A Radio Resource Management Framework for the 3GPP LTE Uplink

By

Amira Mohamed Yehia Abdulhadi Afifi
B.Sc. in Electronics and Communications Engineering – Cairo University

A Thesis Submitted to the
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in Partial Fulfillment of the
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Giza, Egypt
2011
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<tr>
<td>AMC</td>
<td>Adaptive Modulation and Coding</td>
</tr>
<tr>
<td>ARQ</td>
<td>Automatic Repeat Request</td>
</tr>
<tr>
<td>ARP</td>
<td>Allocation and Retention Priority</td>
</tr>
<tr>
<td>ATB</td>
<td>Adaptive Transmission Bandwidth</td>
</tr>
<tr>
<td>AWGN</td>
<td>Additive White Gaussian Noise</td>
</tr>
<tr>
<td>BE</td>
<td>Best Effort</td>
</tr>
<tr>
<td>BER</td>
<td>Bit Error Rate</td>
</tr>
<tr>
<td>BLER</td>
<td>Block Error Rate</td>
</tr>
<tr>
<td>BSR</td>
<td>Buffer Status Reports</td>
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<tr>
<td>CDF</td>
<td>Cumulative Distribution Function</td>
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<tr>
<td>CDMA</td>
<td>Code Division Multiple Access</td>
</tr>
<tr>
<td>CQI</td>
<td>Channel Quality Indicator</td>
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<tr>
<td>CSI</td>
<td>Channel State Information</td>
</tr>
<tr>
<td>DFT</td>
<td>Discrete Fourier Transform</td>
</tr>
<tr>
<td>DL</td>
<td>Downlink</td>
</tr>
<tr>
<td>eNB</td>
<td>E-UTRAN Node B</td>
</tr>
<tr>
<td>ECR</td>
<td>Effective Code Rate</td>
</tr>
<tr>
<td>EPS</td>
<td>Evolved Packet System</td>
</tr>
<tr>
<td>E-UTRAN</td>
<td>Evolved-UTRAN</td>
</tr>
<tr>
<td>Abbreviation</td>
<td>Description</td>
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<tr>
<td>--------------</td>
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</tr>
<tr>
<td>FFT</td>
<td>Fast Fourier Transform</td>
</tr>
<tr>
<td>FDD</td>
<td>Frequency Division Duplexing</td>
</tr>
<tr>
<td>FDPS</td>
<td>Frequency Domain Packet Scheduling</td>
</tr>
<tr>
<td>FPC</td>
<td>Fractional Power Control</td>
</tr>
<tr>
<td>FTP</td>
<td>File Transfer Protocol</td>
</tr>
<tr>
<td>GBR</td>
<td>Guaranteed Bit Rate</td>
</tr>
<tr>
<td>HARQ</td>
<td>Hybrid-ARQ</td>
</tr>
<tr>
<td>HOL</td>
<td>Head Of Line</td>
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<tr>
<td>HSPA</td>
<td>High Speed Packet Access</td>
</tr>
<tr>
<td>ISI</td>
<td>Inter-Symbol Interference</td>
</tr>
<tr>
<td>ICIC</td>
<td>Inter-Cell Interference Coordination</td>
</tr>
<tr>
<td>KPI</td>
<td>Key Performance Indicators</td>
</tr>
<tr>
<td>LA</td>
<td>Link Adaptation</td>
</tr>
<tr>
<td>LTE</td>
<td>Long Term Evolution</td>
</tr>
<tr>
<td>MAC</td>
<td>Media Access Control</td>
</tr>
<tr>
<td>MBR</td>
<td>Multi-Bit Rate</td>
</tr>
<tr>
<td>MCS</td>
<td>Modulation and Coding Scheme</td>
</tr>
<tr>
<td>MIMO</td>
<td>Multiple Input Multiple Output</td>
</tr>
<tr>
<td>NLOS</td>
<td>Non Line Of Sight</td>
</tr>
<tr>
<td>NaN</td>
<td>Not any Number</td>
</tr>
<tr>
<td>Acronym</td>
<td>Definition</td>
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<td>---------</td>
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<tr>
<td><strong>OFDM</strong></td>
<td>Orthogonal Frequency Division Modulation</td>
</tr>
<tr>
<td><strong>OFDMA</strong></td>
<td>Orthogonal Frequency Division Multiple Access</td>
</tr>
<tr>
<td><strong>PAPR</strong></td>
<td>Peak to Average Power Ratio</td>
</tr>
<tr>
<td><strong>PC</strong></td>
<td>Power Control</td>
</tr>
<tr>
<td><strong>PDU</strong></td>
<td>Protocol Data Unit</td>
</tr>
<tr>
<td><strong>PELR</strong></td>
<td>Packet Error Loss Ratio</td>
</tr>
<tr>
<td><strong>PRB</strong></td>
<td>Physical Resource Block</td>
</tr>
<tr>
<td><strong>PHY</strong></td>
<td>Physical Layer</td>
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<tr>
<td><strong>PUSCH</strong></td>
<td>Physical Uplink Shared Channel</td>
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<tr>
<td><strong>PHR</strong></td>
<td>Power Headroom Report</td>
</tr>
<tr>
<td><strong>QCI</strong></td>
<td>QoS Class Identifier</td>
</tr>
<tr>
<td><strong>QoS</strong></td>
<td>Quality of Service</td>
</tr>
<tr>
<td><strong>RRM</strong></td>
<td>Radio Resource Management</td>
</tr>
<tr>
<td><strong>RB</strong></td>
<td>Resource Block</td>
</tr>
<tr>
<td><strong>RT</strong></td>
<td>Real Time</td>
</tr>
<tr>
<td><strong>SAE</strong></td>
<td>System Architecture Evolution</td>
</tr>
<tr>
<td><strong>SC-FDMA</strong></td>
<td>Single Carrier – Frequency Division Multiple Access</td>
</tr>
<tr>
<td><strong>SINR</strong></td>
<td>Signal to Interference plus Noise Ratio</td>
</tr>
<tr>
<td><strong>SISO</strong></td>
<td>Single Input Single Output</td>
</tr>
<tr>
<td><strong>SNR</strong></td>
<td>Signal to Noise Ratio</td>
</tr>
<tr>
<td>Abbreviation</td>
<td>Full Form</td>
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<td>--------------</td>
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<tr>
<td><strong>SR</strong></td>
<td>Scheduling Request</td>
</tr>
<tr>
<td><strong>SRS</strong></td>
<td>Sounding Reference Symbols</td>
</tr>
<tr>
<td><strong>TDD</strong></td>
<td>Time Division Duplexing</td>
</tr>
<tr>
<td><strong>TTI</strong></td>
<td>Transmission Time Interval</td>
</tr>
<tr>
<td><strong>TPC</strong></td>
<td>Transmitter Power Control</td>
</tr>
<tr>
<td><strong>UE</strong></td>
<td>User Equipment</td>
</tr>
<tr>
<td><strong>UL</strong></td>
<td>Uplink</td>
</tr>
<tr>
<td><strong>UTRAN</strong></td>
<td>Universal Terrestrial Radio Access Network</td>
</tr>
<tr>
<td><strong>VoIP</strong></td>
<td>Voice over Internet Protocol</td>
</tr>
<tr>
<td><strong>WCDMA</strong></td>
<td>Wideband Code Division Multiple Access</td>
</tr>
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</table>
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Amira Mohamed Yehia Abdul Hadi Afifi
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ABSTRACT

The improved performance of the 3GPP Long Term Evolution (LTE) over 3G comes at the cost of increased constraints and challenges for the design. In this thesis a complete radio resource management framework for the LTE uplink is proposed. The Radio Resource Management (RRM) framework fulfills the functionalities of transmission bandwidth allocation, power control and Modulation and Coding Scheme (MCS) assignment in accordance to the LTE specifications for uplink transmission.

The LTE specifies Single Carrier-Frequency Division Multiple Access (SC-FDMA) as the access scheme for the uplink transmission. SC-FDMA is used due to its lower Peak to Average Power Ratio (PAPR) feature, but it comes at the cost of imposing a challenge on the scheduler design that subcarriers assigned to one user must be contiguous.

With the advances in technology a wide range of applications now exist each with its specifications and requirements. For example while VoIP applications require small packet delay, FTP applications can tolerate high packet delays. The transmission bandwidth allocation algorithm considers these requirements and tries to fulfill them. An important issue to consider in power control algorithms is inter-cell interference, although Inter-Cell Interference Coordination (ICIC) may decrease the cell throughput but it eventually maximizes the system throughput. The framework maximizes throughput and spectral efficiency while taking into consideration the users’ different classes of Quality of Service (QoS) as well as performing Inter-Cell Interference Coordination (ICIC).
Chapter 1

Introduction

As the services provided to mobile users become more demanding, the mobile telecommunications systems must evolve to meet these expectations. Third Generation (3G) mobile systems which are based on the WCDMA technology are being deployed to meet the increased demand of higher data rates and QoS differentiation. The Third Generation Partnership Project (3GPP) efforts in standardizing the mobile networks has made it the leading choice for mobile operators and in response to the increased demand for higher performance released the first step in the WCDMA evolution, the High Speed Packet Access (HSPA) system which is classified as 3.5G. In parallel to evolving HSPA, 3GPP is also specifying the Long Term Evolution (LTE), a new radio access technology and network architecture, to stay competitive for a longer time frame by providing considerable performance improvement at a reduced cost. This thesis proposes a radio resource management (RRM) scheme within the LTE framework.

1.1 Thesis Scope and Objectives

To reach LTE's design goals set by 3GPP, LTE's new radio access technology and network architecture must be exploited when addressing the RRM design problem. The RRM functionalities include Admission Control (AC), Packet Scheduling (PS) including Hybrid Automatic Repeat Request (HARQ), and fast Link Adaptation (LA) including Adaptive Modulation and Coding (AMC) and Fractional Power Control (FPC). The RRM design problem can be summarized as providing these functionalities under the constraints introduced by the used technology such as contiguity constraint of SC-FDMA, and the constraints introduced by the design goals such as improving spectral efficiency and QoS provisioning.
This thesis addresses the RRM design problem in the LTE framework focusing on QoS-based PS, AMC and FPC. An investigation of the tradeoff between throughput and inter-cell interference for different SINR Target values is also included. A RRM scheme for the Frequency Division Duplex (FDD) mode is introduced. To assess the performance of the scheme it is simulated with four traffic models each belonging to a different QoS class under the assumption of finite buffer size and SISO antennae setup. Perfect channel knowledge is assumed throughout the study and accordingly HARQ is not considered.

The proposed scheme is evaluated through the following Key Performance Indicators (KPI):

- Average throughput per user: The average per-user data throughput is defined as the sum of the average data throughput of each user in the system divided by the total number of users in the system. The average per-user throughput is also referred to as average or mean user throughput.

- Traffic class packet delay: The delay is defined as the time between the packet arriving to the transmission buffer of a UE and the packet delivered to the physical layer for transmission. The cumulative distribution function (CDF) of the delay is obtained for each QoS class.

- Average Interference Power: The average amount of interference leaked from the cell users to the neighboring cells.

### 1.2 Contribution

When tackling the RRM problem previous work focused on one aspect only of the design. The work would either focus on Adaptive Transmission Bandwidth (ATB) only, AMC only, Power Control (PC) or QoS. Some authors combined two of the RRM functionalities in their work. The main contribution
of this thesis is providing a QoS-based RRM scheme that combines the ATB, LA and PC functionalities of RRM.

The proposed RRM scheme takes into consideration the QoS requirements for the different applications. [1] → [6] studied packet scheduling with the aim to maximize spectral efficiency while neglecting the QoS requirements, packets are scheduled without regards to the delay and packet loss. While [9] → [12] which considered the QoS requirements, focused on the downlink scheduling. The work in [13] and [14] focused on uplink scheduling with QoS requirements but did not consider maximizing the spectral efficiency.

The proposed RRM scheme also considers the inter-cell interference generated on neighboring cells. It combines RRM and inter-cell interference coordination (ICIC) in one scheme. This leads to maximizing throughput and minimizing inter-cell interference while respecting the QoS requirements and following the LTE power control method.

Most of the work done on RRM was evaluated using infinitely backlogged buffer traffic model or simple traffic models that generate packets according to a Bernoulli process. The scheme presented in this thesis is evaluated using realistic traffic models for four different applications: VoIP, Interactive Gaming, Video and FTP.

1.3 Thesis Outline

The thesis is organized as follows

Chapter 2 gives an overview of the 3GPP LTE standard and the SAE architecture. The frame structure and different signaling elements required to design the RRM scheme are presented. The uplink power control as defined by the standard is presented as well. The different QoS classes as specified by the 3GPP for LTE are described. Theoretical background on wireless
communication fundamentals is also given.

Chapter 3 describes the RRM design problem in LTE and provides a survey of previous work addressing the different issues discussed in this thesis. The resource allocation problem is studied from the frequency domain or channel dependant scheduling point of view. Literature addressing the QoS requirements and constraints is also reviewed, and finally studies done for the uplink power allocation and interference coordination are summarized.

Chapter 4 describes the proposed RRM scheme. The scheme performs three functionalities: PS, AMC and FPC. The scheme is also used to study the uplink closed loop power control problem with an emphasis on the SINR Target value selection.

Chapter 5 presents the simulation setup and results. An analysis of the results and comparison with work from the literature is then given. The different KPIs’ are evaluated and presented.

Finally, Chapter 6 concludes the thesis and presents some future work to be done.
Chapter 2
Overview of the 3GPP LTE Standard

LTE is the standard defined by 3GPP for radio access. It has two modes of operation Frequency Division Duplex (FDD) and Time Division Duplex (TDD). Due to the difference in capabilities between the mobile stations and the eNBs, the standard differentiates between Uplink (UL) transmission and Downlink (DL) transmission. Since the scope of this thesis is FDD UL resource management, we will only focus on these parts in the standard.

2.1 LTE Physical Layer

2.1.1 Transmission Scheme

The LTE standard adopts OFDM as the underlying technology for the transmission schemes with a difference in the multiplexing technology chosen for the downlink from that chosen for the uplink. OFDMA has been chosen for the downlink as the multiple access scheme. For the uplink SC-FDMA or DFT-Spread OFDM was chosen due to the difference in the capabilities between the UE and the eNB. The transmitter and receiver for OFDMA and SC-FDMA is shown in Figure 2-1. For the UE the power requirements play a big role in the design and implementation of the standard. SC-FDMA has been chosen due to its lower PAPR compared to multi-carrier transmissions which allow for more efficient use of the power amplifier as well as decreasing the complexity of the equalizer.

SC-FDMA has two types of sub-carrier mapping: (1) Interleaved and (2) Localized. In I-FDMA users are assigned subcarriers that are distributed over the entire bandwidth while in L-FDMA users are assigned consecutive or
contiguous subcarriers. The LTE standard uses the L-FDMA as the sub-carrier mapping for the uplink SC-FDMA transmission to exploit the frequency selective gain offered.

**SC-FDMA**

\[
\{x_s\} \rightarrow \text{N-point DFT} \rightarrow \text{Subcarrier Mapping} \rightarrow \text{M-point IDFT} \rightarrow \text{Add CP} / \text{PS} \rightarrow \text{DAC} / \text{RF} \rightarrow \text{Channel} \rightarrow \text{Detected} \rightarrow \text{N-point IDFT} \rightarrow \text{Subcarrier De-mapping/Equalization} \rightarrow \text{M-point DFT} \rightarrow \text{Remove CP} \rightarrow \text{RF} / \text{ADC}
\]

**OFDMA**

\[
\{x_s\} \rightarrow \text{Subcarrier Mapping} \rightarrow \text{M-point IDFT} \rightarrow \text{Add CP} / \text{PS} \rightarrow \text{DAC} / \text{RF} \rightarrow \text{Channel} \rightarrow \text{Detected} \rightarrow \text{Subcarrier De-mapping/Equalization} \rightarrow \text{M-point DFT} \rightarrow \text{Remove CP} \rightarrow \text{RF} / \text{ADC}
\]

*CP: Cyclic Prefix, PS: Pulse Shaping

Figure 2-1 : Transmitter and Receiver for SC-FDMA and OFDMA[20]
2.1.2 Generic Frame Structure

The FDD frame structure is the same for both UL and DL and is shown in Figure 2-2. The frame length is $T_{\text{frame}}=10\text{msec}$. The frame consists of 10 subframes each is 1 msec. The Transmission Time Interval is $\text{TTI} = 1$ subframe. One subframe is made up of 2 slots $T_{\text{slot}}=0.5\text{msec}$. Each slot in the time domain consists of 6 or 7 OFDM symbols long depending on whether short or long cyclic prefix is used respectively. In the frequency domain the Physical Resource Block (PRB) is 12 consecutive subcarriers during one slot. The smallest schedulable entity is the PRB which consists of 12 subcarriers in the frequency domain and one slot in the time domain.
The scheduling in the uplink is done on a TTI basis, therefore assignment is carried out in terms of consecutive pairs of PRBs.

2.1.3 Reference Signals

To enable coherent demodulation at the eNB reference signals for channel estimation are transmitted in the uplink. Uplink reference signals are time multiplexed with uplink data and therefore are transmitted with the same transmission bandwidth as the data. As these signals are only transmitted in the bandwidth assigned to the user they cannot be used to estimate the channel for the user on the other frequencies not assigned to him. To perform channel dependant scheduling the eNB need to estimate the channel quality for the user on the whole bandwidth and to that aim channel-sounding reference signals are also transmitted. The difference between the channel sounding and demodulation reference signals are:

1) Channel-sounding reference signals have much larger transmission bandwidth than demodulation reference signals.

2) Channel-sounding reference signals are transmitted less often than demodulation reference signals.

3) Channel sounding reference signals from more than one user can be transmitted within the same frequency band. This is achieved by sharing the resource in the time domain, frequency domain or using different cyclic shifts.

The blocks assigned for channel sounding reference signals are not available for uplink data transmission.
2.2 Link Adaptation

Link adaptation is the ability of the system to match the data rate of the users to the channel conditions for a given transmission power. This is achieved by Adaptive Modulation and Coding. (AMC) provides two degrees of freedom, the choice of the modulation scheme and the choice of the coding rate. Higher order modulation schemes provide high data rates but are more prone to errors as they are easily affected by variations in interference, noise and errors in channel estimations. Therefore they should be chosen for high SINR values. The code rate can be adapted such that a low code rate is chosen for bad channels.

The LTE standard supports AMC by providing different modulation and coding rates for the different SINR ranges. Table 2-1 is accessed with the Channel Quality Indicator (CQI) value and the corresponding modulation and Effective Code Rate (ECR) is achieved. The CQI is mapped from the SINR value which is calculated from the channel estimation. In the uplink the eNB estimates the channels from the sounding reference signals.

In the design of the AMC scheme for the LTE one of the issues was whether to assign the MCS on a per PRB basis or on an assignment basis that is all PRBs assigned to one user should have the same MCS. Since it was found that assigning the MCS on a PRB basis only improved the performance slightly and that the signaling overhead was much higher, the LTE standard specified that the MCS will be assigned on an assignment basis.

Another issue is the frequency of the AMC decision. Whether the MCS should be selected every TTI or once every period of time is left to the vendor implementation.
Table 2-1: CQI-MCS Table

<table>
<thead>
<tr>
<th>CQI</th>
<th>Modulation</th>
<th>Efficiency</th>
<th>Coding Rate x1024</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>Out of range</td>
<td></td>
<td></td>
</tr>
<tr>
<td>1</td>
<td>4QAM</td>
<td>0.0762</td>
<td>78</td>
</tr>
<tr>
<td>2</td>
<td>4QAM</td>
<td>0.1172</td>
<td>120</td>
</tr>
<tr>
<td>3</td>
<td>4QAM</td>
<td>0.1885</td>
<td>193</td>
</tr>
<tr>
<td>4</td>
<td>4QAM</td>
<td>0.3008</td>
<td>308</td>
</tr>
<tr>
<td>5</td>
<td>4QAM</td>
<td>0.4385</td>
<td>449</td>
</tr>
<tr>
<td>6</td>
<td>4QAM</td>
<td>0.5879</td>
<td>602</td>
</tr>
<tr>
<td>7</td>
<td>16QAM</td>
<td>0.3691</td>
<td>378</td>
</tr>
<tr>
<td>8</td>
<td>16QAM</td>
<td>0.4785</td>
<td>490</td>
</tr>
<tr>
<td>9</td>
<td>16QAM</td>
<td>0.6016</td>
<td>616</td>
</tr>
<tr>
<td>10</td>
<td>64QAM</td>
<td>0.4551</td>
<td>466</td>
</tr>
<tr>
<td>11</td>
<td>64QAM</td>
<td>0.5537</td>
<td>567</td>
</tr>
<tr>
<td>12</td>
<td>64QAM</td>
<td>0.6504</td>
<td>666</td>
</tr>
<tr>
<td>13</td>
<td>64QAM</td>
<td>0.7539</td>
<td>772</td>
</tr>
<tr>
<td>14</td>
<td>64QAM</td>
<td>0.8525</td>
<td>873</td>
</tr>
<tr>
<td>15</td>
<td>64QAM</td>
<td>0.9258</td>
<td>948</td>
</tr>
</tbody>
</table>

2.3 Uplink Power Control

Power control in the uplink is a crucial part of the design and implementation due to the limited UE capabilities and the stringent QoS requirements. A tradeoff between maximizing the UE’s throughput and minimizing the inter-cell interference needs to be established.

The LTE standard defines setting the UE’s transmit power on the PUSCH through equation (2-1) Error! Reference source not found.

\[
P_{\text{PUSCH}} = \min\{P_{\text{max}}, 10\log_{10}M + P_o + \alpha PL + \Delta TF + f(\delta_{\text{PUSCH}})\}[dBm] \quad (2-1)
\]

where

- \(P_{\text{max}}\) is the maximum allowed transmit power that depends on the
UE power class

- $M$ is the number of PRBs assigned to the user in this subframe
- $P_0$ is a parameter with a cell specific nominal component and a UE specific component
- $\alpha$ referred to as the path loss compensation factor is a 3-bit cell specific parameter with values $\{0, 0.4, 0.5, 0.6, 0.7, 0.8, 0.9, 1\}$. $\alpha = 0$ indicates no path loss compensation i.e the power is adjusted without regards to the user’s pathloss, while $\alpha = 1$ indicates full compensation of the path loss. Values between 0 and 1 indicate partial compensation of the path loss.
- $PL$ is the downlink path loss estimate
- $\Delta_{TF}$ is a cell specific parameter function in the Transport Format (MCS)
- $\delta_{PUSCH}$ is a UE specific value referred to as a Transmitter Power Control (TPC) command or closed loop correction value. $f(*)$ can represent an accumulation function or a current absolute value, the function type is a UE specific parameter.

The equation without the parameter $f(\delta_{PUSCH})$ represents open loop power control, which means that the UE adjusts its power without intervention from the eNB. The addition of this parameter converts the equation to closed loop power control where the eNB can adjust the UE’s power through this parameter.

For further information regarding the power control parameters refer to the LTE standard [26].

The choice of these parameters affects the power control scheme. Power control schemes can be categorized based on the value of $\alpha$. For $\alpha=1$ this will lead to full compensation of the path loss, where the transmission power is increased to compensate for the path loss and the higher the path loss the higher the transmission power, giving a conventional power control scheme.
\( \alpha = 0 \) will lead to no compensation of the path loss and therefore no power control. \( 0 < \alpha < 1 \) leads to fractional power control schemes where the path loss is only partially compensated for.

Another categorization for the power control scheme is whether it is open loop or closed loop. In open loop power control scheme the UE set its uplink transmit power based on measurements and estimations done at the UE. In closed loop power control schemes the eNB can control the uplink transmit power of the UE. In equation (2-1) the parameters \( P_o, \alpha, \) and \( \Delta_{TP} \) are cell specific values signaled to the UE by the eNB once. The path loss \( PL \) can be estimated at the UE and the number of PRBs \( M \) is known at the UE. From this it can be seen that the first part of equation (2-1) \( (10\log_{10}M + P_o + \alpha PL + \Delta_{TP}) \) which can be considered as an open loop power control, where the UE can adjust its power without further signaling from the eNB. The addition of \( f(\delta_{PUSCH}) \) will make the power control scheme a closed loop one, since the eNB will signal this parameter to adjust the power of the UE. As can be seen from the uplink transmit power equation, the closed loop power control is based on the open loop point of operation, that is the eNB only send correction values to the open loop power.

### 2.4 QoS and EPS bearers

Nowadays mobile station applications are not limited to voice. This wide range of applications introduced different levels of requirements leading to different QoS classes definitions. While some application could tolerate relatively high packet delay they could not tolerate high packet loss ratio. To address the different requirements for the applications different EPS bearers are established each with a different set of QoS requirements.

The first categorization for the EPS bearers depends on the nature of the QoS they offer.
- Minimum Guaranteed Bit Rate (GBR) bearers: these have dedicated transmission resources allocated for the bearer which is usually done in the admission control functionality of the RRM.

- Non-GBR bearers: these do not guarantee a bit rate and therefore no resources are permanently allocated to the bearer.

Each EPS bearer is also defined with a QoS Class Identifier (QCI) and an Allocation and Retention Priority (ARP). The parameters associated with each QCI have been standardized so that uniform traffic handling can be expected regardless of the eNB vendors’ implementation as shown in Table 2-2. The ARP parameter is used only during bearer establishment i.e. during admission control only, it governs whether or not the bearer could be accepted and the priority of the bearer in relation to a new bearer that would be established. Scheduling decisions are governed by the parameters associated with the bearer’s QCI. All packets forwarded on the same EPS bearer receive the same QoS level, therefore to provide a different QoS level a separate EPS bearer must be established.

**Table 2-2 : Standardized QCI for LTE [27]**

<table>
<thead>
<tr>
<th>QCI</th>
<th>Resource Type</th>
<th>Priority</th>
<th>Packet delay (ms)</th>
<th>Packet error loss rate</th>
<th>Example Services</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>GBR</td>
<td>2</td>
<td>100</td>
<td>$10^{-2}$</td>
<td>Conversational Voice</td>
</tr>
<tr>
<td>2</td>
<td>GBR</td>
<td>4</td>
<td>150</td>
<td>$10^{-3}$</td>
<td>Conversational video (live streaming)</td>
</tr>
<tr>
<td>3</td>
<td>GBR</td>
<td>5</td>
<td>300</td>
<td>$10^{-6}$</td>
<td>Non-conversational video (buffered streaming)</td>
</tr>
<tr>
<td>4</td>
<td>GBR</td>
<td>3</td>
<td>50</td>
<td>$10^{-3}$</td>
<td>Real Time Gaming</td>
</tr>
<tr>
<td>5</td>
<td>Non-GBR</td>
<td>1</td>
<td>100</td>
<td>$10^{-6}$</td>
<td>IMS Signaling</td>
</tr>
<tr>
<td>QCI</td>
<td>Resource Type</td>
<td>Priority</td>
<td>Packet delay (ms)</td>
<td>Packet error loss rate</td>
<td>Example Services</td>
</tr>
<tr>
<td>-----</td>
<td>---------------</td>
<td>----------</td>
<td>-------------------</td>
<td>------------------------</td>
<td>------------------</td>
</tr>
<tr>
<td>6</td>
<td>Non-GBR</td>
<td>7</td>
<td>100</td>
<td>$10^{-3}$</td>
<td>Voice, video (live streaming), interactive gaming</td>
</tr>
<tr>
<td>7</td>
<td>Non-GBR</td>
<td>6</td>
<td>300</td>
<td>$10^{-6}$</td>
<td>Video (buffered streaming)</td>
</tr>
<tr>
<td>8</td>
<td>Non-GBR</td>
<td>8</td>
<td>300</td>
<td>$10^{-6}$</td>
<td>TCP-based (e.g., WWW, e-mail) chat, FTP, p2p file sharing, progressive video, etc.</td>
</tr>
<tr>
<td>9</td>
<td>Non-GBR</td>
<td>9</td>
<td>300</td>
<td>$10^{-6}$</td>
<td></td>
</tr>
</tbody>
</table>


2.5 Reports Sent by UE

2.5.1 Buffer Status Reports

To allocate resources efficiently the uplink scheduler at the eNB needs to know the requirements of the different UEs in terms of pending data. The UE informs the eNB of the amount of data in its buffer by means of Buffer Status Reports (BSR).

EPS bearers are mapped to radio bearers and the radio bearers are mapped to logical channels. BSR’s are reported on logical Channels.

There are two types of BSRs. A Short BSR which reports the amount of data available in one logical channel or logical channel group. The Long BSR reports the amount of data in the buffers of four logical channels or logical channel groups. The choice of which type of BSR to transmit depends on how many logical channels have data in their buffers or the amount of available uplink resources.

When a BSR is triggered and there are not enough resources to transmit it, a 1 bit Scheduling Request is transmitted or the Random Access procedure is performed to request uplink resources to transmit the BSR.

2.5.2 Power Headroom Reports

Power Headroom reports are sent by the UE to the eNB to inform the latter of the amount of available power at the UE. Each UE has a maximum uplink power transmission limit that it cannot exceed; therefore assigning resources to a UE which has reached its maximum power limit will be a waste of resources as the UE will not be able to benefit from the allocation. The eNB can then use the power headroom report to determine how much more bandwidth it can allocate to a UE or how much extra bandwidth was previously
allocated.

The eNB can also use the power headroom reports to calculate the path loss of the users from it using the $P_{\text{PUSCH}}$ power control equation as shown in equation (2-2)

$$P_H = P_{\text{max}} - P_{\text{PUSCH}} \text{[dBm]} \quad (2-2)$$

where $P_H$ is the Power Headroom

- $P_{\text{max}}$ is the maximum allowed uplink transmission power
- $P_{\text{PUSCH}}$ is the uplink transmission power
Chapter 3

Problem Description and Overview of Related Work

3.1 Problem Description

The 3GPP LTE standard was put based on a set of requirements and design targets. To achieve these targets the LTE used new technologies such as multi-carrier, multiple antennae and packet switched radio interface. As these technologies help improve system performance they impose new constraints and challenges for the design and implementation problem.

To exploit the new dimensions of frequency and space introduced by OFDM and MIMO respectively, cross layer techniques between the physical layer and the link layers should be implemented. Some of these techniques which exploit the frequency dimension include channel dependant scheduling, dynamic link adaptation and power control. It is then noted that optimizing the RRM functionalities depend on the physical layer configurations.

In addition to the constraints and requirements introduced by the standard specifications and the technologies used, the RRM design must consider the requirements and constraints of the different types of applications or what is known as providing the appropriate QoS to the appropriate traffic type. Proportional to the importance of this issue is its complexity as more diverse applications with different requirements are introduced.
Figure 3-1: The RRM Design Problem Visualization
3.1.1 RRM Functionalities

As radio resources are scarce and at the same time the number of users and their demands increase, managing the radio resources becomes a crucial point in the performance of any wireless network. The two main resources that need management are the transmission bandwidth and uplink power. The RRM functionalities are:

- **Admission Control:** This function ensures that a new user is not allowed to join the network unless there are enough resources to meet its QoS requirements. This function is performed only during EPS bearer establishment.

- **Scheduling transmission bandwidth:** The number of users is larger than the available bandwidth. The scheduler function is to determine each TTI, which users to be served, the transmission bandwidth allocated to each user and the location of this bandwidth in the spectrum.

- **Link Adaptation:** The link adaptation determines the transport format of the packets for each user, i.e. it assigns a modulation and coding scheme. It also performs power control to have the users operate at a target SINR.

3.1.2 RRM Constraints and Challenges

The implementation of the above RRM functionalities must meet the requirements of the system and users while respecting the constraints of the technologies.

The constraints on the scheduler can be summarized as (1) the number of available PRBs, (2) for each PRB at most one user can be assigned. For the link adaptation functionality the constraints are (1) the available
modulation schemes, (2) the available code rates, (3) the maximum transmission power.

SC-FDMA was chosen as the uplink transmission scheme for LTE. This adds a constraint on the scheduler functionality that the PRBs allocated to one user must be contiguous in frequency.

![Contiguous Assignment of PRBs in SC-FDMA](image)

Figure 3-2: Contiguous Assignment of PRBs in SC-FDMA

The main requirements the RRM functionalities must meet are those related to the QoS parameters. The RRM functionalities should provide each user with the QoS the connection was established based on. This includes meeting the guaranteed bit rate, maximum tolerated packet delay and maximum tolerated packet loss ratio.

Besides the constraints and requirements the RRM functionalities must address the issues of system performance. Design of RRM algorithms should consider issues such as maximizing spectral efficiency and overall system
throughput, minimizing inter-cell interference and providing fairness among users.

### 3.1.3 Inputs to the RRM

To serve its purpose the RRM is provided with inputs to help it take informed decisions.

The channel conditions which are required for the link adaptation and scheduling can be estimated for the whole bandwidth for each user with the help of the channel sounding reference signals. The SINR is calculated for the SRS’s and then mapped to a CQI. For the downlink the UE sends CSI reports to the eNB.

To distinguish active users with data in their buffers from idle users, each user with data pending in its buffers sends a one bit Scheduling Request (SR) to the eNB informing it with the need for UL grants.

The information about the amount of data available at each user’s buffers is collected by means of buffer status reports.

The UE sends Power Headroom Reports to the eNB to inform it about the amount of available power the UE has which is used in the power control and scheduling functionalities of the RRM. The PHR can also be used to estimate the downlink path loss.
Figure 3-3: UE-eNB Signalling
3.2 Related Work

3.2.1 Channel Dependant Scheduling

Frequency Domain Packet Scheduling also known as Channel Dependant Scheduling exploits the multiuser diversity to increase spectral efficiency and throughput. The idea behind channel dependant scheduling is that not all users see the channels in the same way. A channel which one user has very low gain on, can have very high gain for another user as seen in Figure 3-4 [21]. The scheduling decision is the assignment of resource blocks to users in such a way as to maximize a utility function based on throughput. The channels for each user are obtained by providing the SRS to a channel estimation algorithm. To simplify the problem and focus on the scheduling part of it, perfect channel knowledge is assumed in the majority of the literature addressing this problem.

![Channel Dependant Scheduling](image_url)

Figure 3-4 : Channel Dependant Scheduling [21]

3.2.1.1 Heuristic Algorithms for Channel Dependant Scheduling

The following work tackled the channel dependant scheduling from a heuristic point of view to decrease computational complexity and provide a
cost efficient implementation.

S. Lee et al. in [1] evolved the conventional Round Robin algorithm and introduced four algorithms to address the problem of uplink scheduling in the LTE.

- **Algorithm 1**: Carrier by Carrier in turn. Round robin on the RBs starting from RB number 1, the RB is assigned to the user with the highest Proportional Fair (PF) metric \( \lambda^c_i(t) = \frac{r^c_i(t)}{R_i(t)} \), on the RB without violating the contiguity constraint, where \( r \) is the instantaneous channel rate for user \( i \) on RB \( c \) at time \( t \), and \( R \) is the long term average rate of user \( i \).

- **Algorithm 2**: Largest Metric Value RB First. Same as algorithm 1 but instead of starting with RB number 1 start with the RB with the highest metric value, if 2 non-adjacent RBs are assigned to the same user, the RBs in-between these 2 assigned RBs are also assigned to the same user.

- **Algorithm 3**: Riding Peaks. The idea is based on that there is correlation between adjacent RBs i.e. if for a user a certain RB has high metric it is with high probability that the neighboring RBs will also have high metrics. Same as algorithm 2 look at the highest metric value RB first, the RB is assigned to user \( i \) if the RB is adjacent to the previously assigned RBs of user \( i \) or user \( i \) has no RBs assigned yet. Otherwise it is not allocated and we move on to the next RB.

- **Algorithm 4**: RB Grouping. Algorithm 3 is expected to yield good results if the frequency domain exhibits high correlation between two adjacent RBs. However correlation in the frequency
domain is not strong which implies that even though we have an overall frequency correlation the granularity of this correlation is not necessarily as small as one RB. To deal with this issue of small scale variations the unit of consideration will be RB group. A RB group is a number of contiguous RBs viewed as one block together.

In [2] a Search-Tree Based packet scheduling algorithm is proposed. The idea is to draw a binary tree where each node has 2 branches the maximum utility and the second maximum utility. Allocation is decided at the end where the path with the highest overall global utility is chosen. In the simulations the bandwidth is assumed equal and fixed for all UEs therefore neglecting the adaptive transmission bandwidth problem, imperfect channel knowledge is considered through considering HARQ retransmissions, and open loop power control is implemented.

In [3] three heuristic algorithms were proposed. First Maximum Expansion (FME) algorithm assigns resources starting from the highest metric value and expanding the allocation on both sides, the allocation for a UE is stopped whenever another UE with a higher metric is found. The second algorithm Recursive Maximum Expansion (RME) is similar to the first one except that it performs a recursive search of the maximum. Figure 3-5 shows a comparison between the allocation done using RME and FME. Minimum Area Difference (MAD) to the Envelope algorithm’s scope is to derive the resource allocation that provides the minimum difference between its cumulative metric and the envelope-metric, i.e., the envelope of the users’ metrics. An example of how the resources are assigned is shown in Figure 3-6. Although MAD is a more computationally expensive algorithm than the previous two however it gives closer results to the optimal solution. This algorithm aims at minimizing the difference between its cumulative metric and the envelope metric.
Improvements over Round Robin algorithm have been shown.

Figure 3-5: Example of resource allocation by RME and comparison with FME [3]
Figure 3-6 : Example of resource allocation by MAD [3]
In [4] the authors combine the work of [2] and [3] and propose two improvements for the Recursive Maximum Expansion Algorithm. The first algorithm gives higher flexibility in the RB expansion by not limiting the expansion only to the maximum but having it to be within a threshold, while the second algorithm Improved Tree Based Recursive Maximum Expansion gives even more flexibility in the expansion by considering the highest and second highest metric deriving a binary search tree. Results show that with a little higher complexity around 15% gain in spectral efficiency is obtained compared with conventional Recursive Maximum Expansion.

The authors in [5] looked at the problem from a different view by focusing on investigating different scheduling metrics, mainly a combination of two metrics throughput based in the time domain and SINR based in the frequency domain. Scheduling metrics affect the priority of the users. Evaluations are based on a RRM framework with open loop power control and AMC.

A Heuristic Localized Gradient Algorithm (HLGA) is discussed in [6]. The algorithm simplifies the optimal Localized Gradient Algorithm (LGA) by following a simple heuristic in allocating the RBs to the users while maintaining the required allocation constraint and taking retransmission requests into consideration. The basic idea is to search for the RB-user pair with the highest metric value and assigning this RB to this user, if the user has previous allocations that is not contiguous to the new allocation, all in-between RBs are assigned to the user. The HLGA is compared to a reference algorithm LGA which defines and solves an optimization problem. In [7] the authors examined the performance of the HLGA when dynamic traffic models are assumed. To solve the problem of different buffer occupancies and therefore different users' requirements, a procedure called 'pruning' is introduced. The pruning procedure states that RBs are first assigned to users according to the
HLGA regardless of their traffic requirement. Extra RBs allocated to users are then reallocated to either neighboring users in the spectrum or users with no assignments. Pruning is performed on edge RBs to preserve the contiguity constraint.

3.2.1.2 Optimization Problem of Channel Dependant Scheduling

The channel dependant scheduling problem was addressed as an optimization problem to obtain an optimal solution. As these solutions usually need high computation resources they are mostly used as a benchmark and reference for other suboptimal or heuristic solutions. In [6] and [8], the authors discussed the optimization problem and provided suboptimal algorithms to solve it.

In [6] the gradient metric was chosen as the scheduling metric for the scheme. The utility function is a strictly concave smooth utility function. An integer programming assignment problem is formulated with a contiguity constraint. The integer programming optimal solution assumes perfect channel knowledge, in case of measurement delays and estimation errors, the selection rule occasionally picks rates that do not match the channel capacity. Synchronous non adaptive HARQ is assumed to take care of these errors and therefore a constraint is added to allocate to the users having HARQ processes operating on certain RBs those same RBs until successful transmission is achieved. In this case the solution provided is not the optimal one.

In [8] a general LTE UL Frequency Domain Packet Scheduling (FDPS) problem is formulated. The problem defines a profit function which is a general term used to represent various scheduling policies. The aim is to schedule the RBs in a time slot such that the profit is maximized. The LTE UL FDPS problem is shown to be NP-hard and therefore two approximation algorithms are introduced which computes solutions close to the optimum. The first
heuristic algorithm is a greedy strategy based algorithm which divides the LTE UL FDPS problem into several sub problems according to the profit and then apply a greedy method to each problem. The second algorithm is based on a local ratio technique.

The aforementioned algorithms have all assumed full buffer traffic model, fixed traffic classes and only focusing on increasing overall throughput with no regards to interference on other cells.

### 3.2.2 QoS Oriented Scheduling

QoS oriented scheduling considers in addition to increasing throughput, per-flow metrics such as packet loss and packet delay. In today’s wireless networks, more diverse applications, such as video and interactive gaming, are introduced leading to different QoS requirements. The scheduling scheme should consider these variations when assigning resources. QoS parameters including maximum allowed packet delay, maximum allowed packet drop ratio, priority and guaranteed bit rate GBR must be respected. In the channel dependant scheduling problem, the utility function is usually a function of the overall rate is to be maximized under the contiguity constraint. When QoS classes are accounted for in the resource assignment, this will add further constraints on the scheduler.

**3.2.2.1 Downlink QoS Oriented Scheduling**

The following papers discussed QoS-oriented scheduling but from the downlink point of view. Downlink scheduling is a simpler problem than the uplink as the contiguity constraint is removed. Another issue that is different in the downlink from the uplink is that the HOL packet delay in the downlink is available at the eNB while in the uplink it is either estimated or the buffer status is used instead.

The authors in [9] expanded the channel dependant scheduling
problem to include buffer status awareness. AMC is also implemented where, the MCS is chosen depending on the CQI reported by the UE. Proportional fairness is achieved by preferring users with good channel conditions to be scheduled but at the same time if the user is scheduled frequently the priority of the user is decreased. A priority function is proposed to achieve four objectives 1) maximize system throughput 2) decrease packet drop ratio 3) keep certain fairness among users 4) guarantee QoS of multimedia services. The priority is a function of the buffer status, the instantaneous data rate, the average data rate, priority of the service class and GBR for RT services. For each subband the priority function is calculated for all the users and the subband is allocated to the user with the highest priority on this subband.

In [10] a decoupled time and frequency domain scheduler is proposed. The time domain scheduler helps in reducing the complexity of the frequency domain scheduler by minimizing the set of schedulable users. Users are added to the set of schedulable users if they satisfy that they have data to be transmitted and that the data has passed either the buffer amount threshold or the head of line threshold. The schedulable users are then prioritized according to a set target bit rate. The frequency domain scheduler then selects for PRB $k$ the user $n$ that maximizes a chosen metric. Three metrics are investigated

- Proportional Fair:
  \[ M_{PF} = \frac{\hat{d}[k,n]}{R[n]} \]
  where $R[n]$ is the past average throughput of user $n$
  $\hat{d}[k,n]$ is the estimated achievable throughput for user $n$ on PRB $k$

- Proportional Fair Scheduled:
  \[ M_{PFsch} = \frac{\hat{d}[k,n]}{R_{sch}[n]} \]
  where $R_{sch}[n]$ is an estimate of the user throughput if user $n$ was scheduled every sub-frame.
• Carrier Over Interference to Average.

\[ M_{ColtA} = \frac{\hat{C}oI[k,n]}{\sum_{k=1}^{N_{PRB}} \hat{C}oI[k,n]} \]

where \( \hat{C}oI[k,n] \) is an estimation of the SINR on the \( k^{th} \) PRB of the \( n^{th} \) user. The denominator describes the average channel quality of the user.

In [11] the authors propose a delay prioritized scheduling algorithm consisting of three steps. Step 1 the algorithm computes for each user the remaining time of the HOL packet delay to approach the delay threshold. Step 2 the user with the lowest difference between the HOL delay and the threshold is chosen. Step 3 finds the RB with the highest gain for the selected user and assigns the RB to it.

In [12] a similar approach is adopted where the users are classified into three categories according to their HOL packet delay and then the packet scheduler allocates RBs to corresponding users according to their priorities.

3.2.2.2 Uplink QoS Oriented Scheduling

The QoS oriented scheduling was discussed in the uplink in the following papers.

In [13] two algorithms are proposed which consider the end to end packet delay constraint in addition to the contiguity constraint imposed in LTE uplink scheduling. The first algorithm goes through each RB and assigns it to a user taking into account the contiguity constraint if the maximum delay and minimum throughput requirements are satisfied for all the users. Otherwise it assigns first the users with critical delay or throughput as long as RBs are available and do not violate the contiguity constraint. The second algorithm differs from the first one in the use of a different metric value and that RBs are not assigned in order but with respect to users with critical delay requirement. Both algorithms tend to assign one block per user and therefore are not
efficient in the case when the number of users is smaller than the number of RBs, also in both algorithms some users may never be assigned RBs.

In [14] the constraint of the queuing delay is studied under the goal to minimize power transmission. An optimization problem is formulated and a suboptimal algorithm is proposed and evaluated.

3.2.3 Uplink Power Control and Interference Coordination

The problem of power control is closely associated with inter-cell interference. An optimum transmission power is the power that can maximize system throughput. System throughput is a function of the cell throughput and the inter-cell interference. The higher the transmission power the higher the cell throughput but also the higher the inter-cell interference and subsequently the inter-cell interference decrease the throughput for the neighboring cells. So basically power control objective is to achieve a transmission power that maximizes user and cell throughput but at a reasonable inter-cell interference level. The argument for this concept is that if all eNB’s control the level of interference they cause to neighboring cells, then each individual eNB will also receive limited interference from its neighbors resulting in an overall good performance. This behavior is desirable as it is in synchrony with the concept of self-organization (SoN) that is being promoted by 3GPP and other forums to enhance the configuration, optimization, and operations of future wireless networks[22][23].

The LTE uplink power control mechanism constitutes of a closed loop component operating around an open loop point of operation. The open loop component is derived based on a parameterized power control scheme. Conventional power control schemes assume full compensation for the path loss where all users are received with the same SINR, which means that the higher the path loss the higher the transmission power. Since the path loss to
the serving eNB is inversely proportional to the path loss to the neighboring eNB’s, the conventional power control scheme with full compensation could lead to high inter-cell interference levels. The uplink power equation as stated in the 3GPP LTE standard allows the use of Fractional Power Control (FPC) where users with higher path loss operate at a lower SINR requirement. To further improve the performance in terms of interference a closed loop component is added to the open loop. The closed loop component is left to the implementation of the vendor where usually it is associated with a SINR or interference target.

While the open loop term determines the tradeoff between cell capacity and cell outage by compensating for slow variations, the closed loop term affect the long term performance tradeoff between inter-cell interference and system throughput by compensating for fast variations.

3.2.3.1 Performance Evaluation of the LTE Uplink Power Control mechanism

In [15] and [16] the authors have evaluated the performance of the uplink power control mechanism defined in the LTE standard.

The authors in [15] evaluate the FPC equation and study the effect of the different parameters on the interference levels and the SINR which gives an indication on the throughput. The paper’s focus was only on the open loop power control therefore depending on the aim of the PC scheme whether it is to increase capacity or to improve outage performance, a choice for the different parameters of the open loop power control equation was made to achieve that.

While authors in [15] focused on the open loop performance, authors in [16] evaluated the performance of both the open loop and closed loop power control mechanisms. Investigations of the different parameter sets to give (a) high bit rate and (b) high capacity have been done.
3.2.3.2 Uplink Power Control and ICIC

The following work studied the uplink power control problem and proposed power control techniques but neglected the effect of some RRM functionalities.

In [17] the author designed and implemented a closed loop power control scheme in combination with the fractional path loss compensation factor. An investigation for the different values of the path loss compensation factor has been provided as well as a choice for the optimal value that gives best cell edge and mean user throughput. The SINR Target is set by the eNB as the target SINR that a UE should be using. The SINR Target affects two parameters, the throughput and the interference. A mathematical model for setting the SINR Target based on the path loss is described. The proposed model assigns a higher SINR Target for users with low path loss therefore allowing these users to transmit at a higher power. Implementing this model will decrease the inter-cell interference but at the expense of lower throughput for the cell edge users.

In [18] the author studied the open loop power control and introduced two closed loop power control algorithms to enhance the performance.

- The first algorithm is designed to set the transmission power spectral density according to the tradeoff between the UE’s generated signal and interference powers to achieve a higher SINR for this UE. The algorithm calculates the total interference generated by a UE and compares it to an interference limit. While the interference spectral density limit is equal to all users the interference limit is dependent on the allocated bandwidth to the users, but the author assumes equal bandwidth allocation to all users in the simulation. The transmitting power spectral density is
set using the interference spectral density limit.

- A further modification is introduced by adding a factor similar to the path loss compensation, to tune the path loss of the UE to the neighboring eNBs. The two path loss terms (path loss between UE and serving eNB, and path loss between UE and neighboring eNBs) with their compensation factors are combined in a formula to obtain the transmitting power spectral density.

- The second algorithm is designed to set the transmission power according to a throughput measure. While a user may represent a better tradeoff between signal and interference powers, it is not necessary to represent a better tradeoff between gain in throughput and interference. For each user two parameters are calculated, the gain to the system calculated as the increase in throughput due to the increase in power, and the cost to the system calculated as the increase in interference due to the increase in power. To calculate the increase in throughput an estimate of the interference this user is experiencing is estimated. An “interference budget” of the cell is managed by allowing iteratively one UE to increase its transmission power by one step until the target interference is met. The UE will be chosen based on the gain/cost criteria.

In [19] an uplink radio resource management scheme is proposed with an emphasis on admission control, packet scheduling and handover. The admission control algorithm is based on fractional power control. The admission decision for a new user is based on the availability of enough PRBs to satisfy his QoS requirements. To calculate the number of PRBs required to
fulfill the requested GBR, the open loop fractional power control equation is used. The number of required PRBs of existing users is obtained by using the average number of PRBs allocated to these users by the packet scheduler.

The admission control algorithm is then combined with a packet scheduler. The packet scheduling is done in two phases: Time Domain and Frequency Domain. The time domain scheduler prioritizes the users according to their GBR requirements giving higher priority to the user further below his GBR and then selects $N$ users to input to the frequency domain scheduler. The frequency domain scheduler allocates flexible number of PRBs to the users by using an adaptive transmission bandwidth based scheduling which aims at maximizing the sum of a frequency domain metric. QoS is included in the frequency domain by weighting the frequency domain metric according to the QoS requirements. The combined admission control and packet scheduling is evaluated for different traffic profiles.

The work also discusses handover procedure which is out of scope of this thesis.
3.3 Shortcomings of Previous Work

As seen from the previous chapter the LTE uplink RRM problem has been the interest of many researchers. As the RRM problem is a broad subject and even though many findings and contributions have been made there still room for more contributions and improvements.

Previous work when studying the RRM problem mainly focused on one or two functionalities only. Authors would focus on optimizing the frequency domain scheduler, or the AMC or the power control functionality. As this approach optimizes this functionality the effect of the algorithm on the other functionalities is not studied. Therefore an algorithm optimizing the frequency domain packet scheduling for example may not be the optimum one from the point of view of power control and ICIC.

Most previous work also evaluates performance assuming full buffer model and unified traffic model for all the users. Since this evaluation disregards the QoS requirements, the performance of the algorithms in terms of user satisfaction could not be measured. Therefore while an algorithm could yield high performance metrics in terms of throughput, it can possibly give very low metrics with respect to respecting the QoS requirements.
Chapter 4
The Proposed Radio Resource Management Framework

4.1 General Framework

The proposed RRM scheme selects the users to be served in each TTI from the set of active users and assigns the transmission bandwidth, MCS and power for the selected users.

The scheme consists of three stages

1) Priority Assignment: In this stage a relative priority is assigned to each user. This priority is used in the next two stages to decide whether the user will be served in this TTI or not and how many resources will be allocated to it.

2) Time Domain Scheduling: To decrease the complexity of the frequency domain scheduler so as to search in a limited number of users for the optimum solution, the Time Domain Scheduler selects from the set of active users according to the priority the users to be served in this TTI.

3) Frequency Domain Scheduling: The frequency domain scheduler for each user decides the number and location of PRBs to be assigned, the MCS and the closed loop power adjustment.
4.2 Priority Assignment

Before starting the assignment scheme each user must be given a priority for its requirements. This priority can be considered as a relative priority to the other users. The designed priority function shown in equation (4-1) considers two parameters when assigning the priority:

1) The QCI priority / QoS class

2) The delay of the Head Of Line (HOL) packet at the user’s buffers.

\[ P_i = S(d_i) + \frac{QoS_i}{8} \]  

(4-1)

The first term of the priority assignment function, equation (4-1), is the term dealing with the HOL delay while the second term is the consideration of the QoS class parameter. \( QoS_i \) can take one of four values [1 2 3 4], and the
function $S$ has values less than one, therefore to have the QoS term comparable to the delay term and not to have the QoS dominate the priority, the QoS is divided by 8. $d_i$ is the average delay of the HOL packet for user $i$, and $S(x)$ is a sigmoid function. The sigmoid function was chosen as the function type to only give weight to the HOL delay when it approaches its maximum allowed value, as the HOL delay is still small compared to the maximum allowed the main weight of the priority function will be the QoS class. This way the QoS requirements are respected and at the same time possible starvation of some low QoS class users is avoided. The function $S$ [28] is given by:

$$S(d_i) = \frac{1}{1 + e^{-q_i(d_i - D_{i}^{\text{max}})}} \quad (4-2)$$

where $D_{i}^{\text{max}}$ is the maximum allowed delay for user $i$ and $q_i$ as defined by equation (4-3) [28], is a quantization constant which indicates the emergency of the traffic of user $i$ according to its required PELR.

$$q_i = \frac{\text{Level Number}}{\text{Total Number of Levels on the system}} \quad (4-3)$$

Referring to Table 2-2 we find that the minimum PELR is $10^{-6}$ accordingly we set the Total Number of Levels in the system to be equal 6. The level number is the exponent of the PELR, e.g. for VoIP the maximum allowed PELR is $10^{-2}$ therefore a user with a VoIP traffic shall have a Level number = 2.

The HOL delay is not signaled explicitly to the eNB instead the user only signals the number of bits in its buffers using BSRs, therefore to have a realistic implementation; we deduced an approximation equation for the HOL using BSR.

$$S(b_i) = \frac{1}{1 + e^{-q_i(b_i - B_{i}^{\text{max}})}} \quad (4-4)$$
where $b_i$ is the amount of data in bits available in the buffers of user $i$, and $B_i^{\text{max}}$ is the maximum allowed buffer size given by equation (4-5). If we know the maximum packet delay and the average traffic source rate of the UE, we can estimate the maximum allowed buffer size at the UE as the inverse of the multiplication of the traffic source rate and the maximum packet delay.

\[
B_i^{\text{max}} = \frac{1}{\lambda_i b_i^{\text{max}}} \quad (4-5)
\]

$\lambda_i$ is the average traffic source rate in bits/sec.

**4.3 Time Domain Scheduler**

An important issue facing the implementation of wireless standards for mobile terminals is the complexity of the implementation. Mobile terminals represent an embedded system from which arise design challenges such as memory consumption, power limitations and CPU usage. The design and implementation of algorithms for the mobile terminals must then be optimized to have fast execution time, small power consumption and small memory consumption.

In the proposed scheme the main computational complexity is present in the frequency domain scheduler. The high complexity comes from searching for an optimum or suboptimal assignment decision in a large search space. The higher the number of users the higher the complexity.

The time domain scheduler narrows down the search space for the frequency domain scheduler by passing only a set of schedulable users to the frequency domain scheduler which is a subset from the set of active users.

**4.3.1 Calculating the Maximum Channel Capacity**

First step in the time domain scheduler is obtaining the maximum bit rate that can be sent in one TTI using Shannon’s definition equation (4-6)
for the maximum capacity of a communication channel.

\[ C = B \log_2 \left(1 + \frac{S}{N}\right) \]  \hspace{1cm} (4-6)

where \( C \) is the maximum channel capacity in bits/second, \( B \) is the channel’s bandwidth, \( S \) is the total received signal power and \( N \) is the total noise power. \( S/N \) is also referred to as the signal to noise ratio.

Although there are variations of the SNR from one user to another and even for the same user from one part of the bandwidth to another, to calculate the maximum capacity we assume the best channel conditions. By this we assume that the SNR value is the highest possible which will give the maximum CQI index.

The number of bits that can be transmitted per PRB in one TTI is calculated using equation (4-7),

\[ N = N_{sc} \cdot N_s \cdot N_{slots} \cdot ECR \]  \hspace{1cm} (4-7)

where \( N_{sc} \) is the number of subcarriers per PRB, \( N_s \) is the number of symbols per slot, \( N_{slots} \) is the number of slots per TTI, and ECR is the Effective Code Rate (ECR) which is equal to the efficiency obtained from Table 2-1.

To calculate the maximum number of bits that can be transmitted in one TTI, we assume the best channel conditions and from Table 2-1 for CQI=15 we obtain the highest possible modulation scheme and coding rate which is 64QAM i.e. 6 symbols per slot, and ECR = 0.9258.

**4.3.2 Selection Criteria**

The normal case scenario in any wireless network is that the requests of the active users exceed the maximum capacity of the channel. The basic idea of the time domain scheduler is to select from the set of active users the users that
Users are sorted in a descending order of priority. As mentioned in the previous section the priority is a function of QoS requirements and buffer queue size. Users are scheduled in this TTI as long as the cumulative throughput request did not exceed the maximum capacity.

It is possible that users may have equal priority. When sorting these users the user with the worse channel conditions is given higher priority than the user with better channel conditions. This is done to improve the users’ chance of finding a good enough channel to transmit on. The time domain scheduler is depicted in Figure 4-2.

1. Find the set of active users (users with traffic in their buffers)
2. Sort the active users according to their priority as assigned by the priority function.
3. In case more than one user have the same priority, sort the equal priority users such that the worst channel gets served first.
4. Calculate the maximum capacity of the system.
5. The first group of users whose scheduling request fills the maximum capacity is chosen to be served in this TTI.

**Figure 4-2 : Time Domain Scheduler**

**4.4 Frequency Domain Scheduler**

The frequency domain scheduler assigns the transmission bandwidth to the users selected by the time domain scheduler. In addition to the transmission bandwidth the frequency domain scheduler is also responsible for assigning the MCS and adjusting the uplink power.

In short the time domain scheduler is responsible for selecting the users
to be served in this TTI and the frequency domain scheduler is responsible for selecting and assigning the resources to these users.

The users are passed to the frequency domain scheduler sorted by their priorities. The frequency domain scheduler serves the users one at a time starting with the highest priority user. Each user is served until the user’s request is satisfied or the resources are consumed.

Due to the fact that there is a high correlation in the frequency domain between the adjacent PRBs, this means that a PRB adjacent to a PRB with high gain will probably have high gain as well. From this the PRB assignment is done starting with the PRB with the highest gain and then assigning PRBs left and right of the maximum gain PRB. This should help reduce the number of PRBs needed to satisfy the user’s request.

1. Let U be the set of users to be served
2. Let A[i] be the allocation matrix, i is the PRB index. The value of A[i] is the user index assigned to the PRB.
3. Initialize A to be zeros indicating PRB is empty
4. Define PRB_Groupk as a set of empty contiguous RBs in A
5. for User = 1 to number of users do
6.   Get all available PRB_Groups in A
7.   for k = 1 to number of available PRB_Groups do
a. Calculate this user’s average gain on PRB_Groupk
b. Get maximum power allowed to the user set by the system
c. Estimate the user’s overall request from the BSR
d. Obtain the MCS from the power and calculate the average number of PRBs needed to satisfy the user request, the minimum of the calculated number of PRBs and the number of PRBs in the PRB_Group is taken.
e. Calculate the closed loop power adjustment (TPC)
f. Calculate the $P_{\text{PUSCH}}$ using the outputs of steps d and e.
g. Recalculate the needed number of PRBs and the MCS using the calculated transmission power in step f.
h. Find the PRB with the maximum gain in the PRB_Group.
i. Assign the peak PRB to the user.
j. Assign the PRBs left and right of the peak until the calculated number of PRBs is assigned.
8. end for
9. Find the PRB_Group that gives the highest number of bits to the user with the minimum number of PRBs
10. Assign this PRB_Group to this user, update A[i]=user_id for assigned PRBs
11. end for

Figure 4-3 : Frequency Domain Scheduler
For each user the frequency domain scheduler finds the set of schedulable groups of contiguous PRBs. For each PRB group the number of bits that the user can send on this group is calculated. The PRB group which satisfy the user’s request in the minimum number of PRBs or the PRB group which gives the user the highest number of bits that can be sent, is assigned to this user. This is repeated until all the users are served or all the bandwidth is assigned.

4.5 SINR-CQI Mapping and AMC

To perform AMC the scheduler at the eNB needs to know the channel conditions for each user, such that the appropriate modulation scheme and coding rate is assigned. The channel conditions are estimated from the received uplink SRS. The channel estimation from the SRS is outside the scope of this thesis and therefore perfect channel knowledge is assumed.

The next step is to calculate the Signal to Noise Ratio (SNR) for the channel as in equation (4-8). It is more accurate to calculate the Signal to Interference and Noise Ratio (SINR) instead of the SNR, but we neglect interference power for simplicity without loss of generality.

\[
SNR = \frac{P_{RX}}{P_{\text{noise},NF}} \quad (4-8)
\]

where \(P_{RX}\) is the received signal power, \(P_{\text{noise}}\) is the total noise power, and \(NF\) is the noise figure at the receiver.

Since this is a fading channel the SNR is not used directly as is to obtain the channel CQI. An effective SNR is used instead to take the effect of the channel gain and path loss into consideration.

\[
SNR_{\text{fading}} = SNR * \frac{|H|^2}{PL} \quad (4-9)
\]
where $H$ is the channel response matrix and $PL$ is the path loss in [dB]

In [24], the authors generated the BLER-SNR curves for the 15 different MCSs available in the LTE standard (corresponding to the 15 CQI levels). Each curve is for one CQI, where the left most curve represent CQI=1, and the right most curve is for CQI=15. The simulations were carried out for SISO, AWGN channel without using HARQ.

![Figure 4-4: BLER-SNR curves for the 15 CQIs [24]](image)

The LTE standard specifies that the AMC scheme has to ensure a BLER of less than 10%. For each CQI value the minimum required SNR to achieve this target is obtained by accessing the curves with BLER = 10%. We form a table with the CQI and corresponding SNR values obtained for the 10% BLER. We refer to this table as the CQI-SNR 10% BLER table.

The SNR to CQI mapping function compares the SNR$_{\text{fading}}$ calculated for the user, with the CQI-SNR 10% BLER table we formed as explained
above. The CQI of the first SNR value that is less than or equal to the $\text{SNR}_{\text{fading}}$ is chosen for the user. The CQI is then used to access the CQI-MCS table to obtain the modulation scheme and coding rate to be assigned.
4.6 Closed Loop Power Control and ICIC

At first glance a tradeoff between increasing cell throughput and decreasing the generated interference on neighboring cells exist. Deeper analysis shows that if the overall system inter-cell interference decreased, this will provide higher system throughput due to decreasing erroneous transmissions and therefore decreasing retransmissions.

Still the scheduler is faced with the design challenge of maximizing the throughput while minimizing the generated interference which are two conflicting goals as maximizing throughput requires the use of high transmission power while minimizing interference requires the use of lower transmission power. Therefore a balance between the two should be found.

Previous work considered each issue separately, frequency domain scheduling aiming at maximizing throughput has been studied before as well as power control techniques to minimize inter-cell interference, but these studies did not consider the mutual effect these issues have on each other. Therefore the algorithms maximizing throughput did not analyze the effect on inter-cell interference and the algorithms minimizing interference did not consider maximizing the throughput.

In this thesis we present a closed loop power control mechanism that is an integrated part of the resource management framework. The mechanism considers the tradeoff between maximizing throughput and minimizing inter-cell interference and the effect of each requirement on the other is analyzed.

The mechanism is based on the LTE closed loop power control equation (2-1) which is explained in chapter 2.3

\[ P_{PUSCH} = \min\{P_{\text{max}}, 10\log_{10} M + P_0 + \alpha PL + \Delta_{TF} + f(\delta_{PUSCH})\}[dBm] \]
The main goal of the algorithm is setting $f(\delta_{PUSCH})$ such that interference is minimized and throughput is maximized. As can be seen from the above equation the power equation is also dependent on $M$ the number of PRBs assigned to the user. Previous work studying ICIC mostly assumed $M$ as a fixed number known from the beginning usually by equally dividing the bandwidth among the users.

The power control mechanism is presented here as a part of an integrated radio resource management framework assigning both $M$ and $f(\delta_{PUSCH})$ to meet both the requirements of system efficiency and ICIC. Contrary to previous work where power control mechanisms are applied after the frequency domain scheduling, in our work power control and frequency domain scheduling are done simultaneously in the frequency domain scheduler.

The ICIC algorithm is done as a 2 step process:

Step 1a: Using the maximum allowed transmission power for the UE class and the average channel gain we obtain the MCS which the user can transmit with and from that we calculate the number of required PRBs ($M$) to satisfy the user's request.

Step 1b: $f(\delta_{PUSCH})$ is calculated as described below.

Step 2a: Using the calculated number of PRBs ($M$) and $f(\delta_{PUSCH})$ obtained from step 1, the uplink transmission power is calculated using equation (2-1).

Step 2b: The required number of PRBs ($M$) to satisfy the user request is recalculated using the new transmission power.

Note: The above steps for calculating the number of PRBs $M$ and the transmission power can be repeated again but only 2 repetitions gave
satisfactory results.

The main power control mechanism is encapsulated in the TPC command which is done in step 1b, therefore this step shall be elaborated.

Most ICIC algorithms revolve around an interference threshold or SINR threshold and this one is no different. Two variations of the proposed power control scheme are described below.

The setting of either the SINR or the Interference targets has the goal of controlling or limiting the overall total generated interference on neighboring cells. This goal can be achieved with an emphasis on maximizing cell throughput or maximizing user satisfaction.

4.6.1 TPC Scheme 1: SINR Target

In this case there exists a SINR Target that all UEs must meet. The SINR Target can be fixed as a constant value for all the UEs or it can be a range of values operating around a fixed value. In conventional power control algorithms where path loss is not compensated for, all UEs have the same SINR Target regardless of their position or their path loss. In Fractional Path loss Compensation (FPC) power control algorithms the SINR Target is adjusted for each UE according to its path loss, the adjustment is made for only a fraction of the path loss.

Our scheme is a FPC power control scheme. Accordingly before setting the power level for a UE by setting its TPC command, we adjust the UE’s SINR Target according to its path loss.

4.6.1.1 SINR Target Adjustment

It should be noted that since the path loss is proportional to the distance between the UE and the serving eNB, therefore the path loss is proportionally
inverse to the distance between the UE and the cell edge, which means that the UE is close to neighboring eNBs. This means that UEs with high path loss are cell edge users and therefore are more likely to generate interference on neighboring cells. Accordingly to decrease inter-cell interference cell edge users should transmit at low power while users who are closer to the serving eNB can transmit at higher powers. This can be explained that users who have low path loss to the serving eNB, will have high path loss to the neighboring eNBs. Consequently this high path loss will degrade the signal power at the neighboring eNBs making it less likely to interfere on the neighboring cells.

This is translated into the SINR Target adjustment. Cell edge users should transmit at powers that satisfy the SINR Target. Users whose path loss or distance from the cell edge allow for higher transmission powers can transmit at higher SINR Targets. The new SINR Target that is adjusted individually according to the user’s path loss is referred to as $SINR_{Target_{adj}}$. This way the distance from the neighboring cells is exploited to increase the throughput while considering the inter-cell interference. Therefore in this variation the emphasis is on maximizing overall cell throughput.

Let us define $PL_{max}$ as the value where users with path loss equal to or higher than this value, should transmit using the maximum transmission power in order for the signal to have enough strength to respect the SINR Target. The adjusted SINR Target $SINR_{Target_{adj}}$ is obtained using equation (4-10) [17]

$$
SINR_{Target_{adj}} = \\
\begin{cases} 
((\alpha - 1) \ast (PL - PL_{max}) + SINR_{Target}), & PL < PL_{max} \\
SINR_{Target}, & PL \geq PL_{max}
\end{cases} \quad (4-10)
$$

where $\alpha$ is the path loss compensation factor, $PL$ is the user’s path loss to the serving eNB, and $SINR_{Target}$ is the set SINR Target.
The above equation shows that users with path loss that give the user power headroom to use higher transmission power will have a higher SINR Target. Power headroom is the difference between the transmission power and the maximum allowed transmission power. Therefore equation (4-10) increases the SINR Target for users closer to the eNB (have lower path loss).

The LTE PUSCH power control adjustment state is given by the function $f(\delta_{PUSCH})$. The function can represent either accumulation or current absolute value. In this study it is considered to represent current absolute value. In the case of current absolute value, the values taken by $\delta_{PUSCH}$ in dB are {-4,-1,1,4}.

The adjusted SINR Target and the received SINR are entered into a mapping function to obtain the value of $\delta_{PUSCH}$. The mapping function is described in Figure 4-5.

1. Calculate difference between received SINR and adjusted SINR Target
2. Using the SINR difference calculate the required transmission power needed to be added to the received SINR to reach the adjusted SINR Target.
3. Obtain the TPC command as follows
   
   If $P_{\text{required}} \leq -4$ then $TPC = -4$
   
   elseif $-4 < P_{\text{required}} \leq -1$ then $TPC = -1$
   
   elseif $-1 < P_{\text{required}} \leq 1$ then $TPC = 1$

   else $TPC = 4$


**Figure 4-5 : Mapping Function**
4.6.2 TPC Scheme 2: Total Generated Interference Limit

In the second scheme instead of setting an individual SINR Target, a total target on the overall cell interference is set. In other words, a maximum limit for the total interference generated by all the users in one cell is set. For simplicity the cell interference limit is divided among the users in the cell such that each user will have an interference limit it has to respect.

4.6.2.1 Interference Limit Distribution Among the Users

To divide the interference limit on the users present in the cell, there are three different options.

Option1: The interference limit is equally divided among the users. This provides the simplest solution and at the same time provides fairness among the users as well as expected improved interference performance metrics. The downside of using this option is that the large distance of the cell center users from the cell edge is not exploited to increase throughput, neither are the cell edge users given any advantage on the cell center users to compensate for their large path loss. Therefore while this option provides a simple solution with good performance in terms of throughput, it does not provide improved
performance in terms of throughput nor user satisfaction.

Option 2: The second option would be to divide the interference limit in such a way that cell edge users get a higher limit than cell center users. This option exploit the fact that the cell center users are far enough from the neighboring eNBs such that increasing their transmission power will not affect the interference performance and at the same time improve the cell throughput. However in this case it should be noted that cell edge users are now at a more of a disadvantage since in addition to their high path loss they will also have a high interference limit to respect. Therefore despite the improved cell throughput, the overall user satisfaction will be degraded.

Option 3: The final option is to divide the interference limit in such a way that cell edge users get a lower interference limit than cell center users. The idea is that since there is an overall interference limit on the cell, increasing the cell edge users transmission power should degrade the system performance in terms of interference only slightly, however a kind of fairness will be provided to the users as the users with high path loss will be compensated by having a higher allowed transmission power. This scenario should increase cell edge throughput and accordingly cell throughput as well as improving user satisfaction.

It is noted that while option 1 no information about the users is needed, in option 2 and option 3 we need to know the distance of the users from the serving eNB. Realistically the distance between the user and eNB could not be obtained, however and estimate of the distance can be obtained from the user path loss and as explained before PHR sent from the UEs can be used to calculate the path loss. For further simplicity instead of estimating the distance from the path loss and since the path loss and the distance are proportional to each other, the path loss is used directly as an indication of how far the UE is
from the serving eNB.

The interference limit $IL_i$ for each user $i$ can be calculated by the following equation:

$$IL_i = \frac{CIL \cdot w_i}{\sum_k w_k} \quad (4-11)$$

$$CIL = \sum_i IL_i \quad (4-12)$$

where CIL is the cell interference limit in watts, and $w_i$ is the weight of user $i$. According to which option will be implemented the weights of the users are determined. For option 1, $w_i = 1$, $\forall$ $i$, for option 2, $w_i \in \{0,0.25,0.5,1,1.5,1.75,2,2.5\}$, and for option 3, $w_i \in \{0,0.25,0.5,1,1.5,2\}$ where $w_i =1$ for the user with average path loss for both options 2 and 3. The values for the weights are obtained by experimenting with different sets of values until a set is reached which provides good performance.

Note: The sum of the weights is not necessarily equal to one, but the sum of the interference limits per user $IL_i$ is equal to the cell interference limit $CIL$ as indicated in equation (4-12).
The proposed RRM framework is described in this chapter. The RRM functionality is divided into 3 stages. Priority assignment, time domain packet scheduling, and frequency domain packet scheduling.

### Scheme 2 Mapping Function

1. Calculate the power that generates interference equal to the interference limit
2. From the received power calculate the uplink interference generated.
3. Subtract the 2 powers and obtain the $P_{\text{difference}}$
4. Obtain the TPC command as follows
   
<table>
<thead>
<tr>
<th>Condition</th>
<th>TPC</th>
</tr>
</thead>
<tbody>
<tr>
<td>$P_{\text{difference}} \leq -4$</td>
<td>-4</td>
</tr>
<tr>
<td>elseif $-4 &lt; P_{\text{difference}} \leq -1$</td>
<td>-1</td>
</tr>
<tr>
<td>elseif $-1 &lt; P_{\text{difference}} \leq 1$</td>
<td>1</td>
</tr>
<tr>
<td>else</td>
<td>4</td>
</tr>
</tbody>
</table>

### Figure 4-7: Calculating TPC command Scheme 2

### Figure 4-8: Scheme 2 Mapping Function

### 4.7 Summary

The proposed RRM framework is described in this chapter. The RRM functionality is divided into 3 stages. Priority assignment, time domain packet scheduling, and frequency domain packet scheduling.
The priority assignment and the time domain packet scheduler are mainly concerned with meeting the QoS requirements. The frequency domain packet scheduler performs channel dependant scheduling with contiguity constraint, AMC and power control.

ICIC was implemented through two power control algorithms. The first algorithm controlled the users’ power individually, while the second algorithm handled the power control on a cell level.
Chapter 5
System Model and Results Analysis

5.1 System Model

The simulated system is based on the 3GPP LTE. The scope of this thesis is the uplink transmission only which use SC-FDMA as the radio access technology.

Simulations are carried out using MATLAB code. The channels are generated using the WINNER II Channel Models [25].

The system is simulated as a single cell of radius 500 m using the suburban macro-cell scenario C1. In suburban macro-cells the eNB is located well above the rooftops and UEs are outdoors at street level. Buildings are typically low residential detached houses with one or two floors, or blocks of flats with a few floors. Occasional open areas such as parks or playgrounds between the houses make the environment rather open. Streets do not form urban-like regular strict grid structure. Vegetation is modest.

The path loss models used in the WINNER II are of the form

\[ PL = A \log_{10}(d[m]) + B + C \log_{10}\left(\frac{f_c[GHz]}{5.0}\right) + X \quad (5.1) \]

where \( d \) is the distance between the transmitter and receiver in [m], \( f_c \) is the center frequency in [GHz], the fitting parameter \( A \) includes the path loss exponent, \( B \) is the intercept, \( C \) describes the path loss frequency dependence, and \( X \) is an optional environment specific term.

For suburban macro-cell C1 scenarios, \( A=23.8, B=41.2 \) and \( C=20, \)
\( h_{eNB} = 25 \) m, \( h_{UE} = 1.5 \) m

**Table 5-1 : Path loss Model for C1 WINNER II Scenario [25]**

<table>
<thead>
<tr>
<th>Type</th>
<th>Formula</th>
<th>( \sigma )</th>
</tr>
</thead>
<tbody>
<tr>
<td>LOS</td>
<td>( PL = 40.0 \log_{10}(d) + 11.65 - 16.2 \log_{10}(h_{eNB}) - 16.2 \log_{10}(h_{UE}) + 3.8 \log_{10}(f_c/5.0) )</td>
<td>30 m &lt; ( d ) &lt; ( d_{BP} )</td>
</tr>
<tr>
<td>NLOS</td>
<td>( PL = ((44.9 - 6.55 \log_{10}(h_{eNB})) \log_{10}(d) + 31.46 - 5.83 \log_{10}(h_{eNB}) + 23 \log_{10}(f_c/5.0) )</td>
<td>50 m &lt; ( d ) &lt; 5 km</td>
</tr>
</tbody>
</table>

The center frequency according to the LTE specification \( f_c = 2 \)GHz.

According to [29], the LTE specifications support spectrum flexibility by having a list of spectrum allocations. This allow mobile operators to choose the spectrum allocation which is suitable for the user density in an area. The lower the user density the smaller the bandwidth they need and therefore this allows them to save on the bandwidth they purchase. Table 5-2 show the allowed spectrum allocations and their corresponding transmission bandwidth measured in terms of number of PRBs.

**Table 5-2 : Transmission bandwidth configuration \( N_{RB} \) in E-UTRA channel bandwidths [29]**

<table>
<thead>
<tr>
<th>Channel bandwidth ( BW_{Channel} ) [MHz]</th>
<th>1.4</th>
<th>3</th>
<th>5</th>
<th>10</th>
<th>15</th>
<th>20</th>
</tr>
</thead>
<tbody>
<tr>
<td>Transmission bandwidth configuration ( N_{RB} )</td>
<td>6</td>
<td>15</td>
<td>25</td>
<td>50</td>
<td>75</td>
<td>100</td>
</tr>
</tbody>
</table>

The users belong to one of four traffic classes: VoIP, FTP, Interactive Gaming and Video Streaming according to the distribution shown in Table 5-3. The overall system parameters used in the simulations are shown in Table 5-4.
Table 5-3: Traffic Classes Distribution

<table>
<thead>
<tr>
<th>Traffic Class</th>
<th>Percentage</th>
</tr>
</thead>
<tbody>
<tr>
<td>VoIP</td>
<td>30%</td>
</tr>
<tr>
<td>FTP</td>
<td>30%</td>
</tr>
<tr>
<td>Interactive Gaming</td>
<td>20%</td>
</tr>
<tr>
<td>Video Streaming</td>
<td>20%</td>
</tr>
</tbody>
</table>

Table 5-4: Simulation Parameters Summary

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of simulation runs</td>
<td>10</td>
</tr>
<tr>
<td>Simulation run duration</td>
<td>60 seconds</td>
</tr>
<tr>
<td>Center Frequency</td>
<td>2GHz</td>
</tr>
<tr>
<td>System Bandwidth</td>
<td>10MHz</td>
</tr>
<tr>
<td>Subcarrier Spacing</td>
<td>15KHz</td>
</tr>
<tr>
<td>FFT Size</td>
<td>1024</td>
</tr>
<tr>
<td>$F_s$</td>
<td>15.36MHz</td>
</tr>
<tr>
<td>Channel Model</td>
<td>C1 – Suburban Macro-cell</td>
</tr>
<tr>
<td>User Speed</td>
<td>60m/s</td>
</tr>
<tr>
<td>Number of OFDM symbols per slot</td>
<td>7</td>
</tr>
<tr>
<td>Slot Duration</td>
<td>0.5ms</td>
</tr>
<tr>
<td>TTI</td>
<td>1ms</td>
</tr>
<tr>
<td>Noise Power</td>
<td>-160dBm/Hz</td>
</tr>
<tr>
<td>Rx Noise Figure</td>
<td>5dB</td>
</tr>
<tr>
<td>Maximum User Power</td>
<td>24dBm</td>
</tr>
<tr>
<td>$\alpha$[18]</td>
<td>0.8</td>
</tr>
<tr>
<td>$P_o$[18]</td>
<td>-81dBm/Hz</td>
</tr>
<tr>
<td>Number of users</td>
<td>100</td>
</tr>
<tr>
<td>SINR Target</td>
<td>1dB</td>
</tr>
</tbody>
</table>

5.1.1 Calculating FFT Size and Sampling Frequency

To transform from time domain to frequency domain, FFT is employed
therefore we need to obtain the correct FFT size $N_{FFT}$

From the number of PRBs $N_{RB}$ we obtain the number of subcarriers $N_{SC}$

\[ N_{SC} = N_{RB} \times RB_{size} \quad (5-2) \]

where $RB_{size} = 12$ subcarriers/PRB

\[ N_{FFT} = 2^{\text{ceil}(\log_2 N_{SC})} \quad (5-3) \]

where ceil(x) is rounding x to the highest integer.

The system sampling frequency $F_s$ can then be calculated as

\[ F_s = N_{FFT} \times \Delta F \quad (5-4) \]

where $\Delta F = 15KHz$

5.1.2 WINNER II Model

The WINNER II channel model is implemented in MATLAB, input/output interface is defined as follows where ([ ] indicate optional arguments):

\[ [H, [DELAYS], [FULL_OUTPUT]] = \]
\[ \text{WIM} ( \text{WIMPAR}, \text{LAYOUTPAR}, [\text{INITVALUES}] ) \]

Input Parameters:

- **WIMPAR**: This structure contains general simulation parameters. The model generates default values for the structure. In our simulations the default values are used except for the parameters shown in Table 5-5.

- **LAYOUTPAR**: Defines position of terminal stations, their
assigned antenna arrays and gives links of interest for simulation.

- **INITVALUES**: Parameters of the propagation channel. When this parameter is given WIM does not generate the channel parameters randomly, but uses the supplied initial channel values.

Output Parameters:

- **H**: MIMO channel 5Dimensional-matrix which is collection of time-variant Channel Impulse Responses (CIRs) (=f(t,τ)) between all (Tx,Rx) pairs, for all links defined.

- **DELAYS**: Multipath delays for all links, given in seconds.

- **FULL_OUTPUT**: Stores the randomly generated link parameters and the final phases of the complex sinusoids. This MATLAB structure can be used as INITVALUES in subsequent function calls to generate time continuous channel realizations with separate function calls.

**Table 5-5 : WIMPAR parameters values**

<table>
<thead>
<tr>
<th>Parameter Name</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>CenterFrequency</td>
<td>2GHz</td>
</tr>
<tr>
<td>NumTimeSamples</td>
<td>1000</td>
</tr>
<tr>
<td>SampleDensity</td>
<td>1.2491</td>
</tr>
<tr>
<td>PathLossModelUsed</td>
<td>‘No’</td>
</tr>
<tr>
<td>ShadowingModelUsed</td>
<td>‘No’</td>
</tr>
</tbody>
</table>

5.1.2.1 Calculating SampleDensity

The sample density represents the number of time samples per half wavelength which indicates the sampling frequency of the channel. Therefore to obtain the required sampling frequency we must set the SampleDensity
correctly.

\[
Sample \ Density = \frac{s_{\text{light}}}{2 \cdot f_c \cdot v_{UE} \cdot T_{\text{channel}}} \tag{5-5}
\]

where \( s_{\text{light}} \) is the speed of light and equals 299792458 m/s, \( f_c \) is the carrier frequency, \( v_{UE} \) is the velocity of the UE and \( T_{\text{channel}} \) is the channel sampling time.

For fast fading channels the channel sampling time is the system sampling time. This is the case when we are interested in the channel at every time instant for example when we need to convolve the channel with a bit stream. In this case we set \( T_{\text{channel}} = 1/F_s \).

Flat fading is where we assume that the channel is constant with no variations during a block of time. For flat fading channels the channel sampling time is the block time. This is the case when we are interested in the channel value at certain instances for example at the beginning of each TTI.

In our scheme, we schedule users on each TTI, so channel variations lower than one TTI are not captured and therefore we set the channel sampling time to flat fading with \( T_{\text{channel}} = 1TTI = 1ms \).

5.1.2.2 Transforming Channel Matrix to Frequency Domain

The output of the WINNER model is a cell array of length equal to the number of links. Each element in the array is a 4-D (UxSxTxN) channel matrix \( h \).

U: Number of receiver elements

S: Number of transmitter elements

T: Number of time samples
N: Number of paths/taps

The output matrix is in the time domain and since all our work is on PRBs in the frequency domain we need to transform the channel matrix from the time domain to the frequency domain using FFT. Before applying FFT the channel matrix needs to be adjusted following the below steps:

1- The channel matrix resulting from the WINNER model may contain Not any Number (NaN) values, which are MATLAB representation for invalid values. Remove NaN values from the channel matrix

2- Round the delays of the different paths to the nearest multiple of the system sampling time $T_s=1/F_s$.

3- Sum the taps that have the same delay

4- For the delays that have no taps insert zeros. e.g if we take $T_s = 5$ msec, we should have taps(path values) at delays 0,5,10,15 etc, there could exist taps/paths at 0,10,15 so we need to insert 0 at the second position in the taps array.

5- Normalize the channel energy

6- Perform FFT on the adjusted channel

5.1.3 Traffic Models

5.1.3.1 FTP Traffic Model

FTP traffic model is based on a Bernoulli process. Packets are generated every 5 msec with probability 0.5. The packet size follows a uniform distribution with the parameters shown in Table 5-6.
Table 5-6 : FTP Traffic Model Parameters [30]

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Statistical Characterization</th>
</tr>
</thead>
<tbody>
<tr>
<td>Packet Size</td>
<td>Uniform Distribution: $f_x = \frac{1}{b-a}, a \leq x \leq b, a = 200\text{bytes}, b = 1500\text{bytes}$</td>
</tr>
<tr>
<td>Packet arrival</td>
<td>Every 5 ms with probability =0.5</td>
</tr>
</tbody>
</table>

Following the above distribution the FTP rate can be obtained

$$Average \ Packet \ Size = \frac{1500 + 200}{2} = 850 \text{ bytes} \quad (5-6)$$

$$R_{FTP} = \frac{(850 + 0.5)}{5 \times 10^{-3}} = 680 \text{ Kbps} \quad (5-7)$$

5.1.3.2 VoIP Traffic Model

The VoIP is modeled as a 2-state voice activity model.

![2-state voice activity model](image)

Figure 5-1 : 2-state voice activity model [30]

The parameters of the VoIP traffic source are given in Table 5-7.

Table 5-7 : VoIP Traffic Model Parameters [30]

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Characterization</th>
</tr>
</thead>
<tbody>
<tr>
<td>Codec</td>
<td>RTP AMR 12.2, Source rate 12.2 kbps</td>
</tr>
<tr>
<td>Parameter</td>
<td>Statistical Characterization</td>
</tr>
<tr>
<td>---------------------------------------</td>
<td>------------------------------------------------------------------</td>
</tr>
<tr>
<td>Encoder frame length</td>
<td>20 ms</td>
</tr>
<tr>
<td>Voice activity factor (VAF)</td>
<td>50% (c=0.01, d=0.99)</td>
</tr>
<tr>
<td>SID payload</td>
<td>Modeled</td>
</tr>
<tr>
<td></td>
<td>15 bytes (5 Bytes + header)</td>
</tr>
<tr>
<td></td>
<td>SID packet every 160 ms during silence</td>
</tr>
<tr>
<td>Protocol Overhead with compressed</td>
<td>10 bit + padding (RTP-pre-header)</td>
</tr>
<tr>
<td>header</td>
<td>4 Byte (RTP/UDP/IP)</td>
</tr>
<tr>
<td></td>
<td>2 Byte (RLC/security)</td>
</tr>
<tr>
<td></td>
<td>16 bits (CRC)</td>
</tr>
<tr>
<td>Total voice payload on air interface</td>
<td>40 bytes (AMR 12.2)</td>
</tr>
</tbody>
</table>

Following the above distribution the VoIP rate can be obtained

\[
R_{VoIP} = R_{silent} \times \text{probability}_{silent} + R_{voice} \times \text{probability}_{voice}
\]

\[
= \frac{15+8}{160 \times 10^{-3}} \times 0.5 + \frac{40+8}{20 \times 10^{-3}} \times 0.5 = 8375 \text{ bps} \quad (5-8)
\]

Note: The 12.2 Kbps is the source rate that is it is the rate of the voice data without any overheads in the active state.

5.1.3.3 Gaming Traffic Model

The gaming traffic model follow the distribution and parameters shown in Table 5-8.

Table 5-8 : Uplink Gaming Network Traffic Parameters [30]
Packet size  |  Largest Extreme Value Distribution (also known as Fisher-Tippett distribution)  
|  \( f_x = \frac{1}{b} e^{-\frac{x-a}{b}} e^{-\frac{y}{x-a}} \), \( a = 45 \text{Bytes}, b = 5.7 \)  
|  Values for this distribution can be generated by the following procedure:  
|  \( x = a - b \ln(-\ln y) \), where \( y \) is drawn from a uniform distribution in the range \([0,1]\)  
|  Because packet size has to be integer number of bytes, the largest integer less than or equal to \( x \) is used as the actual packet size  

UDP header  |  Deterministic (2 Bytes). This is added to the packet size accounting for the UDP header after header compression.  

---

**Average Packet Size = 41.7 bytes**

Following the above distribution the gaming rate can be obtained

\[
R_{\text{gaming}} = \frac{((41.7+2)*8)}{40*10^{-3}} = 8.74 \text{ Kbps} \quad (5.9)
\]

5.1.3.4 Video Streaming Traffic Model

Video data is divided into frames where each of these frames is decomposed into a fixed number of slices each transmitted as a single packet. The size of these packets and their inter-arrival time are modeled to have a truncated Pareto distribution, while the number of packets in a frame and the inter-arrival time between frames is deterministic. The traffic model assumes a source video rate of 64 Kbps with other parameters and distributions defined in Table 5-9.

**Table 5-9 : Video Streaming Traffic Parameters [30]**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Statistical Characterization</th>
</tr>
</thead>
<tbody>
<tr>
<td>Inter-Arrival time between the</td>
<td>Deterministic</td>
</tr>
<tr>
<td>beginning of each frame</td>
<td>100 ms (based on 10 frames per second)</td>
</tr>
<tr>
<td>Number of packets (slices) in a</td>
<td>Deterministic, 8 packets per frame</td>
</tr>
<tr>
<td>frame</td>
<td></td>
</tr>
</tbody>
</table>

68
| Packet (slice) size | Truncated Pareto Distribution  
|-------------------|---------------------------------|  
|                   | Minimum= 53 Bytes, Mean=100 Bytes, Maximum =250 Bytes  
|                   | (Before Truncation)  
|                   | PDF: $f_x = \frac{\alpha x^\alpha}{\alpha + 1}, k \leq x < m$ , $f_x = \left(\frac{k}{m}\right)^\alpha, x = m$ ,  
|                   | $\alpha = 1.2, k = 20$Bytes  
| Inter-arrival time between packets (slices) in a frame | Truncated Pareto Distribution  
|                   | Minimum= 2.5 ms, Mean=$m=6$ ms, Maximum =$12.5$ ms (Before Truncation)  
|                   | PDF: $f_x = \frac{\alpha x^\alpha}{\alpha + 1}$, $k \leq x < m$ , $f_x = \left(\frac{k}{m}\right)^\alpha, x = m$ , $\alpha = 1.2, k = 2.5$ms  

Following the above distribution the video rate can be obtained

$$R_{video} = \frac{(800 \text{byte/frame} \times 10 \text{frames/sec} \times 8 \text{bits/byte})}{1} = 64 \text{ Kbps}$$

(5-10)
5.2 Simulation Results

5.2.1 Fixed Load and Fixed SINR Target

5.2.1.1 Comparison Algorithm

The proposed RRM framework is compared to the First Maximum Expansion (FME) algorithm described in [3]. Results for the throughput and packet delay are obtained for both the RRM framework and the FME algorithm. The main idea of the FME algorithm is starting the assignment with the PRB that has the highest metric value and “expanding” the assignment on both sides of the allocation matrix. The metric could be the proportional fair metric, or the channel gain. The algorithm steps are summarized below:

1- Search for highest metric in matrix and assign the RB to this UE (UE0-RB0)

2- Check for the maximum value of the metric in the column on the left and right of the assigned RB0. Expand in the direction of the higher maximum.

3- In the chosen direction for column by column get the maximum metric in this column. Assign this RB to the UE with this highest metric if the contiguity condition is not broken, otherwise to the UE assigned in the previous column. Repeat until all the columns in this direction are assigned.

4- Repeat step 3 for the other direction.
5.2.1.2 Calculation of maximum achievable throughput and moving average parameters

To have a benchmark to compare the simulation results with we calculate the maximum achievable throughput for the current TTI. The maximum achievable throughput is calculated by obtaining the set of active users and allocating a PRB to the user who has highest channel gain on this PRB. For the calculation of the maximum achievable throughput the contiguity constraint is neglected.

A moving average is calculated for the maximum achievable throughput and the throughput per TTI.

The moving average of parameter $X$ is calculated as

$$
X(t) = \alpha \cdot MovingAvgX(t-1) + (1-\alpha) \cdot X(t) \quad (5-11)
$$

where we set $\alpha=0.8$, giving higher weight to the average than the instantaneous value.

5.2.1.3 Fixed Load and Fixed SINR Target Results

The simulation is carried out according to the default parameters in Table 5-4 and the traffic mix in Table 5-3.
Packet Delay and QoS Requirements

Figure 5-2: Delay CDF for the four traffic classes obtained for the RRM framework and the FME Algorithm

According to the QoS requirements defined for the different traffic classes shown in Table 2-2, it can be seen from Figure 5-2 that for the proposed RRM framework the higher the QoS class priority the higher is the slope of the curve, which shows that higher priority packets has smaller delays. It is also noted that the three traffic classes VoIP, gaming and video, which have delay constraints have all the packets sent before the maximum delay is reached, while for FTP since it is a best effort traffic class the delay is considerably higher. Note also that because video has higher QoS class priority than gaming, it had high probability at low delays. On the other hand, since gaming has lower maximum delay requirement than video the curve slope increase drastically as it approaches the maximum delay. Comparing the results with the FME algorithm it can be seen that the priority as depicted by the QoS parameters was not respected.

Packets that exceed their maximum deadline are dropped from the buffers. We calculate the packet drop ratio as the number of dropped packets.
relative to the total number of generated packets. Results for the FME overall packet drop ratio is around 40%. For the proposed RRM framework no packets were dropped for VoIP and video traffic, while gaming and FTP had packet drop ratios in the ranges of $10^{-3}$ and $10^{-2}$, showing that the PELR QoS parameter as depicted in Table 2-2 has been respected as well.
5.2.1.3.2 Cell Throughput

Figure 5-3: Cell Throughput simulated for the RRM Framework and the FME algorithm and compared to the maximum achievable throughput.

Figure 5-3 shows the maximum attainable throughput for the system and also the achieved throughput using our RRM framework and the FME algorithm. The plots are for the moving average of the maximum attainable throughput and the achieved throughput. Although the FME algorithm gives higher throughput due to its usage of maximum transmission power and allocations of PRBs based only on channel conditions, it is seen that the proposed RRM framework yields throughput that is very close to the FME, it is noted that for the RRM framework due to respecting the QoS parameters and respecting the packets’ deadlines this helped in avoiding high packet drop rates.

It should be noted that the achieved throughput will always be less than the maximum achievable throughput due to:
1. Contiguity constraint: PRBs assigned to one user must be contiguous, which means that a PRB can be assigned to a user who is not the best candidate for this PRB, from the point of view of channel conditions, or a PRB can be assigned to a user who does not need the PRB to send data on but the PRB is assigned as it is in between PRBs assigned to this user.

2. QoS constraints: The QoS constraints are reflected in the user’s priority, a PRB can be assigned to a user who is not the best candidate since this user has higher priority than other users and is served first.

3. Request constraint: The PRBs are assigned depending on the user’s request. A user can be the best candidate for a PRB and is not assigned this PRB since he does not need it and his request is already fulfilled.

4. The inter-cell interference strategy that limits the transmission power.
5.2.1.3.3 Generated Interference

Figure 5-4 shows the total generated uplink interference for the proposed RRM framework and the FME algorithm.

Figure 5-4 shows the total generated uplink interference on neighboring cells. It can be seen that with no power control mechanism as is the case with the FME algorithm, high values of the generated interference are obtained, while a considerable performance gain is obtained by using power control. Since this evaluation is done considering a single cell, we can conjecture that in a multi-cell environment, the performance of schemes without power control will have an impact on throughput due to the increased interference from neighboring cells. From Figures 5-3 and 5-4 it can be seen that the effect of the tradeoff between minimizing inter-cell interference and maximizing throughput can be handled through having an integrated framework which considers both factors in the assignment.
5.2.2 Fixed Load and Different SINR Targets

In ICIC the set SINR Target is a very important parameter. The SINR Target defines the tradeoff between maximizing cell throughput and minimizing inter-cell interference. Accordingly we need to analyze the different values of the SINR Target to set it at a value that gives high throughput at acceptable inter-cell interference.

5.2.2.1 Total Generated Interference Estimation

To check the effect of the SINR Target on the inter-cell interference, the total generated interference by the cell is calculated.

Using the same path loss model used to calculate the path loss between the serving eNB and the UE, the path loss between the UE and the neighboring eNB is calculated. It is assumed that the UE will cause interference on one neighboring eNB only that is the closest neighboring eNB to the UE. The distance between the UE and the neighboring eNB is calculated using the below equation.

\[ d_{\text{neighbor}} = \text{CellRadius} + (\text{CellRadius} - d_{\text{serving}}) \] (5-12)

where \( d_{\text{serving}} \) is the distance between the UE and the serving eNB. In our simulations it is obtained from the WINNER model output, while in real life it is estimated from the path loss. The \( \text{CellRadius} \) equals 500 m.

The total interference is the sum of the interference generated by all the active UEs on the serving cell.
5.2.2.2 Fixed Load and Different SINR Targets Results

The default simulation parameters defined in Table 5-4 are used with the following exceptions.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Simulation run duration</td>
<td>30 seconds</td>
</tr>
<tr>
<td>Number of users</td>
<td>20</td>
</tr>
<tr>
<td>Users traffic type</td>
<td>FTP</td>
</tr>
<tr>
<td>SINR Target range</td>
<td>[-5 5] dB</td>
</tr>
</tbody>
</table>
As expected the higher the SINR Target the higher the generated uplink interference, as seen in Figure 5-6. It should be noted that the increase in the interference is not very high due to the adjustments done on the SINR Target for cell edge users. It can be seen that the increase in the interference beyond SINR Target 1dB is small.
Higher SINR Targets means higher transmission power which is mapped to higher MCS and therefore higher throughput. This is evident in Figure 5-7. The high throughput values are due to the adjustments done on the SINR Target for users far from the cell edge. Users who are near the cell center and far from the cell edge are given an adjusted SINR Target which is higher than the SINR Target, this allow them to transmit with higher power without having concerns on the generated interference.

From the above graphs the optimum SINR Target can be chosen by looking for the SINR Target that yields high throughput with an acceptable generated interference.
5.2.3 Fixed Load and Different Interference Limit

The second scheme for ICIC involves the use of an overall cell interference limit instead of a user SINR Target. Three different options were proposed on how to divide the interference limit on the users.

- Option 1: Divide the limit equally on the users.
- Option 2: Lower weights (limits) assigned to users with higher path loss, giving them lower transmission power.
- Option 3: Lower weights (limits) assigned to users with lower path loss, giving them lower transmission power.

The default simulation parameters defined in Table 5-4 are used with the following exceptions.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Simulation run duration</td>
<td>30 seconds</td>
</tr>
<tr>
<td>Number of users</td>
<td>20</td>
</tr>
<tr>
<td>Users traffic type</td>
<td>FTP</td>
</tr>
<tr>
<td>SINR Target</td>
<td>Not Used</td>
</tr>
</tbody>
</table>
As can be seen from Figure 5-8 when low interference limit is given to users with high path loss this gives the best result as can be expected as the users who transmit with high power are far from the cell edge. It is noted that as the set cell interference limit increase so does the generated cell interference up until around -105 dBm, where the generated cell interference saturates. It can also be seen that the generated cell interference is always equal to or lower than the cell interference limit.
Figure 5-9 shows the total cell throughput versus the cell interference limit. As the interference limit decreases, the throughput increases due to the higher transmission power that can be used, until the power reaches the maximum allowed transmission power beyond which any decrease in the interference limit does not have an effect on the throughput, as can be seen at around -105 dBm. When low weights are given to users with high path loss, this means that users with low path loss will get higher weights leading to higher limits and therefore they can transmit with higher powers, which leads to an increase in the overall cell throughput as the users with good channel conditions are favored.
Figure 5-10: Cell Edge Throughput vs. Cell Interference limit

Figure 5-10 shows the 10 percentile cell throughput which is an estimate for the cell edge throughput. It is evident that when cell edge users (users with high path loss) are allowed to transmit with high powers, by assigning them higher weights and larger limits, this improve the cell edge throughput.

From Figure 5-8, Figure 5-9 and Figure 5-10 it is noticed that beyond -105 dBm interference limit the three options start to yield the same results whether in terms of generated interference or throughput, therefore we can deduce that the maximum interference that can be generated by the cell is around -105 dBm, and from the three figures we can choose an interference limit to set for the cell.
5.2.4 Different Loading and Fixed SINR Target

In this section we will analyze the system performance under different load conditions. For simplicity the system model will only include two traffic classes: (1) VoIP and (2) FTP. The VoIP loading was fixed at 209 Kbps and the system load was increased by increasing the FTP load starting from 1100 Kbps until reaching 6600 Kbps.

5.2.4.1 Modified FTP Traffic Model

The same traffic model defined in Table 5-6 is used with the set of parameters shown in Table 5-10.

**Table 5-10 : FTP Traffic Model Parameters for System Loading Simulation**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Minimum Packet Size</td>
<td>15 bytes</td>
</tr>
<tr>
<td>Maximum Packet Size</td>
<td>40 bytes</td>
</tr>
<tr>
<td>Probability of packet generation</td>
<td>0.5</td>
</tr>
<tr>
<td>Packet Arrival Rate</td>
<td>5 msec</td>
</tr>
</tbody>
</table>
5.2.4.2 Different Loading and Fixed SINR Target Results

As can be seen from Figure 5-11, as long as the load is small and the system bandwidth is sufficient the delay is almost near zero. As the number of FTP load and accordingly the system load increase the delay increase. It is noted that the VoIP delay start to increase slightly but remain within its limit of 50 msec when the FTP load reach 4400 Kbps, in comparison the FTP delay increase dramatically when the system load increase. This shows that the QoS parameters were respected and at the same time the delay of the best effort traffic was minimized as much as the system can support, from this figure we can estimate the system capacity beyond which the performance starts to degrade.
Figure 5-12: Throughput vs. FTP load

As can be expected as the load (number of users) increase the throughput increase as seen in Figure 5-12, this is due to more efficient use of the bandwidth as well as better exploitation of the multi user diversity. However this is only applicable until we reach system saturation. As the number of VoIP users is constant at 25 therefore the VoIP throughput is almost constant, while as the number of FTP users increase the FTP throughput increase as well.
Chapter 6
Conclusions and Future Work

6.1 Conclusions

In this thesis we proposed a complete radio resource management framework for the LTE uplink transmission. The system model represented multiple users with different channel conditions, QoS profiles and different traffic generation parameters. The formulated problem is to determine on a TTI basis the users who will be served in this TTI, the number and location of the PRBs for each user, and the allocated MCS and power to transmit on these PRBs.

It is shown from the literature that solving the frequency domain packet scheduling problem alone is NP-hard therefore the problem of complete resource management which is a more complex problem will be impractical to solve optimally in real time systems. We proposed a heuristic three step algorithm that solves the problem of complete resource management. The framework maximizes the system throughput while minimizing the inter-cell interference and considering the QoS requirements of the users.

The three main goals of the framework were: (1) Respect QoS parameters of the different users, (2) Maximize system throughput, and (3) Minimize inter-cell interference. To analyze the framework in terms of these goals three simulation setups were done.

Simulation setup 1: Four different traffic types with different QoS parameters and requirements were used. Fixed number of users and fixed SINR_Target was also used. The simulation results show that the QoS requirements represented by the packet delay and the packet error loss ratio of the different classes have been satisfied while minimizing the reduction in
throughput.

Simulation setup 2: Fixed number of users belonging to four different QoS classes is simulated. The SINR\_Target was gradually increased to analyze the effect on the inter-cell interference. It has been shown that there is a tradeoff between maximizing throughput and minimizing inter-cell interference.

Simulation setup 3: With fixed SINR\_Target, number of users belonging to two different QoS classes was increased gradually. The effect of system loading has been analyzed in terms of throughput and packet delay. The maximum system load beyond which the system is overloaded and contention starts to occur is discovered.

### 6.2 Future Work

Two Inter-Cell Interference Coordination schemes were proposed in the thesis, however simulations were done using a single cell. Future work will include multi cell simulations to better study the schemes and how to enhance them. In multi cell simulations there will be two types of networks: homogenous and heterogeneous. Where the interference limit will be the same for all cells in homogenous networks, the interference limit can be different for different cells in heterogeneous networks. This will allow to study which type of network has better performance as well as studying how the different interference limits will be set for each cell in case of heterogeneous networks.

The proposed framework addressed some implementation issues of the LTE standard. However some more aspects may be considered including, Hybrid Automatic Repeat Request (HARQ), and actual available bandwidth for allocation. The inclusion of the HARQ will have two effects. First a more realistic implementation where perfect channel knowledge is not assumed will
be implemented, i.e. probability of erroneous transmissions and retransmissions will be considered. The second effect is that the packet scheduler must consider the HARQ retransmissions first, as according to the LTE standard HARQ retransmissions must use the same PRBs used in the first transmission, as well as that they have to be scheduled first before new transmissions.
REFERENCES


[25] WINNER II channel model


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ملخص

التحسن في الأداء في نظام "التطور البعيد المدى" اللاسلكية على نظام "الجيل الثالث" تأتي على حساب إضافة قيود وتحديات جديدة لتصميم النظام. في هذه الرسالة نقدم اطار شامل لتوزيع الموارد في نظام "التطور البعيد المدى" اللاسلكية للارسال من المستخدم إلى محطة البث. الاطار الشامل لتوزيع الموارد يحقق الوظائف الآتية وفقاً لمواصفات نظام "التطور البعيد المدى" اللاسلكية للارسال من المستخدم إلى محطة البث: تخصيص عرض النطاق الترددوي للارسال، التحكم في قوة الإرسال، تخصيص نظام التعديل والترميز.

تحدد نظام "التطور البعيد المدى" تكنولوجيا "الأستخدام المتعدد للترددات المتعامدة – حامل أوحد" لاتاحة استخدام عرض النطاق الترددوي من قبل أكثر من مستخدم في نفس الوقت للارسال من المستخدم إلى محطة البث. يتم استخدام هذه التكنولوجيا لميزة صغيرة نسبة بين أعلى قوة ارسال ومتوسط قوة الارسال، ولكن هذه الميزة تفرض أن يكون تخصيص الحوامل التابعة لمستخدم ما متجاور مما يفرض قيد على تصميم النظام.

مع التقدم في التكنولوجيا يوجد الآن تطبيقات متعددة و يوجد لكل تطبيق تحديات و متطلبات خاصة به. على سبيل المثال تطبيق الاتصالات عبر بروتوكول الإنترنت يتطلب أن يكون التأخر في توصيل حزم المعلومات أقل ما يكون بينما تطبيقات بروتوكول نقل الملفات لا تتطلب ذلك. يتم أخذ هذه المتطلبات المختلفة في الاعتبار عند تخصيص عرض النطاق الترددوي للارسال. عند التحكم في قوة الإرسال يجب الأخذ في الاعتبار قوة التداخل المولدة على الخلايا الجاورة، في المقابل من أن تخفيض قوة التداخل المولدة على الخلايا الجاورة يؤدي إلى تخفيض أنتاجية الخلية إلا أنها تؤدي إلى زيادة أنتاجية النظام ككل. الاتجاه المقترح يؤدي إلى زيادة الانتاجية والكفاءة الطيفية بينما يراعى مستوى الخدمة القدم لكل مستخدم بالإضافة إلى تنفيذ نظام تنسيق التداخل بين الخلايا.
اطار شامل لتوزيع الموارد في نظم "التطور البعيد المدى" اللاسلكية للارسال من المستخدم إلى محطة البث

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