise Ratio (SINR) values, to transmission quality values, such as Block Error Rate (BLER) or throughput.

A family of methods known as Effective SINR Mapping (ESM) is commonly used. These methods map an instantaneous set of SINR samples into a scalar value, which is called effective SINR. The ESM models implemented in the simulation platform are the Exponential ESM (EESM) and the Mutual Information Effective SINR Mapping (MIESM) that uses as its information measure the so-called mutual information [76].

In the simulation platform, the L2S mapping is based on link level results obtained with a link level simulator in whose development participated the author of the Thesis. It is worth noting that link level results obtained in this Thesis have been widely used in WINNER+ by third parties in the evaluation of IMT-Advanced technologies. Section 3.4.1 contains an example of those results.

A.2.4 Base Station and Mobile Terminal

BSs and MSs are the main simulation entities. Both entities present a layered architecture consisting of three main layers: the physical layer, Medium Access Control (MAC) layer and Radio Link Control (RLC) layer. Next, functions and peculiarities of these layers are described.

Physical layer

The main function of the simulation platform physical layer is the calculation of the SINR obtained in the MS after the receiver filter and before the decoder. To this end, the physical layer has an interface with the channel module, to know the state of the channel links, and also with the network layout, since interfering relations are contained in this module. Both the SINR of reference signals and data packets sent by the BS are calculated. Different receiver filters have been modelled: for Single Input Multiple Output (SIMO) the Maximum Ratio Combining (MRC) is used while for Multiple-Input Multiple-Output (MIMO) the Minimum Mean Square Error (MMSE) receiver with interference suppression or interference cancellation is considered.

MAC layer

The MAC layer of the simulation platform presents three main functions:

1. Hybrid ARQ (HARQ) management

   The developed LTE-Advanced simulator emulates in detail all the HARQ functionality. In the simulation platform, each time a packet is transmitted from a BS, the MS receives HARQ information (e.g. Modulation and Coding Scheme (MCS) of this packet) from its pair entity and SINR information from the physical layer. All this information is translated into a BLER value thanks to the L2S mapping. Randomly, and taking into account the BLER calculated, the transmitted block can be erroneously received. Soft combining is implemented by each HARQ process by the combination of SINR values of retransmitted packets.

2. Link Adaptation (LA)

   One of the key techniques of LTE-Advanced is the LA that allows the transmitter to adapt the transmission format, including the MCS, transmission rank and precoder among other parameters, to the channel quality variations.

   In order to support the LA operation, several methods are implemented to perform the channel state report. First, explicit channel estimates can be reported to the BS. For example, one channel estimated can be reported per subcarrier. To reduce the overhead of such a transmission only channel correlation matrices could be transmitted. Besides these methods, other kind of non-explicit information could be transmitted. Following this approach, the MS performs SINR measurements and calculates the most suitable format for transmission. This information includes which is the most suitable MCS, multi-antenna scheme, and precoding matrices and rank in case of spatial multiplexing. This information is conveyed in a set of reports known as Channel Quality Indicator (CQI), Precoding Matrix Indicator (PMI) and Rank Indicator (RI). In this process, it is necessary to have an interface with the physical layer to know the channel state and also with the link abstraction model to be able to translate the channel state into a transmission quality estimate. The simulation platform generates CQI, PMI and RI reports according to the standard.

An additional source of information used to perform the LA is the HARQ feedback sent by the MS to the BS. For instance, if a negative acknowledgement is received by the BS, it means that the transmission format used in a previous transmission was incorrect and adaptation is needed. The simulation platform includes the algorithms described in previous chapters to optimize the LA.

3. Scheduling

   A scheduling module is implemented in the BS that makes resource allocation decision in each Transmission Time Interval (TTI). This module interacts with the HARQ and LA modules. Interaction with HARQ is necessary to decide if a user must perform a retransmission or a new transmission, and in the case of retransmission to know the format of such retransmission. Interaction with LA module is important to perform optimal decisions that take into account the channel quality experienced by each user. Several algorithms have been implemented in the simulation platform although, the Proportional Fair algorithm with several variants is the most relevant to this Thesis.

RLC layer

In the BS, the RLC entity receives packets from higher layers and multiplexes or divides them to adapt packet sizes to MAC layer needs. In the MS, the RLC entity receives packets from the MAC and reassembles the original packets to pass them to higher layers.

In the full-buffer simulations, RLC operation is not considered. However, in the simulations with realistic traffic performed in Chapter 8, RLC is used to perform more accurate evaluations

The implemented data transfer modes in the RLC sublayer are the Acknowledged Mode (AM) and Un-acknowledged Mode (UM) [122]. In AM, RLC headers are added to form the RLC Protocol Data Units (PDUs) in order to provide a reliable in sequence delivery service and enable the segmentation and reassembly or concatenation and demultiplexing tasks. In case of MAC transmission failure, retransmissions are carried out at RLC level. Moreover, the RLC entity performs RLC Service Data Unit (SDU) discarding when the maximum time delay set for upper layer packets is exceeded. In the simulations performed in Chapter 8, RLC is used in UM mode to prioritize latencies over packet losses.

A.2.5 Traffic source

Two traffic models are used for evaluation of proposed IMT-Advanced technologies: full-buffer traffic model and Voice over IP (VoIP) model.

In the full-buffer traffic model, there is an infinite amount of data bits waiting for transmission to each receiver entity. Although this model is not realistic, it can be used to get interesting insights of the spectrum efficiency of each system.

The VoIP model used in IMT-Advanced evaluation is detailed in [7]. It assumes a codec rate of 12.2 kbps, and an encoder frame length of 20 ms. It is also important to note that a voice activity factor of 50% has been considered. Although this model is implemented in the simulation platform, it has not been used in this Thesis.

Moreover, the simulations conducted in Chapter 8 consider a constant bit rate traffic model. It is worth noting that the traffic source that implements this modelling takes into account the overhead of the Transmission Control Protocol (TCP), Internet Protocol (IP) and Packet Data Convergence Protocol (PDCP) headers needed in End-to-End (E2E) transmissions. Nevertheless, those higher layers are not explicitly simulated.

A.3 Simulation platform calibration

In order to ensure the validity of the results obtained with the simulation platform, it is necessary to conduct a calibration process after the development phase. In this process, several tests are defined, and outcomes of different simulators are cross-checked to detect incoherences. If results are aligned, the simulators are considered well calibrated. Otherwise, more work is needed to achieve calibration. It is important to define the calibration tests correctly to ensure that the key building blocks of the simulation are calibrated.

A stepwise calibration methodology was followed in the framework of the WINNER+ project. This process consists of three steps: calibration of the large-scale parameters of the channel model, calibration of the small scale parameters of the channel model and calibration of a baseline system configuration. The first two steps involve mainly the calibration of the channel model, that is, a technology-agnostic building block, while the last step is focused on the calibration of the MAC layer and link abstraction model that are technology dependent.

A.3.1 Channel model calibration

In the calibration of the large-scale parameters of the channel model, all the multipath effects are neglected. Simulations are conducted to obtain for each user a path gain and a wideband SINR value. The path gain is defined as the average signal gain between a MS and its serving BS. It includes distance attenuation, shadowing and antenna gains. The wideband SINR is the ratio of the average power received from the serving cell and the average interference power received from other cells plus noise (only accounting for large-scale channel effects). Sometimes this measure is called geometry factor since it presents a high dependency on the positions of the BSs. Then, the calibration of this measure ensures not only the calibration of the channel model but also the calibration of the network layout. In addition, it involves the calibration of the physical layer considered in the simulation platform functional description, since the SINR is calculated in this layer. In addition to the evaluation principles and assumptions in [7] the assumptions shown in Table A.1 have been considered.

Table A.1: Additional simulation assumptions for large-scale parameters channel model calibration.

Parameter	Value			
Cell selection	1 dB handover margin			
Feeder loss	2 dB			
BS antenna tilt	InH	UMi	UMa	RMa
	0º	12º	12º	6º

Figure A.2 shows the Cumulative Distribution Function (CDF) of the path gain (left) and wideband SINR (right) obtained from a set of simulations conducted with the LTE-Advanced simulation platform developed in this Thesis. It has been shown the CDF for each one of the test scenarios considered in IMT-Advanced evaluations. Comparison of the presented results with those obtained in the WINNER+ project [123] and those obtained by the 3GPP [24] proves the calibration of the presented simulator.

In the calibration of the small scale parameters of the channel model, the set of characteristics whose distributions are investigated include the delay spread and the departure and arrival angular spread at the BS and MS, re-

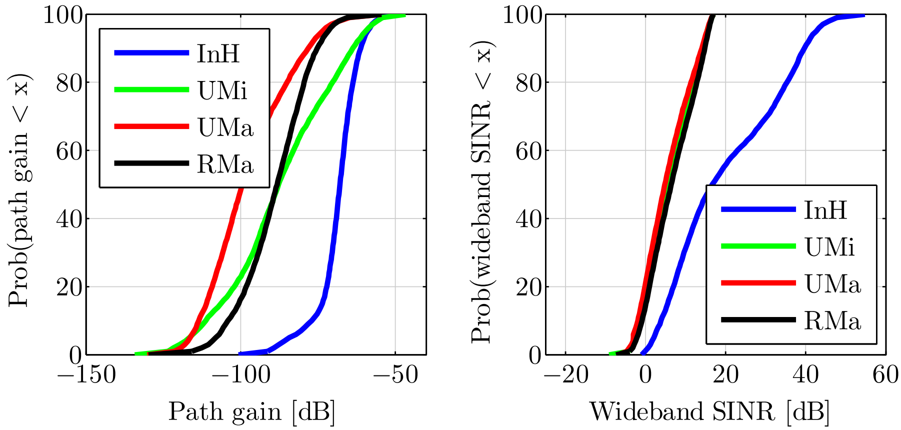


Figure A.2: Distribution of path gain (left) and distribution of wideband SINR (right) in the channel model large-scale parameters calibration step.

spectively. The root mean square DS and circular angular spread at the BS (Angle of Departure (AoD)) and MS (Angle of Arrival (AoA)) are calculated for a large number of radio links. Mathematical definitions of these spread measures are included in [123]. The calibration is performed separately for Line of Sight (LoS), Non Line of Sight (NLoS), and Outdoor to Indoor (OtoI) propagation conditions. For simplicity, this calibration step assumes omni-directional antennas at both the BS and the MS. Delay Spread (DS) distributions for each test scenario in LoS and NLoS are shown in Figure A.3, while in Figure A.4 and Figure A.5 the AoA spread and AoD spread are shown. The distributions of the small scale parameters have been compared with the results of other partners in the framework of the WINNER+ project [123], proving the correct implementation of the IMT-Advanced channel model in the LTE-Advanced simulation platform developed in this Thesis.

A.3.2 Baseline configuration calibration

In a third stage, the focus of the calibration is for the first time on the MAC layer and link abstraction model.

Spectral efficiencies, user throughput distributions and SINR distributions have been obtained and compared using a basic LTE configuration. In this configuration, baseline assumptions are made for non-standardized algorithms. For example, the chosen downlink scheduler is the Round Robin algorithm with

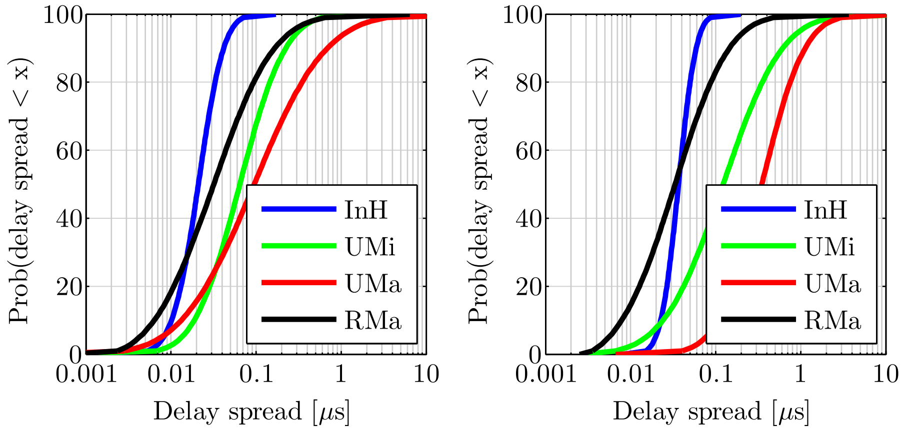


Figure A.3: Distribution of delay spread for LoS (left) and NLoS (right) in the channel model small scale parameters calibration step.

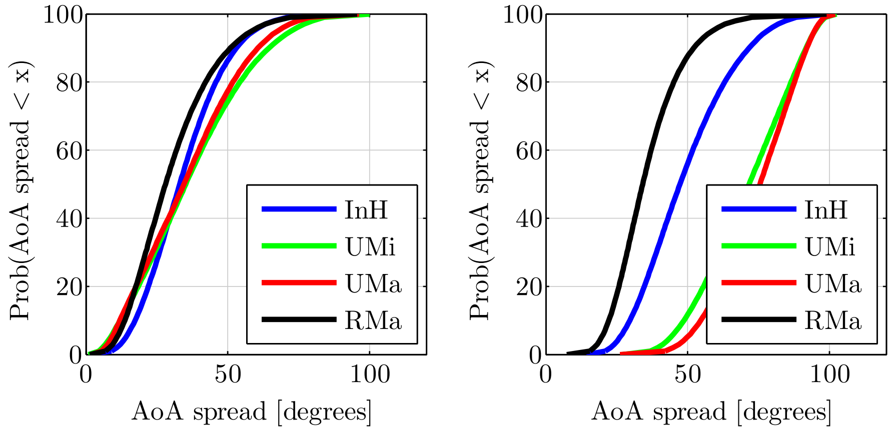


Figure A.4: Distribution of angle of arrival spread for LoS (left) and NLoS (right) in the channel model small scale parameters calibration step.

full bandwidth allocation, that is, the whole set of resources is allocated to one of the users served by a cell in each transmission time. The DL transmission scheme is SIMO with one antenna at the transmitter side and two antennas at the receiver side. The antennas are vertically polarized with 0.5 wavelength

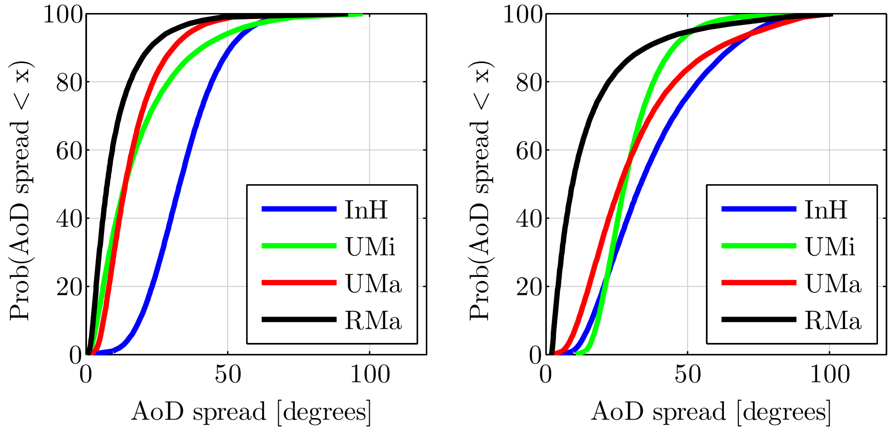


Figure A.5: Distribution of angle of departure spread for LoS (left) and NLoS (right) in the channel model small scale parameters calibration step.

separation at User Equipment (UE). In order to get a gain from diversity, MRC is used at the receiver. LA is achieved through a wideband CQI sent with a 5 ms periodicity and without measurement errors. Channel estimation is also ideal. A significant difference with the assumptions considered in the calibration of the large-scale parameters of the channel model is that in this stage feeder loss is assumed to be 0 dB. Table A.2 summarizes the considered simulation assumptions.

Results of this stage are presented using three performance indicators. First, cell spectral efficiency is calculated as the total amount of bits correctly transmitted by the BS per unit of time and frequency. Second, the cell-edge user spectral efficiency is the spectral efficiency met by at least the 95% of the simulated MSs. Both performance indicators are heavily affected by the number of active users in the scenario under test. A user density of 10 MS per cell has been considered in the simulations. The third performance indicator is the post antenna combination SINR, that is, the SINR obtained after the MRC receiver and before the decoder. A unique SINR value is obtained by linear averaging over time and frequency for each user. Spectral efficiencies are presented in Table A.3 while SINR and user spectral efficiency distributions are shown in Figure A.6.

Table A.3 includes the results from the simulator developed in this Thesis (labelled as "iTEAM"), from the 3GPP [24], and from WINNER+. The latter are obtained from internal information from WINNER+ similar to that found in [123]. These results are average values of the results of several evaluators.

Table A.2: Baseline configuration calibration simulation assumptions.

Parameter	Value
General	Parameters and assumptions not explicitly stated here according to ITU guidelines M.2135 and 3GPP specifications
Duplex method	FDD
Network synchronization	Synchronized
Handover margin	1 dB
Downlink transmission scheme	SIMO 1×2
Downlink scheduler	Round robin with full bandwidth allocation
Downlink link adaptation	Wideband CQI, no PMI on PUCCH (mode 1-0), 5 ms periodicity, 6 ms delay total (measurement in subframe n is used in subframe n+6), CQI measurement error: None, MCSs based on LTE transport formats [78]
Downlink HARQ	Maximum four transmissions
Downlink receiver type	MRC
Antenna configuration	Vertically polarized antennas, 0.5 wavelength separation at UE, 10 wavelength separation at base station
Channel estimation	Ideal, both demodulation and sounding
Control channel overhead, acknowledgements etc.	L=3 symbols for DL CCHs, M=4 resource blocks for UL CCH, overhead for demodulation reference signals.
BS antenna downtilt	InH scenario: 0^o UMi & UMa scenarios: 12^o RMa scenario: 6^o
Feeder loss	0 dB
Inter-cell interference modelling	Explicit

In order to perform quantitative characterization of calibration quality, 3GPP and WINNER+ used a parameter known as coefficient of variation calculated as $c_v = \frac{\sigma_{SE}}{\mu_{SE}}$ where σ_{SE} and μ_{SE} are respectively: standard deviation and mean of the cell or cell-edge spectral efficiency of the results obtained from the multiple evaluators. Table A.4 shows the coefficients of variation calculated by 3GPP and WINNER+. Tables A.3 and A.4 demonstrate that results obtained with the LTE-Advanced simulation platform are consistent with those obtained in the WINNER+ project and in the 3GPP since result differences are acceptable given the presented coefficients of variation.

Table A.3: Cell and cell-edge user spectral efficiencies.

Metric	Evaluator	Scenario			
		InH	UMi	UMa	RMa
Cell spectral efficiency (b/s/Hz)	iTEAM	2.6	1.2	1.0	1.2
	3GPP	2.3	1.2	1.0	1.2
	WINNER+	2.5	1.2	1.0	1.1
Cell-edge user spectral efficiency (b/s/Hz)	iTEAM	0.082	0.027	0.022	0.027
	3GPP	0.082	0.028	0.022	0.027
	WINNER+	0.078	0.029	0.019	0.024

Table A.4: Coefficients of variation of the results.

Metric	Evaluator	Scenario			
		InH	UMi	UMa	RMa
Cell spectral efficiency	3GPP	3%	9%	8%	6%
	WINNER+	6%	9%	12%	6%
Cell-edge user spectral efficiency	3GPP	9%	19%	17%	14%
	WINNER+	20%	28%	26%	16%

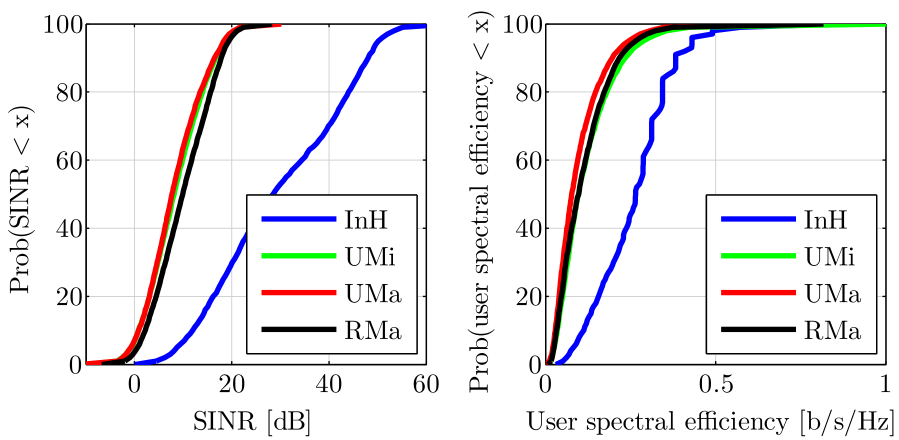


Figure A.6: Distribution of SINR after antenna combination (left) and distribution of user spectral efficiency (right)

Appendix B

Statistical Significance Assessment

This appendix describes the methodology used in this Thesis to assess the statistical relevance of the studied performance indicators with emphasis on two widely used ones: the cell spectral efficiency and cell edge user spectral efficiency.

Three simulation parameters affect the statistical relevance of the results:

- N_{sim}: the number of simulation runs.
- T_{sim}: the simulation duration.
- ΔT_{sim}: the initial transient time in which statistics are not collected.

The significance of T_{sim} and ΔT_{sim} is related to the convergence of the indicators in a simulation run. Typically, in a simulation run, the system starts in a transient state and ends with a steady state. This fact has an impact on the evolution of the performance indicators. At the beginning of the simulation, they can change at a fast pace until the steady state is reached when they tend to converge to a certain value (with some oscillation around this value). In order to obtain stable results it is necessary to perform simulations with enough duration so as to ensure that the system is in the steady state, hence the importance of configuring T_{sim} appropriately. Besides, it is useful to consider an initial time in which statistics are not collected, ΔT_{sim}, avoiding the divergence of results that is characteristic of the transient state. Taking into account this time accelerates the convergence of the results. Typically, the time needed by the simulation to converge is reduced as ΔT_{sim} increases, up to a certain value.

In order to determine the T_{sim} and ΔT_{sim} to be used, a series of long simulation runs are first conducted. T_{sim} is the time needed to ensure convergence. ΔT_{sim} is selected as the highest value beyond which T_{sim} is no further decreased.

Multiple simulation runs must be performed to estimate the studied performance indicators. Usually, the precision of this estimation increases with the number of simulations, N_{sim}. Ideally, the value of the indicator should converge to certain value as the number of simulations increases.

Our approach to determine N_{sim} is to perform a high number of simulations and study how the performance indicator evolves with the increasing number of simulations. The indicator should converge to a given value.

Using this procedure we consider that significant results can be obtained in the simulations of this Thesis where spectral efficiency results are provided with a ΔT_{sim} of 0.2 seconds. The values of N_{sim} and T_{sim} depend on the specific scenario under study. In the InH scenario, at least, 100 simulations of 10 seconds are performed, while in the rest of scenarios 10 simulations of 5 seconds are needed to have enough statistical significance.

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